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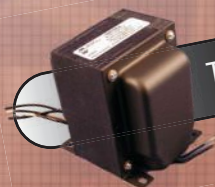


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The Binaural Way

From time to time Binaural Recording re-emerges as a hot topic, but it seems this time we could be facing a big revival. After all, it makes sense to offer binaural recorded content considering that people increasingly use headphones, a market which continues to grow—according to Futuresource Consulting's latest 2017 quarterly report, headphone business volume has grown 24% and forecasts show continued growth throughout 2017.

We have also seen binaural reproduction increasingly mentioned in radio and television broadcast evolution demonstrations, with the BBC even going as far as scheduling regular programming in binaural (BBC Radio 3). And in France, the BiLi Project (www.bili-project.org), an experimental research project and consortium concentrating on binaural listening that promotes interesting demonstrations and regular broadcasts. See also ARTE (www.artradio.com/theme/binaural).

All new-generation digital broadcast standards, including ATSC 3.0 (atsc.org), consider binaural rendering on headphones a natural extension of the immersive audio standards—MPEG-H 3D in particular. <http://ieeexplore.ieee.org/document/7874294/> More information is also available in our Standards Review article on MPEG-H found online.

www.audioxpress.com/article/Standards-Review-Considerations-for-the-MPEG-H-Audio-Standard

A simple search on the Internet about binaural recording and playback offers a world of information and content to explore. There's no doubt that binaural sound technology enables the creation of immersive spatial audio experiences for headphone listeners, which in turn is inspiring new companies to explore simple and affordable solutions for users to play with binaural sound using modern mobile devices. More recently, a new field of research has brought renewed interest in using binaural recording (and ambisonics) to generate immersive interactive experiences targeting virtual reality (VR)—probably its most promising application field for the future. At the same time, there's been some progress in getting binaural reproduced on speakers using Crosstalk Canceling techniques, but that would be the topic for a dedicated article (which *audioXpress* is exploring).

Of course, as it's been frequently discussed at Audio Engineering Society (AES) conventions and other forums, it's all about the content. Some detractors claim binaural is not a "natural" way to enjoy music while others think it makes sense for music recording. Just visit the *audioXpress* June 2017 Highlights page on www.audioxpress.com to download or stream it.

In this edition of *audioXpress*, Neville Roberts details a recording session promoted by British record label Chasing The Dragon, which believes experimenting with binaural makes complete sense and are determined to prove it, using the Neumann KU 100 system in combination with established techniques. The label is making the results available in a variety of formats, from a direct vinyl cut, to 24/192 PCM and DSD—downloads also available at <http://chasingthedragon.co.uk>. To complement Roberts' article, Chasing The Dragon has kindly made available two 1 minute 30 second excerpts from the Chabrier's "España" recording as a single, 24 bit /192 kHz PCM WAV file for our readers to listen and compare. The first half is from the conventional stereo mix and the second half is from the binaural recording as a studio master file.

Neville Roberts shared some interesting additional thoughts on this experience: "The diffuse-field approach of the Neumann, coupled with the refinements offered by the KU 100, makes for a really exciting opportunity for recordings that will suit the growing number of headphone listeners and yet should still be playable well through loudspeakers."

Interestingly, Sennheiser and Brüel & Kjær both recently launched two completely different new binaural recording solutions. Sennheiser introduced the new AMBEO Smart Surround solution—a compact earphone for immersive 3D audio recording that lets consumers capture and listen to binaural audio recordings on smartphones with unprecedented simplicity. This is another tool in Sennheiser's ambitious AMBEO 3D audio technology program, which the German company believes will be "the future of audio."

Brüel & Kjær also announced a new binaural microphone set recording solution—type 4101-B, which the Danish company believes will be interesting also for virtual reality and augmented reality, apart from having useful applications in test and measurement. More details are available on the *audioXpress* website (www.audioxpress.com/tags/binaural).

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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.

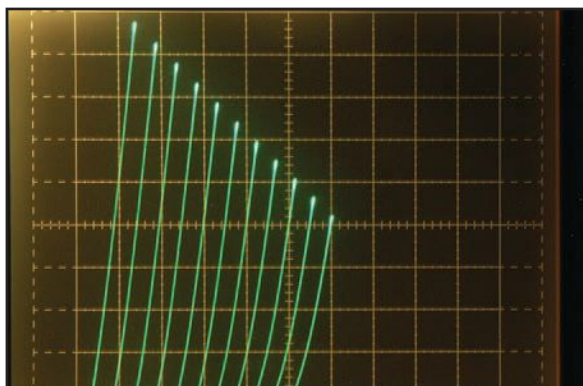
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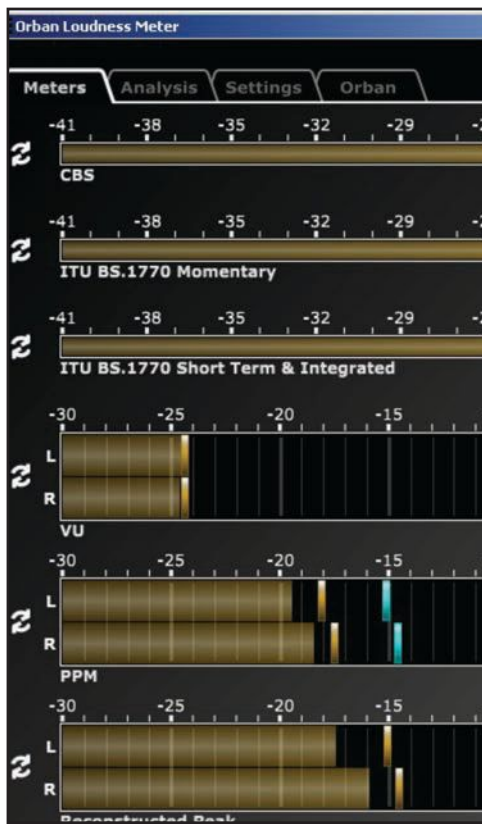
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This article revisits the author's "Sources 101" article, originally published in *audioXpress* in April and May 2007, as well as several follow-on letters, available online on the audioXpress website.

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Loudness Revisited (Part 1)

It All Started with Volume Control

Now that media streaming has become a preferred way of listening to music, we all want to avoid the need to constantly adjust the volume control. Loudness normalization, the attempt to avoid having to adjust the volume, has become desirable, if not a true necessity. But it isn't that easy a problem to solve.

By
Ron Tipton
(United States)

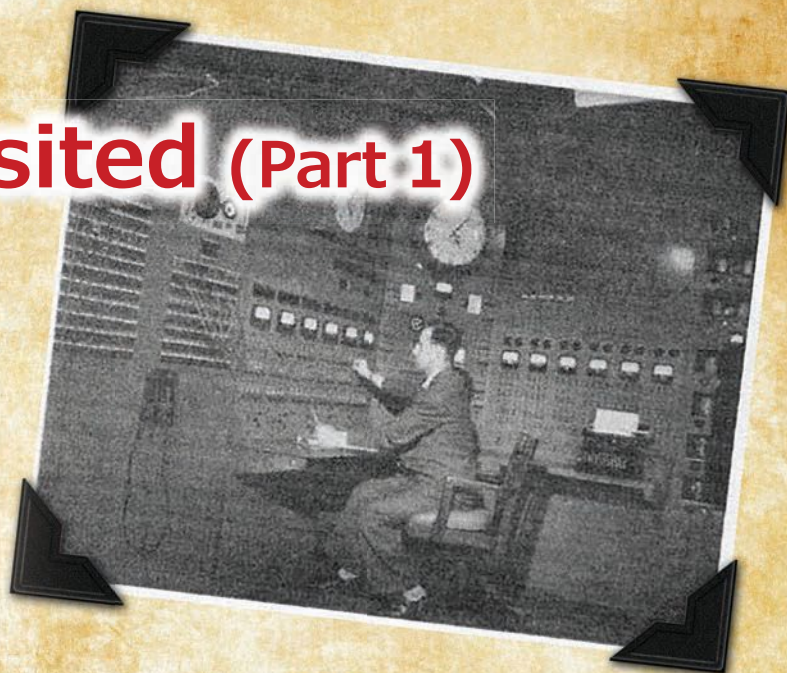


Photo 1: A single operator in the CBS Shortwave Master Control Room is responsible for controlling the volume on 12 outgoing lines. Not an easy job because of the meter settling time. The original CBS Shortwave broadcasting began in New York City on November 3, 1935 with the call sign W2XOV.

Radio broadcasters beginning in the 1930s didn't like listener complaints about "volume control riding." Some of the early work on this problem was done by Harold A. Chinn at Columbia Broadcasting System (better known as CBS). So, we'll start with a brief history of loudness control or "automatic volume control" as it was called at that time—but there wasn't anything "automatic" about it.

A Brief History of Volume Control

The first problem to solve is how to measure the volume. Chinn authored and co-authored several papers on this subject from the mid-1940s to the early 1950s. The electronics of that era limited his choices: Do you measure the RMS or the peak?

According to Chinn: "Data on peak checking showed such a marked advantage for the RMS type as compared with the peak instrument that it was decided to employ the RMS type of instrument. Other considerations included the possibility of employing copper-oxide rectifiers thereby eliminating vacuum tubes..." **Figure 1** shows the basic volume control and meter for 600 Ω line bridging.

No averaging capacitors were used in this circuit so the ballistics of the meter are important. Slight under-damping was found to work best. Although several meter scales were used, it seems the preferred scale was percentage, with 100% prominently displayed at about 70% full scale followed by a red "danger" zone.

As shown in **Photo 1**, a single operator had control of 12 meters—apparently that was where the "automatic" name originated. This was not an easy job because the meter settling time was 5 to 10 seconds for speech and 1 to 2 minutes for music.

Laboratory measurements showed this meter to be consistent to within about 2.5 dB for male and female speech, male singing, and dance and symphony orchestra music. Although some people can hear a 1 dB volume change, the norm seems to be about 2 dB as reported by Mark Sanfilippo in his article, "Human Hearing: Amplitude Sensitivity Part 1" (www.audiophiles.com/room-acoustics/human-hearing-amplitude-sensitivity-part-1).

Apparently, this meter worked well for many years because Chinn goes on to say: "The measure of success

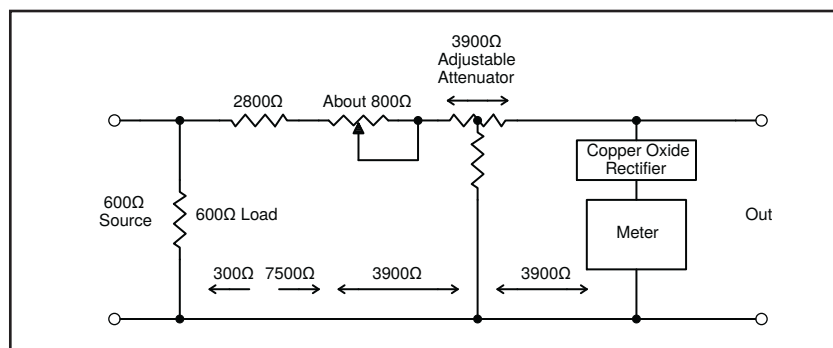


Figure 1: This 600 Ω line bridging volume control and meter used a copper-oxide rectifier for the rectifier characteristics and to eliminate a vacuum tube diode. The 3,900 Ω adjustable attenuator sets the line volume while the "about 800 Ω " control compensates for the difference between meters so they all read the same for a standard input.

in this instance being almost the complete absence of listener complaints concerning the relative loudness of speech and music.” (A more detailed paper on loudness measurement history is included in the Supplementary Material as Marks-and-Florentine.pdf.) CBS loudness designs are still alive and well. But, let’s skip ahead and look at some Orban products.

Enter the Orban Processor

Orban designs and manufactures commercial hardware loudness controllers for radio and TV broadcasters and streaming services. They typically cost many thousands of dollars so readers are probably more interested in their free loudness meter for Windows and Mac.

According to an Orban white paper (included in the Supplementary Material), Robert Orban writes: “Since first licensing the CBS algorithm and using it in its Optimod-TV 8182 in the early 1980s, Orban has continually refined and developed this technology for 30+ years...”

The perhaps more common loudness measuring algorithm, ITU-R BS.1770, had the shortcoming of integrating the entire musical track, which included near silence and very loud passages. The standard was revised to 1770-2, 1770-3, and finally (so far) to 1770-4. Gating was included to ignore the quietest 10% and the loudest 5%. The recommended normalization is based on this new integrated value.

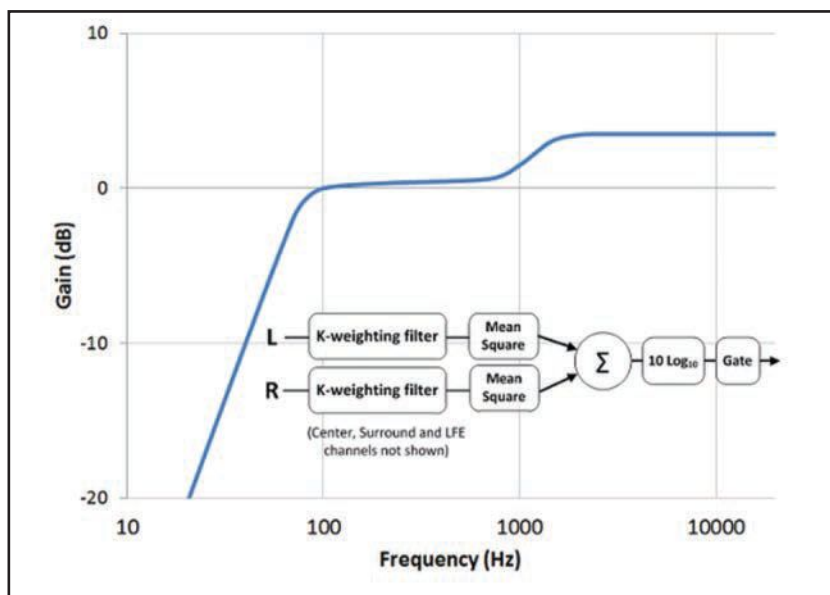
In contrast, the CBS algorithm recommends normalization based on a so-called “anchor element,” which is usually dialog except when the material is all music. This agrees with Dolby Laboratories’ assertion that most listeners set their volume controls for intelligible dialog.

However, now some BS.1770 software loudness meter vendors recommend normalization based on the 3-second sliding integration, “S,” rather than the whole selection integrated “I” value. Other vendors are still recommending using “I.”

I seemed to have better results with “S,” as mentioned in my Streaming Music Experiences articles (*audioXpress*, January–May 2017). However, “S” may work better for shorter selections and my tracks were commonly about 3 minutes in duration. Perhaps it would be wise to experiment.

The BS.1770 and the 3342 Loudness Meters

Figure 2 shows the BS.1770 block diagram with the K-weighting filter frequency response. K-weighting is controversial because it’s counter-intuitive. It has a 4 dB rise in gain above about 1,200 Hz, while the Fletcher-Munson (F-M) hearing sensitivity curves shows we have excellent acuity in



that frequency range (see **Figure 3**). To read more about the controversy, see the blog about K-weighting on gearslut.com. However, Bronwyn L. Jones and Emil L. Torick assert in a Society of Motion Picture and Television Engineers (SMPTE) paper that the F-M curves are not applicable to loudness measuring. Perhaps K-weighting is correct because the developers say psychoacoustics (the way we humans hear) were applied. The Jones and Torick algorithm improves upon the original loudness measurement algorithm developed by CBS researchers in 1967. Its foundation is psychoacoustic studies done at CBS Laboratories over a two-year period by Torick and the late Benjamin Bauer, who built on S. S. Stevens’ 1950s-era work at Harvard University.

Broadcast loudness standards have evolved differently around the world, but they are mostly

Figure 2: The stereo BS.1770 loudness meter block diagram is shown with the K-weighting filter’s frequency response. All channels are summed to produce a single output value.

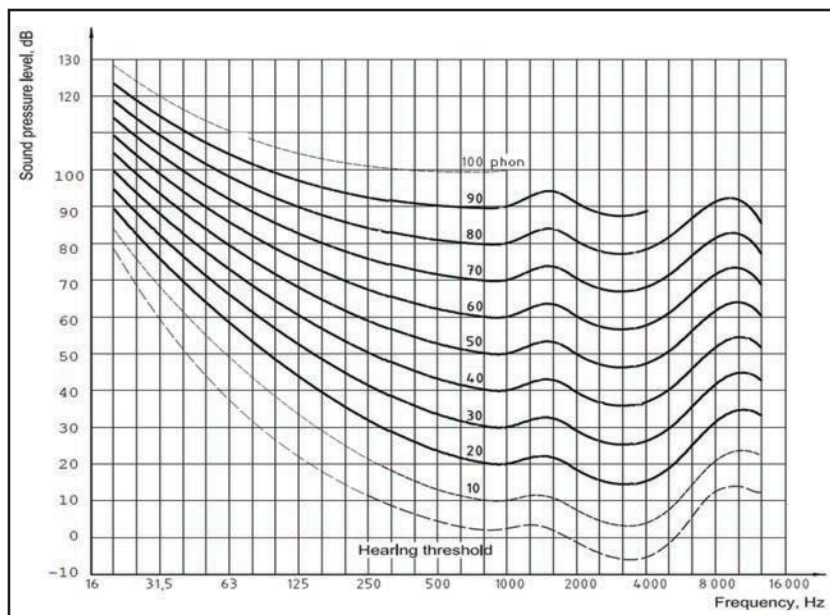


Figure 3: The Fletcher-Munson (F-M) curves graph our hearing sensitivity vs. frequency. Our hearing acuity falls off rapidly below about 800 Hz. Emil L. Torick and Benjamin Bauer maintain that this graph is not useful for loudness measurement.

based on ITU BS.1770-3, which is the International Telecommunication Union's BS.1770 recommendation, published in August 2012. The ITU standard concerns Broadcast Loudness and True-peak Level measurements (which will be addressed in a separate article), and the loudness part is based on an Leq measurement employing K-weighting.



Photo 2: This screenshot of the Orban Loudness Meter shows my Settings choices. An ASIO compatible sound card is required. I had to choose "ASIO 2.0 - ESI Juli@" from the drop down menu. Under the ITU BS.1770 heading I found the Integration Time setting gave the same results for any time value.

Broadcast loudness standards based on ITU BS.1770 include ATSC A/85 (the US), EBU R128 (Europe), OP-59 (Australia) and TR-B32 (Japan).

The P/LOUD group, which is part of The European Broadcasting Union (EBU), has defined the R128 standard based on ITU BS.1770. However, the group also added new tools such as a relative gate that ensures even more consistent loudness across genres and types of program material. Some of these tools have been implemented in the updated version of ITU's recommendation: ITU BS.1770-3.

Basically, the R128 standard builds on four technical documents: EBU Tech 3341 (which specifies loudness metering and defined LUFS), EBU Tech 3342 (measurements and normalization), EBU Tech 3343 (guidelines for production) and EBU Tech 3344 (guidelines for distribution in various formats). EBU Tech 3342 specifies audio normalization based on loudness measurements, where "the average loudness level, or Program Loudness, should be used in combination with Maximum True-peak Level and Loudness Range (LRA) to correct program material in accordance to the R128 specifications."^[1]

The EBU-TECH-3342 meter has a similar block diagram, but without the K-weighting filter. The zip file called ebu-loudness-test-setv05.zip contains 70 audio files to test loudness meters for 3342 compliance. The tech3342.pdf contains test instructions. The files are included in the Supplementary Material on the audioXpress website.

Orban Loudness Meter

Also, we have this from the *Orban Loudness Meter User Manual*: "Loudness is subjective... A meter that purports to measure loudness must agree with a panel of human listeners." Apparently, the Chinn 1951 meter did measure fairly well based on the lack of listener complaints. Perhaps normalization based on

Project Files

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

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[1] "Tech 3343: Guidelines For Production Of Programmes In Accordance With EBU R 128," 0148 European Broadcasting Union, <https://tech.ebu.ch/docs/tech/tech3343.pdf>.

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E. L. Torick and B. Bauer, "Researches in Loudness Measurement," *Institute of Electrical and Electronics Engineers (IEEE) Transactions on Audio and Electroacoustics*, Volume AU-14, September 1966.

“S” will stand the test of time. But as I mentioned at the beginning of this article, the path to equal perceived loudness isn’t an easy one.

The Orban meter can simultaneously display instantaneous peaks, volume unit (VU), Peak Program Meter (PPM), CBS Technology Center loudness, and ITU BS.1770 loudness. Unfortunately, the meter does not include full peak hold, but it is still useful. The User Manual says the Windows Audio Session Application Programming Interface (WASAPI) loopback works to meter music being played through your WASAPI sound card. I could not get this to work with any of my cards that support WASAPI so I used an external audio source. Using another computer, I connected its sound card’s stereo out to the Orban Meter computer’s ASIO sound card in. This arrangement worked fine. The 23-page User Manual is included in the Supplementary Material, and I suggest you read it because it contains a lot of useful information!

Photo 2 shows my choices on the Settings screen. The User Manual recommends that both the BS.1770 Integration Time and the LBA Integration Time be set to a value at least as long as the sound segment to be measured. I got the same measured results for Integration Times of 3 seconds and 240 seconds for a track length of 180 seconds.

Photo 3 shows a capture of the Meter’s screen. Because there is no peak hold, this capture was taken at a time near the end of the measurement. The bar labeled ITU BS.1770 Short Term & Integrated is just the Integrated “I” time. Apparently, the Short Term value is labeled BS.1770 Integrated Loudness (see **Photo 4**). The histogram display is a measure of the loudness range.

For the screenshot shown in **Photo 3**, the program was written so the CBS and ITU BS.1770 Short Term and Integrated bars are comparable even though the units are decibels and Loudness K-weighted relative to Full Scale (LKFS). Also LKFS and Loudness units relative to Full Scale (LUFS) are the same.

The Orban meter is the only one I found that measures both CBS and BS.1770. Version 2.8.2 runs on Windows 7 and higher. For Windows XP and Mac (but not OSX) you need version 2.0.8. Both are free downloads from the Orban website. However, the CBS measurements do not produce a short-term “S” value so you would have to use the “I” value for normalization.

In the next article, I’ll begin a comparison of free and not-free loudness measuring software and demonstrate how you can use the measurements to normalize sound tracks from a variety of sources including 78 RPM records, audio and video cassettes, reel-to-reel tape, 45 RPM and LP records, downloads, and streaming captures. 📧



Photo 3: The Meters screen was captured near the end of the 3-minute track. The radio-button at the left of the LBA (loudness range) bar does hold its reading and the third bar’s “I” value but everything else is cleared. You do need to turn the Meters on by clicking the on/off button, bottom center. The button lights when it’s on.



Photo 4: The Analysis screen does a faster-than-playing-time scan of the selected file in the left side pane, in this case alpert08.wav. This is the eighth track, “Let It Be Me,” of the LP album *Herb Alpert and the Tijuana Brass, The Lonely Bull*, A&M Records 101. The left and right waveforms are shown with the loudness range histogram. The BS.1770 Integrated Loudness value is apparently the short-term or “S” reading. The Highest Reconstructed Peak should be less than 0 dBFS.



Even's EarPrint Technology

Precision Listening for Earphones and Headphones

Even is reinventing personal audio with technology that tunes sound to each listener's unique hearing with its patent-pending Even EarPrint process. A veteran group of sound designers, acoustic engineers, mastering technicians, and audiologists have perfected a personalization process for headphones and earphones that measures the user's reactions to a series of eight different frequencies in each ear in only 90 seconds. This enables correction of earphone and headphone responses each time they are used.

By
Mike Klasco

(United States)

Most of our readers are familiar with the room compensation/speaker correction systems for recording studios and home listening. The calibration process consists of the generation of test signals that are sent to the amplifier and the individual speakers. The response of the speakers at various locations are then captured by a measurement microphone at the listening position(s).

While these compensation techniques for room/speakers can make substantial improvements to the accuracy of sound reproduction, there are some intrinsic limitations. Specifically, during calibration you have to choose whether you want one sweet spot or a compromised average for multiple listening locations—neither of which address the listener's hearing deficiencies.

Most of us start out without hearing defects, but life has a way of putting wear and tear on our hearing. Too many rock concerts or too many years on subway trains take their toll. NYC commuters typically have a cookie cutter bite mid-band from screeching wheels and then there is "the musician's ear" or worse "the shooter's ear," all of which affect your hearing. It is not unusual for the left ear to have

a response different from the right ear. Stiffening to the tympanic membrane due to aging droops the top-end response.

Audiology testing procedures for hearing aid fitting is not really the right procedure for earphone/headphones as the bandwidth is not wide enough for music auditioning. The standard and most common type of hearing test is pure tone audiometry, which we will discuss in more detail—but essentially the process measures the air and the bone conduction thresholds for each ear in a set of six standard frequencies from 250 to 8,000 Hz.

EQ correction for earphones and headphones can benefit even more from response processing than speaker/room correction. The patent-pending Even EarPrint technology (currently five patents have been filed with more on the way) measures your reactions to a series of eight different frequencies in each ear, based on the industry-standard "threshold of hearing test" used by audiologists and the test has been extended to better fit the music spectrum. A high-resolution version uses 16 bands (wider frequency coverage, narrower bands, and higher-resolution amplitude range).

EarPrint then compensates differently for each ear, creating a precise improvement in the sound and music listening experience. The listener is now “in the correction loop” along with the earphone/headphone including compensation for your left and right ears idiosyncrasies. Unlike loudspeakers and rooms, there is no compromise when averaging for multiple listening positions. While earphone and headphone responses can vary each time you put them on, the Even EarPrint calibration procedure is only 90 seconds and the process can be done each time for critical auditioning.

The Inception of the Even EarPrint

Danny Aronson is a musician, commercial sound designer, and the co-founder of Even (see **Photo 1**). Aronson eventually realized his hearing was deteriorating over time. He had the same issue with his eyesight, which was easily solved with a pair of prescription glasses. In a conversation with Ofer Raz, a veteran CTO and entrepreneur, Aronson and Raz wondered why headphones weren’t more like glasses. And questioned whether headphones could adapt sound to the way each individually hears, as is



Photo 1: Danny Aronson, Even's co-founder and CEO, is a classically trained composer, musician, and commercial sound designer. When Danny realized his hearing was deteriorating over time, he asked: “Why aren’t headphones like glasses?”

done with eyesight, to provide 20/20 hearing.

Each person’s hearing is as unique as a fingerprint—everyone hears frequencies differently and each person hears differently in each ear (see **Figure 1**). Hearing also changes over time—and not for the better. Most people think that only people with severe hearing loss need any kind of audio enhancement. The truth is that we ALL need audio

Introducing The Even EarPrint

Personalizing sound to your unique hearing profile

Each and every person’s hearing is unique - it’s imperfect, unbalanced, and changes over time, yet current audio products treat us all the same.
It’s time personal audio got... personal

Using ground breaking DSP technology, Even’s patented onboard hearing test will determine your EarPrint - your unique hearing profile, and our proprietary software will tune each and every sound to you, in real time, for each ear individually, resulting in a listening experience like no other. Developed over two years by a group of Audiologists, sound designers, mastering engineers and musicians, Even’s EarPrint technology has been recognized as a revolutionary step in personal audio.



Try our online demo at www.weareeven.com | Licensing inquiries: partner@weareeven.com



Figure 1: The patented EarPrint process creates a unique “fingerprint” for a person’s hearing. The pictogram shows an evaluation process available on Even’s website.

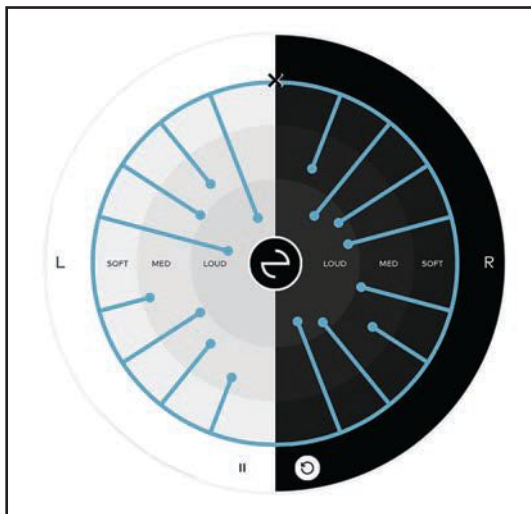


Photo 2: Even’s H1 Headphones feature the company’s patented EarPrint technology with on-board hearing test and real-time frequency compensation, in a state-of-the-art design with 50 mm beryllium drivers.



Photo 3: The Even EarPrint’s electronics fit into an earphone remote control and include a dedicated button to toggle the compensation on and off.

that’s tuned to the way we hear. Legacy audio devices are a “one-sound-fits-all-once-and-for-all” design.

In a category that equates extravagant price with quality isn’t it time that someone challenged the basic precepts of the audio world—the notion that we all hear exactly the same and that we all have perfect hearing—and created an affordable solution? Especially when a visit to a hearing aid center could set you back a few thousand dollars. Shouldn’t everyone be able to hear more precisely, without the need to pay an arm and a leg for what is basically an outdated, one-sound-fits-all design? But how do you get people to understand what they are missing and (literally) paying heavily for it?

Connecting with a team of audiologists, mastering engineers, and sound designers, Even developed a 90 second, self-administered “EarPrint” evaluation test that determines how you hear in each ear across a wide range of frequencies and provides a patent-pending portfolio of algorithms that compensates, in real time, for your specific hearing.

For reference, a conventional audiological hearing test provides an evaluation of the sensitivity of a person’s sense of hearing and is most often performed by an audiologist using an audiometer. An audiometer is used to determine a person’s hearing sensitivity at different frequencies.

Prior to the hearing test itself, the ears of the client are usually examined with an otoscope to make sure they are free of wax, the eardrum is intact, the ears are not infected, and the middle ear is free of fluid (indicating a middle ear infection). The most common reasons to develop hearing loss are due to genetic disorder, aging problems, exposure to noise pollution, infections, birth complications, trauma to the ear, and certain medications or toxins.

The standard and most common type of hearing test is pure tone audiometry, which measures the air and bone conduction thresholds for each ear in a set of six standard frequencies from 250 to 8000 Hz. The test is conducted in a sound booth using a pair of headphones connected to an external audiometer. The result of the test is an audiogram diagram, which plots a person’s hearing sensitivity at the tested frequencies.

The Even EarPrint is similar to a conventional audiologist evaluation but targeted toward optimization for music auditioning. Today, the Even’s EarPrint electronics fit into an earphone remote control dongle or a Bluetooth DSP processor. Even is now licensing its EarPrint process to earphone and headphone brands and manufacturers in addition to offering its own Even-branded earphones and headphones (see **Photo 2**).

The Even EarPrint Test Workflow

The Even EarPrint uses a modified threshold of hearing test to get a user's "Music Auditioning Audiogram," which is an indication for the algorithm to figure out what compensation is needed for a better music listening experience. The hearing test uses filtered narrow bands of music. It is a change from the classic sine waves and, according to Even, is more suitable for a music-oriented hearing test.

The algorithm is the sophisticated part of the system that refers to the EarPrint and sends instructions to the compensation part of the system as to how much to change and in what frequency range. It also takes into account the differences between ears and an internal balance of the "ear print."

The compensation is a tailored equalization curve for the specific scenario, it uses "musical" curves researched to be effective for music listening with as little phase shifting as possible. All processing is done in real time. The system has an A/B function to compare the before and after process. Sound is sampled in high resolution to enable good reproduction of music. The system is implemented in a chip within the remote and does not require an app to be installed or even to be connected to a device for the process (see **Photo 3**). This makes it compatible with most of the music players, cell phones, and laptop computers on the market (see **Photo 4**). For more information visit www.geteven.co. 

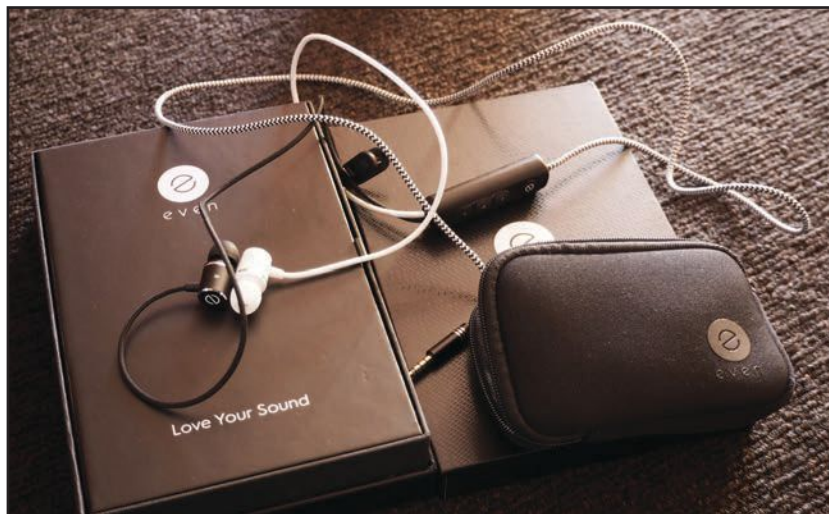
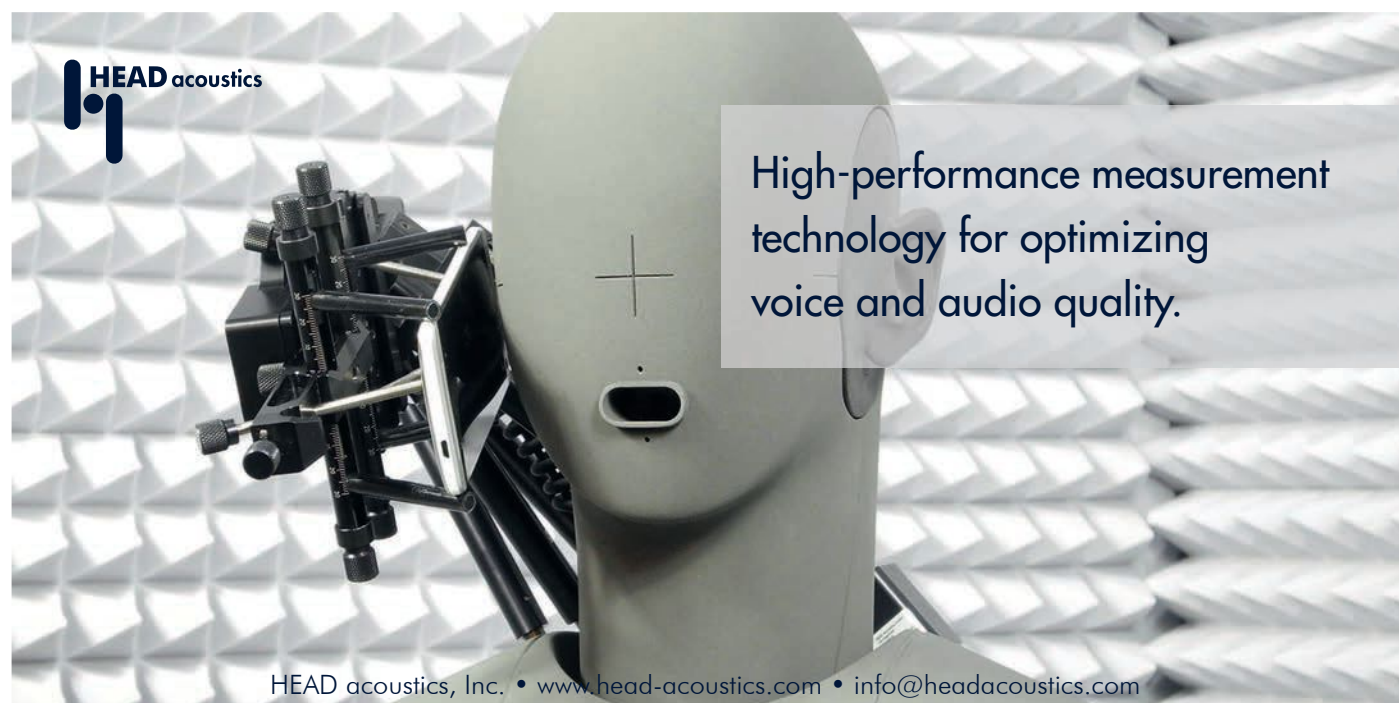


Photo 4: Even's E1 Earphones with EarPrint technology feature 10 mm dynamic drivers for a personalized and pleasant musical experience.

Editor's Note: For readers interested in a comprehensive survey and analysis of these speaker/room correction techniques, we recommend Ron Tipton's "Room Correction" article series (audioXpress, July–December, 2016). 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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Even H1 Headphones—Hands-On Evaluation

We passed a set of the Even earphones and headphones to Alex Vlach, our audio tester, to get his opinion on this new technology.

By **Alex Vlach**

(United States)

If you haven't heard about Even's H1 headphones yet, chances are you soon will because the company has developed a clever and unique feature that could potentially change the way headphones are manufactured and vastly improve the listening experience of headphones in general.

This feature is included on Even's two introductory products: the Even H1 portable closed-back headphones and the Even E1 earphones. This review will focus on the H1 headphones.

Even created a device that inserts into the headphone cables and resembles an inline volume controller, however, this is much more than a simple volume controller with an on/off switch. The built-in unit (dongle) is a sound processor that takes an "ear print" of your hearing and stores it in memory (see **Photo 1**). Within the dongle is a lithium ion battery that provides 12 hours of operation, along with the usual functions of the inline remote control—Play, Pause, Volume, Call Control, and microphone.

The concept makes a lot of sense if you think about it since presumably a 52-year-old man does not have the same hearing as an 18-year-old girl. Both can use the exact same headphones, yet hear differently. But what if you could customize a headphone's frequency response and personalize it to your unique hearing? That is exactly what this new product from Even offers, it lets you create your own "ear print!"

Even's entire EarPrint process takes less than two minutes to set up. By turning on the unit you are greeted with a voice to guide you through the calibration, which involves listening to eight test tones in each channel. You are asked to click a button as soon as you audibly detect each test signal. For best results, it is important to perform the calibration in a quiet environment with minimal background noise.

Once your ear print is set and stored in the memory, you can easily compare the results by finding your favorite track on any device and turning your ear print ON or OFF to hear the difference.

I completed my evaluation test and was impressed by the difference I heard when toggling my profile on and off. When activated, the overall audio signal gets a slight boost since the amplitude of some frequencies are increased to compensate for your hearing loss. The end result is that things just seem to snap into place sound-wise.

Even H1 Headphones

The H1 is an over-the-ear closed-back headphone. It is stylish and beautifully designed with attractive walnut

wood ear cups. The drivers are generous—50 mm diameter with chemical vapor deposition (CVD) coated beryllium. The hinges are made from cast metal that pivot smoothly and feel solid. The faux leather ear pads are soft and comfortable. The same goes for the headband. The weight is medium, about 250 grams. The cables are detachable. They click firmly into the ear cup connectors and the headphones fold flat to be conveniently stored in the supplied carry case.

I evaluated the Even H1 headphones using my portable AK 120II player and a variety of files ranging from MP3 to FLAC and 96 kHz/24 bit pulse code modulation (PCM). The Even headphones present a warm and smooth sound with a rich and satisfying amount of bass. Those who find aggressive or sizzly treble annoying after a while won't have to worry about listening fatigue with the more laid back nature of the Even headphones.

I was expecting a reduced sound stage due to the nature of a closed-back design, however, I can report I didn't experience any shortcomings in that area. This is a versatile type of headphone you can use everywhere—at home, at the office, outside on the go, or travelling.

Overall, Even has put out a great first effort—a gorgeous, well-built, and very good sounding headphone with the ability to customize the audio frequencies to maximize its performance "even" further.



Photo 1: Uniquely combining science and design, Even's H1 Headphones feature a patented EarPrint technology with an on-board hearing test and real-time frequency compensation.

About the Author

Alex Vlach is a Project Manager for a large telecommunications company. He has a college degree in telecommunications and he has had a fascination for electronics since early childhood. His analytical mind coupled with 30 years as an audio enthusiast lends itself to evaluate a wide range of products.

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Head Space

A Binaural Recording Session



The concept of binaural or “dummy head” recordings was first thought of back in 1881. Then in the 1970s, binaural recordings for headphone listeners were all the rage. After that time, binaural recordings faded out of fashion and were considered to be too gimmicky for mainstream recordings—until now. New developments in binaural recording have inspired the UK-based record label Chasing The Dragon to start producing binaural recordings alongside its conventional recordings. The company recently conducted a unique experiment to investigate this format and compare it against more conventional recording techniques.

By
Neville Roberts

(United Kingdom)

Binaural recording as a concept was formulated as far back as 1881, shortly after the idea of sound recording itself was dreamed up. In 1857, a Parisian printer named Édouard-Léon Scott de Martinville devised a way of recording sound as a series of wavy lines drawn on a soot-covered paper cylinder called the “phonograph.” Unfortunately, from our point of view, he didn’t think of a way of playing the recording back! Although Scott de Martinville was a printer by trade, he was also an inventor and his interest in human anatomy led him to devise a system of recording sound, based on the human ear.

Scott de Martinville’s premise was that the best way to record sound would be to separately capture the sounds heard by the left ear and the sounds heard by the right ear. Later in 1881, another French inventor, Clement Ader, positioned several pairs of telephone transmitters across the front of the stage at the Paris Opera House and connected

them to listeners about a mile away. These listeners monitored the performances on the stage by holding a pair of telephone receivers to their heads—one to each ear. One pair of microphones was required for each listener because, in the days before amplification, a single pair of transmitting microphones could only power one pair of receivers.

In 1931, Alan Dower Blumlein invented what he called “binaural sound,” which we now call stereophonic sound or simply “stereo.” His invention was actually for loudspeaker stereo and this replaced binaural recording as the default two-channel system.

Blumlein’s approach was to position a pair of “Figure Eight” bi-directional microphones at 90° to each other (the so-called “coincident crossed-pair”). These microphones captured the left and right audio signals, which were then recorded and subsequently played back on a two-channel

system. Blumlein showed that the correct way to record stereo for loudspeaker replay was to use either coincident directional microphones or a pair of microphones spaced apart by roughly the width of a human head.

However, for headphone listening, it was better that the sound reaching each ear is modified by the acoustic properties of the human head. The brain is able to detect the spatial modifications that result from sound diffraction around the physical features of a head and from the delays between sounds reaching one ear with respect to the other ear. This spatial information is used by the brain to interpret where a sound actually originates. In 1933, the telephone company AT&T made quite a stir at the Chicago World Fair by demonstrating the first head-based binaural system. They created a mechanical man, "Oscar," which had microphones for ears and was set up in a glass room surrounded by listeners on headphones.

Dummy Head Development

At the 1972 International Radio and Television Exhibition in Berlin, German microphone company Neumann produced the KU-80, which was the first of a generation of commercial dummy head microphones. The KU-80 was originally designed as a test device for environmental acoustics research. However, it soon became evident that this dummy head had potential for creating binaural recordings. Initially used for recording realistic radio dramas, this new technology led to a growing interest in binaural music and a plethora of demonstration



Top: Mike Valentine is mastering the binaural recording. At right: This is the cover for Chasing The Dragon's *España* project.

LPs followed. These recordings were characterized by sound effects that ranged from typewriters to birds twittering in a forest, superimposed on popular orchestral pieces to showcase the binaural effect on headphones.

However, there are problems with reproducing binaural recordings on a loudspeaker system. One issue is that these recordings really only work when the listener's head is in one particular place and facing in one direction. Another problem is that the recordings can sound dull, lifeless, and boomy. Finally, there is the issue of crosstalk, which can damage the spatial information that your brain is trying to decode.



These rehearsals utilized the Decca Tree microphone configuration and a KU-100 dummy head (shown at right).



The Neumann KU-100 dummy head looks on during rehearsals.

To maximize the benefits of a binaural recording, you need to make sure that the signal from the left microphone only goes to your left ear and that the signal from the right microphone only goes to your right ear. This enables your brain to decode the spatial information from the recordings—and the easiest way to do this is with headphones. However, when you listen to a binaural recording through loudspeakers, the left/right signal separation isn't preserved. This is why conventional recordings typically use several microphones placed at strategic locations and on specific instruments to record the sounds, which are mixed together at a mixing desk by a recording engineer to produce the required stereo master.



During the recording, the Decca Tree microphone configuration and KU-100 dummy head sat right in the center of the orchestra.

One of the problems with the early binaural recordings was the use of free-field microphones in the dummy head. A free-field sound source is a sound field where the sound waves are free to expand outward forever from the source and are not subject to reflections or reverberations, as would be experienced in an anechoic chamber. Conversely, a diffuse-field, or random incidence field, is a sound field where the sound waves arrive from all directions. With a diffuse-field, it is harder for the brain to determine where the source of the sound.

Free-field microphones are designed to compensate for the effect of the microphone itself in the field and need to be pointed toward the sound source. Diffuse-field microphones are designed to respond in a uniform manner to sound arriving from any direction. The KU-80 dummy head used free-field microphones to simulate the free-field frequency response of the human ear at a particular point in the ear canal, but this resulted in a dull sound when listening with loudspeakers. In addition, the rise in the frequency response around 1.5 kHz at the eardrum, which is re-created by the KU-80, produced a "barrel" sound with loudspeakers.

Owing to the complex shape of the outer ear, the directional characteristic of the KU-80 was extremely frequency dependent. To address these issues and noting that modern headphones have a linear diffuse-field frequency response, in 1985, Neumann produced the KU-81 dummy head, which used diffuse-field microphones and was designed to produce recordings that worked well with both headphone and loudspeaker systems. Now, Neumann has updated this design and produced the KU-100, which improves the diffuse-field frequency response and uses omni-directional microphones. A re-design of the outer ears has improved the left to right spatial characteristics and prevents artificial vertical displacements of sound sources.

Tackling the issue of crosstalk is another challenge. Attempts have been made to solve this problem by using electronic filters, but these can introduce "spectral coloration." This means some frequencies are boosted and this adversely affects the sound quality of the recording. Research is being undertaken to minimize this degradation of sound quality by using smooth crosstalk cancellation, which applies different filter responses to different bands of frequencies. However, all filters are going to color the sound in some way. Although this is acceptable for virtual reality applications, it does raise alarm bells for audio enthusiasts, who want to keep the precious audio signal as pure as possible.

All these innovations have led the audiophile record label "Chasing The Dragon" to experiment



Rosie Middleton is pictured during rehearsals for the ground-breaking recording of *España*.

with the KU-100 to produce binaural recordings that can be played on headphones and with loudspeakers, by virtue of the careful positioning of the dummy head during the recording process. Furthermore, the company wanted to compare a dummy head recording with recordings of the same performance that were made simultaneously, using a conventional multi-microphone setup feeding a mixing deck to produce a conventional stereo mix.

The idea was to record the stereo mix signals digitally on two channels of the Nagra 6 digital recorder at 24 bit/192 kHz PCM as well as on a Tascam DA-3000 DSD recorder at the "double DSD" rate of 5.6 MHz, while at the same time recording the binaural signal on another two channels of a Nagra 6. This, as far as I am aware, would be a first-time opportunity to directly compare a conventional stereo recording against a binaural recording of the identical performance.

All that was needed was an orchestra to record, a location, and a date to undertake the recording. Mike Valentine of Chasing The Dragon—the mastermind behind the experiment—had already planned to record the National Symphony Orchestra under the baton of Debbie Wiseman MBE with the mezzo soprano Rosie Middleton. The result was to be a new direct-to-disk LP entitled *España*, which is a tribute to Spain and includes pieces by Nikolai Rimsky-Korsakov, Georges Bizet, and Alexis-Emmanuel Chabrier. Alongside the conventional microphone



Debbie Wiseman conducts the National Symphony Orchestra for this unique recording.

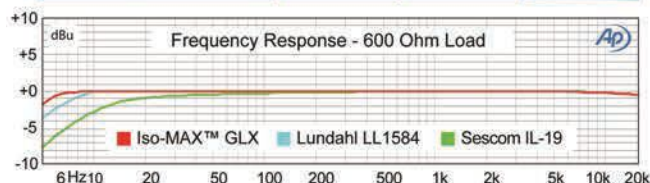
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Distortion @ +4dB:	.005%	.06%	.3%
Phase shift @ 20Hz:	0°	0°	20°
Ground lift switch:	Yes	No	No
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setup for the direct-to-disk LP and digital recordings, the plan was to also use the KU-100 dummy head to make a binaural recording of the performance. The location was the world-renowned AIR Studios in London and the date was February 10, 2017. In addition to myself, several other journalists had been invited, including Michael Fremer of Analog Planet, who flew to the UK for the occasion.



The Nagra 6 PCM digital recorder and TASCAM DSD recorder captured the performance.



The Studer 0.5" tape recorder was used to produce an analog master tape as a backup.

The Recording Day

The conventional multi-microphone setup used a combination of a Decca Tree of three microphones and several individual microphones positioned throughout the orchestra, feeding the studio's Neve mixing desk. A Neumann U47 tube microphone was set up to record the mezzo soprano. The output of the desk was a conventional stereo mix, which fed directly to a Neumann VMS 80 cutting lathe to make the direct-to-disk LP, to two channels of the 8-track Nagra 6 digital recorder to make a 24 bit/192 kHz PCM digital copy, to the Tascam DSD recorder to make a 5.6 MHz Double DSD digital copy, and to a Studer A820 0.5" tape deck running at 30" per second to produce an analog master tape as a backup.

For the binaural experiments, Valentine installed a KU-100 dummy head to produce a binaural stereo signal that was recorded onto two other channels of the Nagra digital recorder at 24 bit/192 kHz PCM. The binaural recording from this session was to be released as a separate binaural LP, a CD, and a studio master quality digital download. The conventional stereo mix will also be available as a direct-to-disk LP, a CD, and studio master quality digital downloads in 24 bit/192 kHz PCM and 5.6 MHz Double DSD.

Following an hour or so of morning rehearsals, it was time to make the first recording for Side 1 of the LP. This consisted of "Habanera" and "The Gypsy Song" from Bizet's *Carmen*—featuring the voice of Rosie Middleton—followed by Chabrier's "España." In the control room, I was able to listen directly to the output of the binaural recording on headphones while the recording was being made. At the same time, I was able to watch the orchestra through the panoramic glass window of the control room. I was amazed to have the impression that the sound was actually coming from each individual instrument, even though I was listening on headphones.

The requirements of a direct-to-disk LP mean that the recording of each side has to be achieved in a single take. During my discussion with Debbie Wiseman after the recording, I asked what additional challenges a single take had put on her as a conductor. She replied that for her, it was all about getting the atmosphere right. She explained that the pressure to "get it right" in one go creates a concert atmosphere and this was precisely what she wanted. Under these conditions, the orchestra feels that it is giving a real performance to a real audience. This results in something that is less dry and sterile than a conventional recording, which can be edited and tweaked afterward.

In my opinion, she certainly achieved her goal of creating a concert atmosphere. Incidentally, this is not the only time that Wiseman has had to cope

About the Author

Neville Roberts, a graduate in Physics, is a Chartered Scientist, Chartered Engineer, Chartered Physicist, and a Fellow of the Institution of Engineering and Technology. He is a retired Director of the UK National Health Service (NHS) and has also worked for the UK Ministry of Defence and in the private sector. He is a regular contributor to a UK audio magazine, *Hi-Fi Choice*, and has a particular interest in tube audio design and is a vinyl devotee. Apart from being an audio enthusiast, he enjoys a wide range of music including classical, especially baroque, light orchestral and jazz. He is a keen photographer and enjoys growing both tropical and hardy orchids at his home near Bournemouth in Dorset. He is also on the Committee of the Bournemouth Orchid Society.

with a unique challenge: As part of the Queen's Diamond Jubilee celebrations in 2012, she conducted an orchestra on an open barge going down the River Thames in London. She told me that even though it was raining hard and everyone was getting soaked, no one let that spoil the day and it turned out to be an enjoyable event for all concerned. The result was a wonderful musical extravaganza.

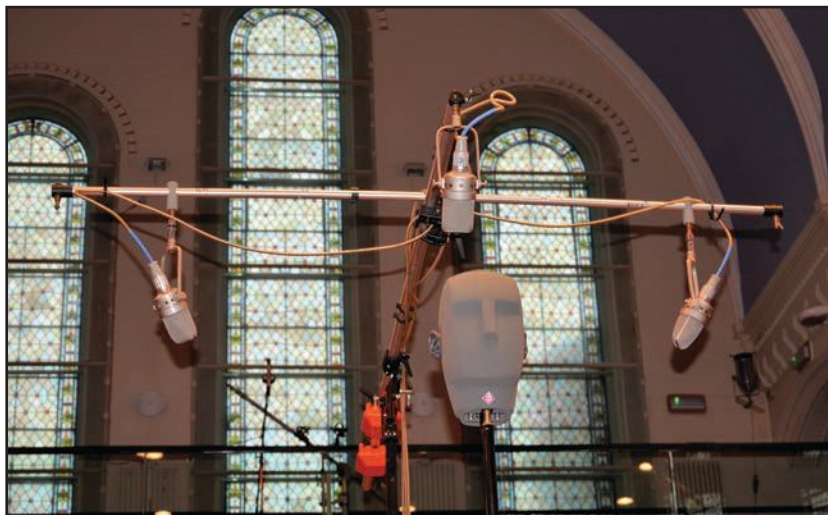
To get a performer's perspective, I spoke with Rosie Middleton after the recording. She too had been very aware that there was no chance to get anything wrong. As always, when preparing for performances, Middleton found herself muttering the words of the pieces everywhere she went for the previous month and a half. She also worked with répétiteurs to make sure that everything was vocally perfect. As for working "nose to nose" with a dummy head, Middleton said that she forgot the head was there until she noticed it about 30 seconds into the recording—and it was a bit of a shock! However, once she got over the initial shock, the dummy head just became part of the furniture. As a lover of the vinyl medium, she was really looking forward to hearing the LP and was excited about the prospect of comparing the binaural and conventional recordings.

The Results

To carry out my assessment, Valentine lent me a set of the top-of-the-range Stax SR-009 headphones with the SRM-007tII Kimik edition tube energizer. Incidentally, these were the same headphones that were used to master the binaural recordings. As mentioned earlier, this recording session provided the opportunity to compare not only digital vs. analog recordings but, for the first time, a binaural



Recording engineer Rupert Coulson, Mike Valentine, and Michael Fremer monitor the performance from the Neve mixing desk.



The Neumann KU-100 dummy head is pictured with the Decca Tree setup.



Cutting engineer John Webber uses a microscope to check the Neumann VMS 80 cutting lathe. At right is the lathe making the LP mastering.

recording against a conventional stereo mix—of the identical performance.

Starting with the recording technology, the one thing that all the recordings had in common was exceptional quality. The direct-to-disk LP is still, for me, just about as good as it gets. With regard to the digital recordings, they are both superb but to my ears, the double DSD recording just nudges ahead of the 24 bit/192 kHz PCM in that it has a slightly more refined dimension of authority and reality. Both digital files present a sound that is slightly more analytical than the analog LP version, but it would be wrong of me to declare an outright winner as personal taste plays a part here.

When it comes to the two different microphone setups, things are a little clearer. There is no doubt that the conventional stereo mix sounds really awesome on both speakers and headphones. The

binaural recording produced by the KU-100 head certainly results in a superior binaural recording that sounds incredible on headphones, while still producing a very acceptable sound through speakers. The sound does not suffer from the “hole in the middle of the sound stage” effect or from the very “left and right” sounds that I was used to with the early binaural recordings of the 1970s. Instead, I am presented with a wide sound stage, even on speakers, that fills my listening room. If I were to be a hyper-critical audiophile, I would say that with loudspeakers the binaural recording tends to move the sound stage further away behind the plane of the speakers. As a result, I feel as if I am sitting toward the rear of a concert hall. So my preference is for the conventional stereo mix when listening on speakers. However, the binaural recording really comes into its own on headphones. It presents a true three-dimensionality to the music reproduction with a far greater perception of space in front that you don’t get with the conventional stereo recording. I also experienced a greater involvement with the performances when listening to the binaural recording on headphones.

To summarize, binaural recording has certainly come a long way since the 1970s. It now has the capacity to offer the audio enthusiast what is arguably the best possible headphone listening experience, providing greater intimacy and involvement with the music. As with the early stereo recordings, binaural is no longer the domain of sound effects recordings, with trains rushing through one’s head or clocks chiming along with an orchestral performance of Charles Orth’s “In A Clock Store.” Although, in my opinion, a conventional stereo recording sounds best on loudspeakers, I am certain that the majority of the listening public will be very satisfied with the sound of a modern binaural recording on speakers, just as conventional stereo sounds excellent on headphones.

For me, the conventional stereo mix sounds best on speakers and binaural is best on headphones. But don’t just take my word for it—Chasing The Dragon has kindly made available two 1 min 30 second excerpts from the Chabrier’s “España” recording as a studio master 24 bit/192 kHz PCM file, which you can play and compare for yourself. The first excerpt is taken from the conventional stereo mix and the second from the binaural recording. This 215 Mb file can be streamed from my website at: <http://rssconsultancy.co.uk/files/ChasingTheDragon/Stereo-vs-Binaural-Chabrier-Espana.wav> or, it can be downloaded from <http://rssconsultancy.co.uk/files/ChasingTheDragon> by right-clicking on the filename and selecting ‘Save target as ...’. The success of the



Chasing The Dragon’s second binaural recording took place during a live concert—Mozart By Candlelight. Note the dummy head’s position.



This is the dummy head viewed from the front.

Chasing The Dragon

Chasing The Dragon is the specialist UK recording label run by Mike and Françoise Valentine. They have produced a number of audiophile recordings on LP and CD as direct-to-disk albums, including "A Tribute to Ella Fitzgerald" and Vivaldi's "The Four Seasons." As with the other recordings, the LPs and CDs of "España" and "Mozart by Candlelight" can be purchased from the US distributor, Elusive Disk at www.elusivedisc.com. In addition, digital downloads in studio-master quality 24 bit/192 kHz PCM and DSD are available via the Chasing The Dragon website at www.chasingthedragon.co.uk. You can also play MP3 samples directly from the website.

binaural recording for headphone listeners is very apparent, and all my expectations have certainly been exceeded. As a result, Chasing The Dragon will continue to produce a growing range of binaural records, CDs, and digital downloads to run alongside conventional stereo audiophile recordings.

Mozart by Candlelight

The following month, Chasing The Dragon conducted a second binaural recording. This time, it was during a live concert, Mozart by Candlelight, held at St Martin-in-the-Fields, which is located in the heart of London. Situated on the northeast corner of Trafalgar Square, St Martin-in-the-Fields is a church steeped in beautiful Georgian architecture. It is known for its superb acoustics and this premier London concert venue has been at the heart of the capital's music making for more than 250 years. From performances by George Frideric Handel and Wolfgang Amadeus Mozart during the 18th century through to the founding of the world-renowned Academy of St Martin in the Fields in the 1950s and to today, St Martin's hosts some of the finest ensembles, choirs and soloists in Britain.

On March 30, 2017, the church was host to the Locrian Ensemble of London who performed pieces by Mozart under the baton of Rimma Sushanskaya. The Mozart performances included "Eine kleine Nachtmusik," "Concerto for Flute and Harp," and the "Symphony No 29 in A."

The recording of the live concert was made on the KU-100 dummy head to produce a binaural stereo signal that was recorded onto the Nagra recorder at 24 bit/192 kHz PCM and also the Tascam DSD recorder. In addition, a spaced pair of DPA omnidirectional microphones were installed either side of the dummy head to make a conventional stereo recording onto the Nagra.

A particular challenge, which became evident during rehearsals, was getting the balance right between the flute and the harp. The flute is relatively loud in comparison with the harp and this required some subtle changes to the position of the dummy head and the DPA microphones. With a conventional multi-microphone setup, it is relatively easy to correct any imbalances via the mixing deck. Of course, this was not possible with a dummy head, spaced pair microphones, or with a live performance! However, it was found that moving the head back by about 50 cm and changing the angle slightly resolved the problem without adversely affecting the sound from the rest of the orchestra.

Listening to the recording afterward on headphones, I was amazed at the combination of extraordinary detail from all the instruments with

the superb ambiance of the church environment. When I closed my eyes, it was as though I had been transported back to the church and was listening to the live performance again! In particular, I actually preferred the binaural recording played back on my loudspeaker system over the stereo version recorded on the DPA microphones. I find the atmosphere and presence is somehow clearer with the binaural recording, which also has greater depth. The spaced-pair recording tends to bring the orchestra closer to the listener than the binaural version.

To conclude, binaural recording has really come of age and I understand that Chasing The Dragon has planned to make more themed recordings like *España* in the near future. This is certainly a very exciting time for headphone audio enthusiasts! 

A New Part Introduction for 2016

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

LSJ689 Series



The image shows three LSJ689 components: a large TO-18 package, a small SOT-23 package, and a tiny SOT-23 package. To the right is a graph titled 'LSJ689 Series' showing 'en Noise Voltage (pV/√Hz)' on the y-axis (log scale from 1 to 10) versus 'f=Frequency (Hz)' on the x-axis (log scale from 10 to 10000). A curve labeled 'New LSJ689 Dual P-CH JFET' starts at approximately 3.5 pV/√Hz at 10 Hz and decreases to about 1.8 pV/√Hz at 10000 Hz.

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Assessing Audio Quality Using the MATT Test

Learn how to use Musical Articulation Test Tones (MATT) to identify audio quality issues and obtain the highest quality sound possible from the components in your listening environment.

By
Richard Honeycutt
(United States)

We humans have been paying attention to optimizing sound in rooms for a long time. The ancient Greeks and Romans took steps to optimize speech intelligibility when building theaters. But probably the earliest era in which specific attention was paid to good room design for music performance was in Baroque times (J. S. Bach and contemporaries).

Modern acoustical engineering started with the work of Wallace C. Sabine on Harvard University's Fogg Lecture Hall. Sabine quickly recognized that the wretched lack of speech intelligibility resulted from excessive reverberation, so he developed methods for measuring and controlling reverberation. Thus, reverberation measurement and control became the first technological tool in the modern acoustician's toolbox. Reverberation affects speech intelligibility and musical articulation, meaning that concert halls designed for sustained Gregorian Chants muddled the sound of highly articulated Baroque or Classical music. Sabine applied his "reverberation tool" later when he was the acoustic designer for the Boston Symphony Hall, and he recognized the need for sound to be evenly distributed throughout the seating area.

Early:Late Energy Ratio

In large rooms such as concert halls and theaters, the only significant frequency-dependent acoustical effect is the variation of acoustical absorption with frequency. There is no reason for

anything like a "frequency response" measurement in these rooms. But in the latter part of the 20th century, other acoustical parameters were identified that have as much or more effect on music and speech articulation. These fell into the category of "early:late" energy ratios.

Figure 1 shows the impulse response (IR) of a somewhat unusual listening room. This IR shows the decay of acoustical energy over time. An example of an early:late ratio is "Clarity index" or "C80." C80 is the 10× the log of the ratio of total energy contained in the first 80 ms after the onset of an impulsive sound to the total energy after 80 ms. Clarity is specified in decibels and the acceptable range is about +1 to -4 dB, although the best concert halls have C80 between about -1 and -4 dB.

A related early: late energy ratio is "Definition" or "D50," which is the ratio of energy in the first 50 ms to total energy. Definition is specified in percentage and correlates more closely with speech intelligibility than with music articulation—C80 is more applicable in evaluating a room for music.

Speech Intelligibility

After the introduction of early:late energy ratios to the evaluation of auditorium acoustics, quantitative evaluation of speech intelligibility became an area of great interest. The first speech intelligibility metric in common use was called "percentage Articulation Loss of Consonants" or "%Alcons" and was introduced by V. M. A. Peutz

in 1971.^[1] The basis of determining intelligibility was the ability of listeners to correctly distinguish consonants pronounced by a talker in the room. A standardized list of test syllables was used. Peutz developed a method of predicting %Alcons from the reverberation time (RT) and background noise level in the room.

Tammo Houtgast, Herman Steeneken, et al., introduced a method of specifying speech intelligibility using the Modulation Transfer Function (MTF), which quantified the effect upon speech articulation produced by any defect in the electroacoustical and/or acoustical path between the talker and the listener.^[2]

The metric is called the Speech Transmission Index (STI). The underlying theory is that speech can be considered to originate as a continuous wave created by the vocal folds, and then modulated by the throat, lips, and tongue. The “intelligence” or information is carried by the modulation and the depth of the modulation correlates with speech intelligibility. Reverberation, background noise, and other acoustical or electro-acoustical effects can decrease the modulation depth of the acoustic wave reaching the listener’s ears, hurting speech intelligibility.

Note that neither of these specifications of speech intelligibility addresses the naturalness of speech: Both exclusively target the articulation of speech. The sound of a letter “t,” when stretched by reverberation, masked by noise, or electronically distorted, is difficult to distinguish from the sound of a “p” or a “d.” **Figure 2** and **Figure 3** show an anechoically recorded speech signal and that same signal played in a room having a 1.5-second RT, respectively. Notice how the reverberation closes up the spaces between sound bursts (i.e., decreases the depth of modulation). Speech in the reverberant room would have a lower STI, indicating poorer speech intelligibility.

Small Room Acoustics

In the early part of the 20th century, radio programming frequently originated as live performances in the radio station’s studios, so the sound of the studio and the control room became important. With the advent of home studios and home theaters in recent years, the subject of small-room acoustics has become more prominent.

Contrary to the situation in auditoriums and theaters, small rooms seldom have issues with reverberation. In fact, the smaller dimensions and corresponding shorter delay times in control rooms, studios, and so forth make the use of Sabine’s

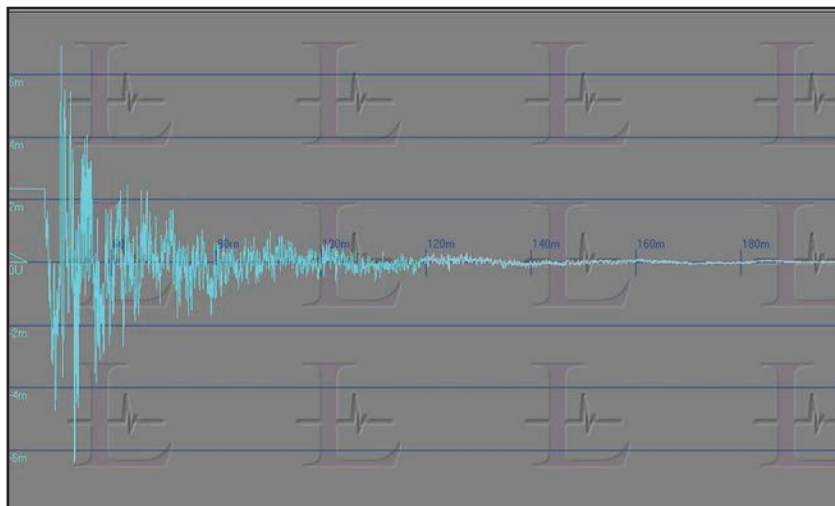


Figure 1: The impulse response of a listening room shows the room’s effect on the sound in the time domain.

statistical methods for specifying RT questionable for small rooms.

Small rooms do have two problems of their own: “boxy” sound quality and resonant modes. Unfortunately, these issues are often confused with one another. Boominess can result from resonant modes, especially when the acoustical absorption in the room is inadequate at low frequencies. However, boxiness can exist along with boominess or by itself. It is often caused by sound reflecting from surfaces near the speakers, then reinforcing or cancelling the direct sound at specific frequencies. This is called “comb filtering” because the resulting frequency response looks like a comb. Boxiness can be described as the impression of a small, acoustically reflective room we perceive from the aggregate of all sound (direct plus early reflections) within the 35-ms “fusion window” perceived by the ear-brain system. (Thanks to Art Noxon for this definition.) Decreasing the early reflections decreases boxiness.



Figure 2: Notice the spaces between sound bursts in this anechoic recording of speech.

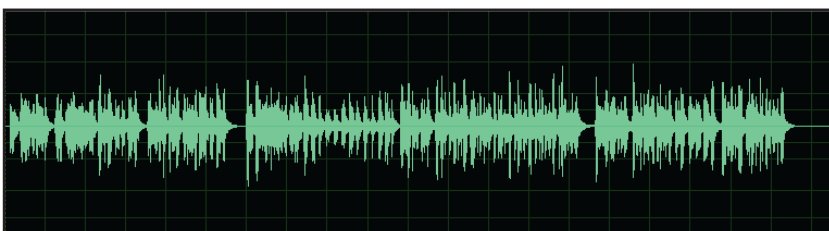


Figure 3: Notice how the reverberation in this 1.5-second room fills in the spaces between sound bursts, decreasing modulation depth and STI.

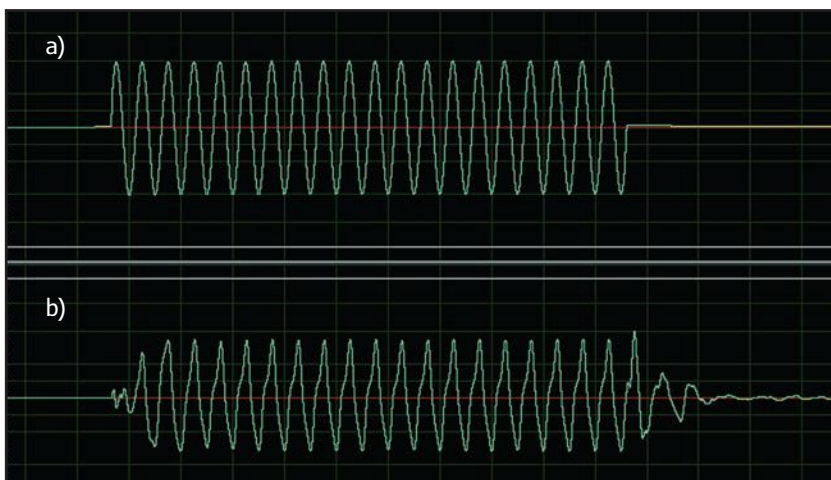


Figure 4a: This tone-burst test signal is part of a series of half-second, 40-Hz sine-wave bursts separated by approximately half-second periods of silence. 4b: The series of tone bursts was played and recorded in the listening room.

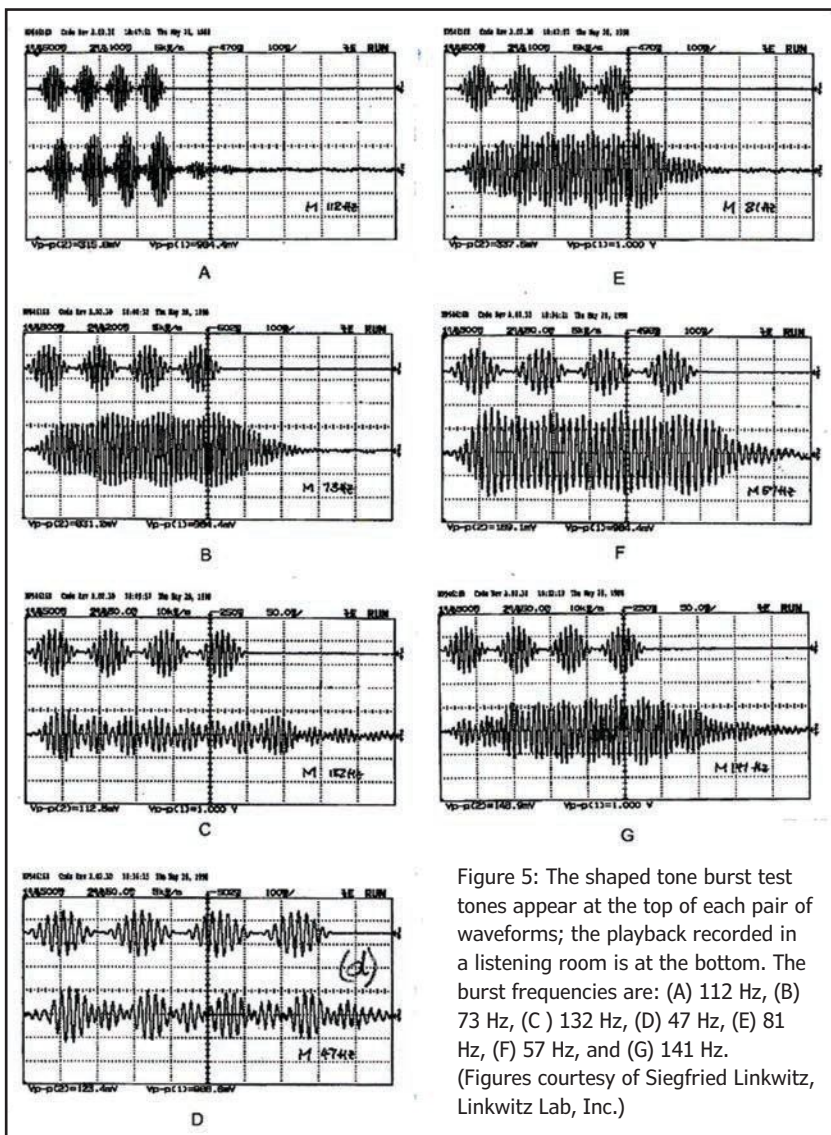


Figure 5: The shaped tone burst test tones appear at the top of each pair of waveforms; the playback recorded in a listening room is at the bottom. The burst frequencies are: (A) 112 Hz, (B) 73 Hz, (C) 132 Hz, (D) 47 Hz, (E) 81 Hz, (F) 57 Hz, and (G) 141 Hz. (Figures courtesy of Siegfried Linkwitz, Linkwitz Lab, Inc.)

Naturally, the first attempts at taming acoustically unruly small rooms used tools already in the toolbox: reverberation control by absorption, sound diffusion by means of barrel diffusers, and later, equalization of the sound system. These methods provided some improvement. In particular, boxiness can be reduced by placing acoustical absorption on offending surfaces. However, even though modes can cause boominess, they also cause articulation problems. Applying a fix in the frequency domain via equalization will not correct problems in the time domain.

At any modal frequency, not only is the level of the sound increased by the modal effect, but the effective RT at that frequency is increased. This is particularly true for pronounced modes having very little damping (acoustical absorption). Such a mode is characterized by a high quality factor or “Q.” Quality factor is the ratio of energy stored to energy dissipated per cycle. Thus, a high Q mode stores a lot of energy in each cycle at the modal frequency. This energy is then released into the room, lengthening the reverberant decay, which in turn muddles articulation. So while a notch filter can flatten the frequency response of the room at low frequencies, it does nothing to improve articulation.

Measuring Room Effects

One of the earliest tests tied in an attempt to measure the room’s effect on articulation was the low-frequency square-wave test. The results were not always enlightening, because no speaker can truly reproduce a square wave, although an excellent low-frequency horn comes closest. Next, engineers realized that a series of low-frequency tone bursts (see **Figure 4a**) would more nearly simulate musical articulation, and would be a test signal that a good speaker system could inject into the room. **Figure 4b** shows a series of 40-Hz tone bursts recorded in the room whose impulse response was shown in **Figure 1**. Notice that the tone-burst test shows that the pulse train takes a few cycles to build to full amplitude, and then takes a longer time to decay after the pulse signal ends.

In 1980, Siegfried Linkwitz reported on investigations using shaped five-cycle tone bursts with a raised-cosine envelope.^[3] **Figure 5** shows shaped tone-burst tests with the frequencies at various proximities to resonant modes. The effect of the modes in extending the decay—filling in the spaces between bursts—can clearly be seen. **Figure 5a** has a frequency far from any modes. **Figures 5b, 5f, and 5g** show the frequencies of response peaks. **Figures 5c, 5d, and 5e** show the frequencies of response notches.

Figure 6 shows the subwoofer signal of an anechoic speech track, low-passed at 75 Hz, with the same track played and recorded in a room having a 1.5-second RT. Notice that, despite the widely held notion that all transient energy resides in high frequencies, there is substantial low-frequency energy in this signal. Reverberation smears it just as with a full-spectrum signal. For more information on shaped tone burst testing visit Linkwitz Lab (www.linkwitzlab.com/burst-cd.htm).

MATT Testing

Although tone-burst testing can reveal articulation degradation by room effects, evaluating the results takes some understanding and experience. Also, a single-frequency tone-burst test provides limited information. Around 1990, Art Noxon of Acoustic Sciences Corp. extended the MTF testing method, using a test tone composed of a swept sine wave gated at 8 Hz with a 50% duty cycle (1/16 second of tone burst followed by 1/16 second of silence). The frequency is swept from 28 Hz to 780 Hz and then back down to 28 Hz (see **Figure 7** and **Figure 8**). This is called the Musical Articulation Test Tone (MATT) signal. It is available for download at www.acousticsscience.com/matt.

Angelo Farina, et al., published a paper on Acoustical Quality Testing (AQT) in 2001.^[4] The procedure included MATT testing. It seems to have been intended primarily as a procedure for determining audio quality in automobiles. In 2006, B. M. Fazenda published a paper on using the MTF to measure room acoustics performance.^[5] The MATT signal can be analyzed using specific software, but it also produces specific sound signatures that can be identified by listeners. Quoting the developer:

“During room playback, a number of different effects will be audible.

Ta-Ta-Ta-Ta, the sound of an articulate group of tone bursts. Usually there will be some 8 to 10 clean bursts in such a group, lasting about one second. A typical room will have only a few articulate groups of signals in the 75 second test.

Tattle-Tattle-Tattle-Tattle, the tell-tale sound of the room’s double-tongue response. Large spans of the track will have this sound. Notice that the tonal pulse rate is really twice that of the real signal. Too much energy occupies the dwell period of the test signal.

Toodle-oodle-oodle-oodle, the sound of the garbled room. Notice that it is a softer, less impacted sound. It’s close to a slurred double-tongue response.



Figure 6: The anechoic subwoofer signal for a speech track (a) can be compared with the same track played and recorded in a room having a 1.5-second RT (b).

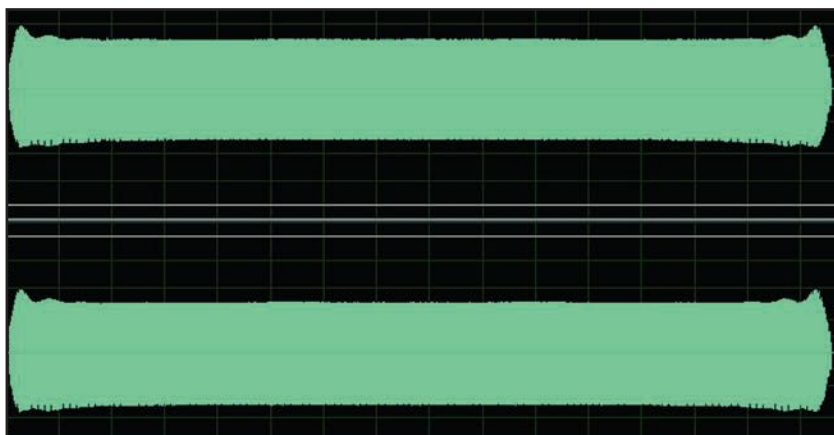


Figure 7: The complete MATT test signal is shown for both left and right channels.

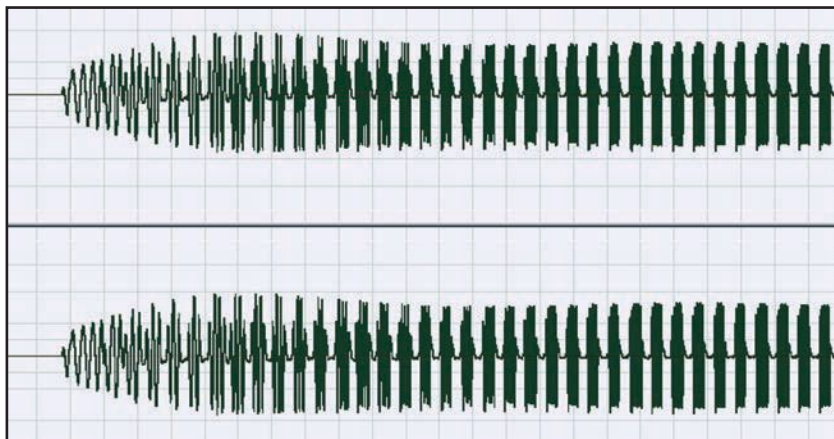


Figure 8: By zooming in on the time axis, we can clearly see the tone bursts and silent periods.

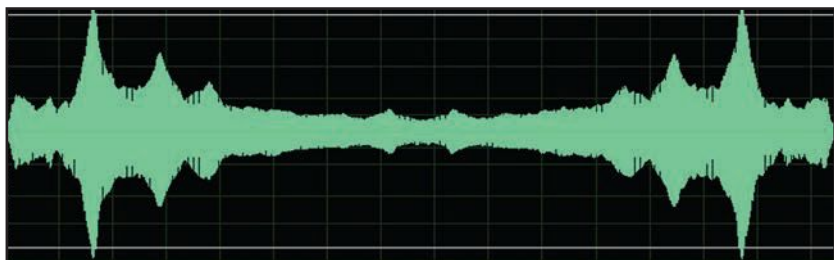


Figure 9: The MATT signal was played back and recorded in the listening room.

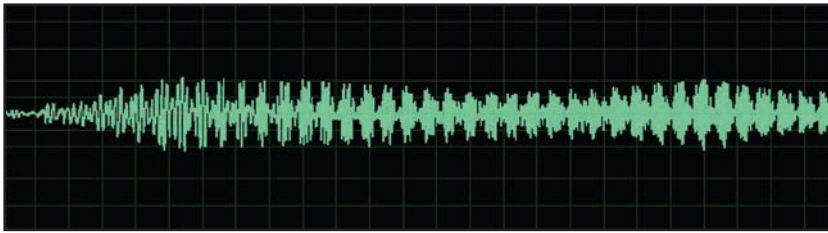


Figure 10: Expanding the first five seconds enables us to see the room's effects on the individual tone bursts.

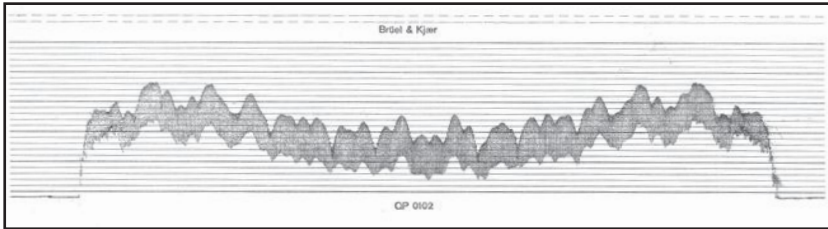


Figure 11: The recorded-in-room MATT response, after post-processing, shows frequency and amplitude more clearly.

Freq =	Full Band	Hz
RT60 =	0.243	sec, Reverb Time
EDT =	0.024	sec, Early Decay Time
t(l) =	0.0542	sec, Initial-Time-Delay Gap
C80 =	26.469	dB, Objective Clarity (80ms)
C50 =	17.601	dB, Objective Clarity (50ms)
G =	0.058	dB, Strength of Arriving Energy
ALC =	1.455	%, Articulation Loss Consonants
STI-N =	0.879	Speech Transmission Index (noise free)

Figure 12: These are the acoustic parameters calculated for the listening room, based upon the measured impulse response.

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Resources

Acoustic Sciences Corp., www.acousticssciences.com
 Linkwitz Lab, www.linkwitzlab.com/burst-cd.htm.

Tathump-Tathump-Tathump, is a more accurate presentation of the TA-TA. The "thump" is the turn-on and turn-off transient effects. This subtle transient coloration becomes totally inaudible with anything but articulate room playback. The thump is a damped 45 Hz ringing with only two oscillations of presence following each burst transition."^[6]

Figure 9 and **Figure 10** show the MATT signal played and recorded in the same room used for the tone-burst tests shown in **Figure 4**. The envelope of the recorded signal (see **Figure 9**) shows the low-frequency response of this unequalized room. In the expanded view (see **Figure 10**), it is difficult to distinguish the tone bursts until about 1 second into the test signal, corresponding to about 33 Hz in the sine sweep. **Figure 11** shows the recorded-in-room MATT signal after post-processing by Acoustic Sciences Corp. to clarify the frequencies and amplitudes within the wave. More quantitative test results can be had by submitting a recording of the MATT playback in the user's listening room and paying a small analysis fee to Acoustic Sciences Corp.

Just for comparison, **Figure 12** shows the clarity, definition, %Alcons, STI, and other acoustical parameters for the listening room. The aural impression from the MATT playback was "Ta-ta-ta," indicating good articulation in this room, as supported by the very high C80 and STI scores. It is important to note, though, that C80 and STI are full-bandwidth calculations; whereas the MATT test targets tonal frequency effects, especially including the low-frequency (subwoofer) range in the room.

Among the test CDs on my shelf, there are several signals for testing acoustical and electro-acoustical environments. These include:

- Sine sweeps
- Pink noise: wideband and narrowband (to 1/18 octave)
- Anechoic music recordings
- Square waves

Although each of these has valid uses, none is as effective for identifying low-frequency defects in small listening rooms as is the MATT test. All of the others also require recording and/or post-analysis of the test signal reproduction to yield meaningful information. MATT stands alone in its ability to yield repeatable results through subjective evaluation.

Author's Note: I would like to thank Art Noxon of Acoustic Sciences Corp. for his assistance in researching and preparing this article.



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Current Feedback and Voltage Feedback Fallacies

Discover the math and science behind this author's assertion that the classifications for current feedback and voltage feedback are inaccurate.

$$AV \cong \frac{R_{EQ} (R_1 + R_2)}{s \left(C_{EQ} R_{EQ} R_1 R_2 + C_{EQ} R_{EQ} R_1 r_E + C_{EQ} R_{EQ} R_2 r_E \right) + R_1 R_2 +}$$

By
Michael Kiwanuka
(United Kingdom)

It is presently almost universally asserted that amplifiers may be classified into two main groups: "current feedback" or "voltage feedback," allegedly depending on whether they respond to current inputs or voltage inputs, respectively.^[1] This classification is misconceived and did not exist 35 years ago for sound technical reasons.

Feedback Amplifier Configurations

Regarding the so-called "current feedback" amplifier topology of **Figure 1**, it is suggested by Sergio Franco in his book *Design with Operational Amplifiers and Analog Integrated Circuits* that the current sunk by the negative feedback network from the inverting input of the amplifier is equal to the current sourced into the impedance at the output of the complementary current mirrors so that:

$$v_{OUT} = i_N \left(\frac{R_{EQ}}{1 + sC_{EQ} R_{EQ}} \right) \quad (1)$$

Franco infers from this equation that the error signal driving the amplifier's forward-path is the current i_N and states that the impedance is the amplifier's "open-loop transimpedance gain."

Consequently, he concludes that "current feedback amplifiers" may also be called "transimpedance amplifiers." Franco is far from alone in espousing these views. Derek Bowers also discussed it in Chapter 16 of *Analog IC Design: The Current-Mode Approach*. But, they are, nevertheless, fatally flawed. To see why, a clear appreciation of what really constitutes a current feedback amplifier, a voltage feedback amplifier, and a transimpedance amplifier is required.

There are only four forms of single major-loop electronic negative feedback in existence:^[2]

1. Shunt (Current) Applied, Shunt (Voltage) Derived Negative Feedback—In **Figure 2**, the feedback transfer function is a transadmittance, giving rise to a transimpedance amplifier or a transadmittance feedback amplifier.
2. Shunt (Current) Applied, Series (Current) Derived Negative Feedback—In **Figure 3**, the feedback transfer function is a current ratio giving rise to a current feedback amplifier or, simply, a current amplifier.
3. Series (Voltage) Applied, Series (Current) Derived Negative Feedback—With **Figure 4**, the feedback transfer function is a transimpedance,

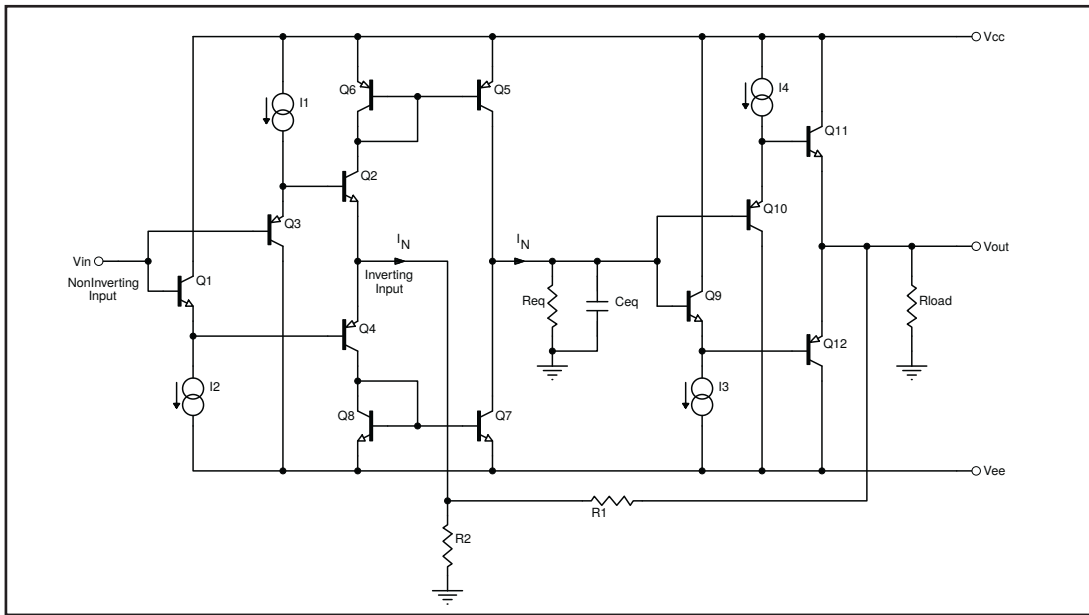


Figure 1: This circuit is almost universally called a “current feedback” amplifier; however, it is simply a voltage amplifier.

which gives a transadmittance amplifier or a transimpedance feedback amplifier.

4. Series (Voltage) Applied, Shunt (Voltage) Derived Negative Feedback—The feedback transfer function shown in **Figure 5** is a voltage ratio resulting in a voltage feedback amplifier or, alternatively, a voltage amplifier.

From the above definitions, it is apparent that the type of amplifier obtained after the application of single major-loop negative feedback is determined by the feedback network’s transfer function and not by the topology of the amplifier’s forward-path or the current drawn by the feedback network from the amplifier’s inverting input in the case of the circuit shown in **Figure 1**.

It is also self-evident that the term “current feedback” only applies to an amplifier in which the negative feedback transfer function is a current ratio—that is, one in which the negative feedback is shunt applied and series derived, giving a current amplifier or current-controlled current source.

Additionally, the amplifier shown in **Figure 1** is a voltage amplifier and not a transimpedance amplifier because its negative feedback transfer function is a voltage ratio (that is, one in which the feedback is series applied and shunt derived by means of a simple voltage divider) irrespective of the topology of the amplifier or of the loading of the feedback summing node (the inverting input) on the feedback network.

Greater insight into why the circuit shown in **Figure 1** really isn’t a current feedback amplifier can be gleaned by consideration of the fact that, in the first instance, the current flowing in the negative feedback network for a given output voltage is not a function of the current flowing through the amplifier’s load. In other words, the load current is not controlled by the negative feedback, as it

should with a current feedback amplifier. Instead, the negative feedback network in the circuit shown in **Figure 1** is a voltage divider draped across the output of the amplifier which implies, clearly, that it is the output voltage that is sampled by the divider which then applies a fraction of that output voltage to the inverting input of the amplifier. The current flowing through the negative feedback voltage divider is, therefore, completely irrelevant.

A transimpedance amplifier, on the other hand, possesses transadmittance feedback (that is, an amplifier with shunt applied and shunt derived negative feedback) and is, in fact, a current-controlled voltage source. The circuit shown in **Figure 1** is, therefore, most certainly not a transimpedance amplifier, regardless of its assumed internal workings.

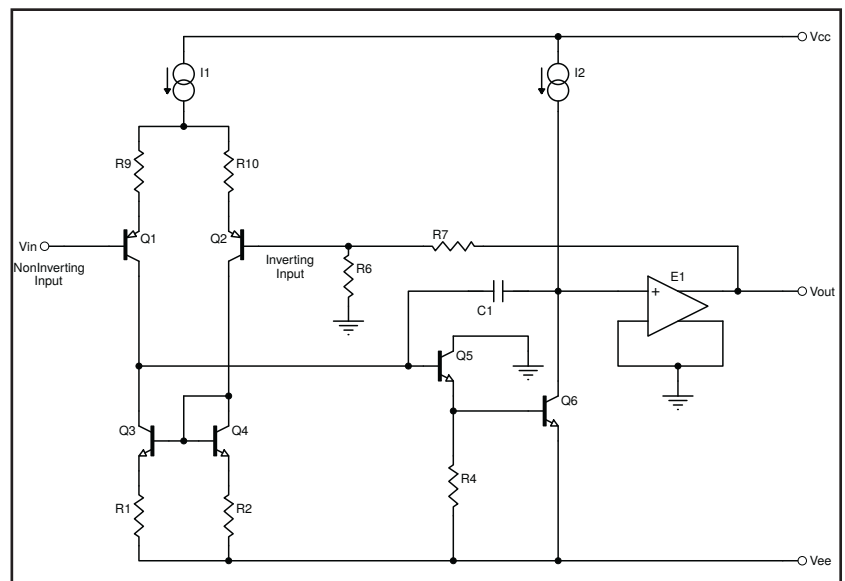


Figure 2: Transadmittance feedback gives rise to a transimpedance amplifier or a current-controlled voltage source (CCVS).

How So-called “Current Feedback” and “Voltage Feedback” Amplifiers Really Work

The critical error in Franco’s **Equation 1** is that it does not account for the amplification occasioned by the complementary pair of input transistors Q2 and Q4 (see **Figure 1**), which operate in common-emitter mode with respect to the amplifier’s forward-path and common-base mode as far as the loop-transmission path is concerned. In both modes, it is the voltage gain of these transistors that is relevant, irrespective of the fact that significant current flows between the inverting input and the feedback network due to the loading of the former on the latter.

Since the complementary common-emitter pair Q2 and Q4 is biased in Class-B (with the transistors forward-biased alternately) courtesy of emitter followers Q1 and Q3, the forward-path voltage gain of the amplifier with respect to its non-inverting input is that of a single common-emitter amplifier with $R_{EQ}/(1/sC_{EQ})$ as its load. The latter is simply divided by the sum of the parallel combination of the feedback resistors R1/R2 and the emitter intrinsic resistance r_E to obtain the approximate forward-path voltage gain A_V :

$$AV \cong \frac{R_{EQ}(R_1 + R_2)}{s(C_{EQ}R_{EQ}R_1R_2 + C_{EQ}R_{EQ}R_1r_E + C_{EQ}R_{EQ}R_2r_E) + R_1R_2 + R_1r_E + R_2r_E} \quad (2)$$

Figure 3: Current feedback gives rise to a current amplifier or a current-controlled current source (CCCS).

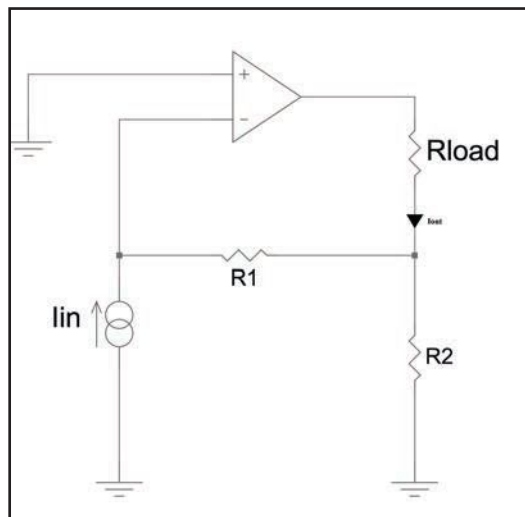
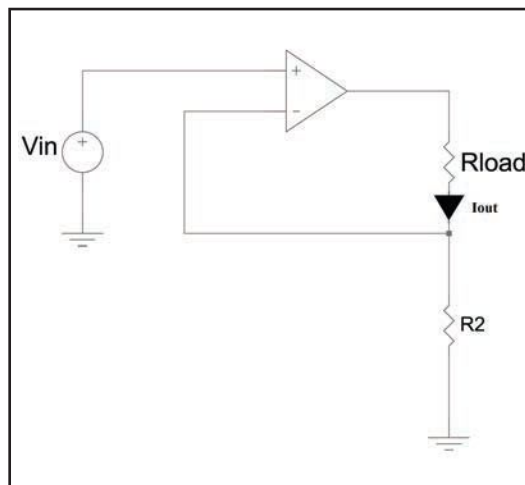


Figure 4: Transimpedance feedback gives rise to a transadmittance amplifier or a voltage-controlled current source (VCCS).



Note that the complementary current mirrors in **Figure 1** do not generate voltage gain but merely provide phase inversion and level shifting.

The method we used earlier to determine the forward-path gain of a voltage feedback amplifier in which the feedback summing node (the inverting input) of the amplifier significantly loads the feedback voltage divider is ably explicated by D. H. Horrocks in his book *Feedback Circuits and Op Amps*. The loading effects of the feedback network on the output of the amplifier are here deemed negligible.

As indicated above, the fact that the complementary common-emitter pair in the circuit shown in **Figure 1** source significant current to the negative feedback network is merely indicative of the loading of the amplifier’s inverting input on the feedback network, and it certainly does not imply that the circuit is a current feedback amplifier.

The feedback voltage divider places a fraction of the amplifier’s output voltage at the inverting input where it is then subtracted from the input voltage at the non-inverting input to generate the error voltage which drives the amplifier’s forward-path. Therefore, contrary to opinions from Franco and from Walt Jung in his book *Op Amp Applications Handbook*, the negative feedback does not act to drive the current from the inverting input to zero because the error signal is a voltage and not a current. Indeed, the negative feedback does not act to drive this error voltage to zero either because, in a physically realisable amplifier, a finite error voltage is required to drive the amplifier’s forward-path in order to cause it to generate the demanded output voltage. This is a common misconception. Therefore, to reiterate this important point, the amplifier from **Figure 1** responds to voltages rather than currents at both its inverting and non-inverting inputs, and, consequently, the error signal driving its forward-path is a voltage and not a current.

This is also true of the topology shown in **Figure 6**, which is erroneously and almost universally referred to as a “voltage feedback amplifier” even when its feedback connections do not justify it. Of course, it is evident that in this specific case, the circuit shown in **Figure 6** is, in fact, a voltage feedback amplifier because it possesses series (voltage) applied, shunt (voltage) derived negative feedback, and, therefore, its feedback transfer function is a voltage ratio. But, crucially, this will not be true if either of the other three forms of negative feedback is applied

instead. Clearly, in the absence of major loop negative feedback the circuits shown in **Figure 1** and **Figure 6** are simply voltage amplifiers.

One of the most important functions of major loop negative feedback is to convert a basic voltage amplifier into one of the following configurations: a voltage-controlled voltage source (VCVS), a voltage-controlled current source (VCCS), a current-controlled current source (CCCS), or a current-controlled voltage source (CCVS). This is achieved by the negative feedback modifying the amplifier's input and output impedances, with shunt negative feedback connections reducing the impedance, while series negative feedback connections increase the impedance. Note, however, that the low impedance at the inverting input of the voltage amplifier of **Figure 1** broadly precludes its use in inverting applications with shunt applied negative feedback; such applications require a voltage amplifier with a differential-pair input stage as used in the circuit shown in **Figure 6**.

A significant difference between the circuits shown in **Figure 1** and **Figure 6** is in the

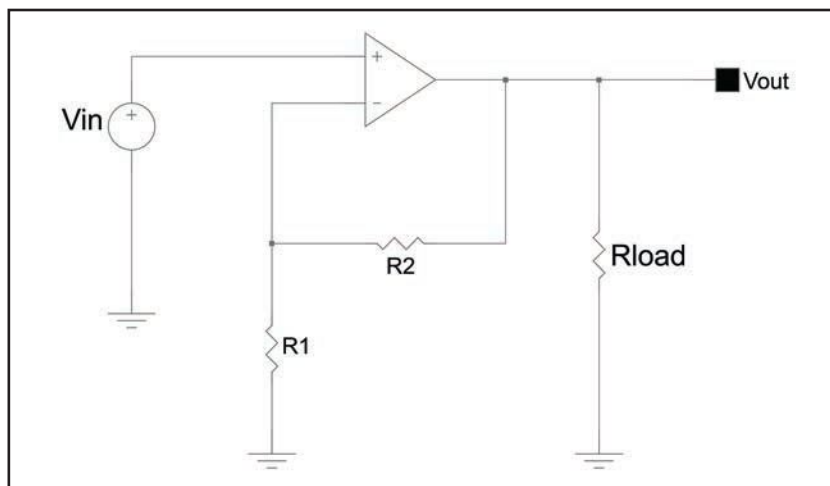


Figure 5: Voltage feedback gives rise to a voltage amplifier or a voltage-controlled voltage source (VCVS).

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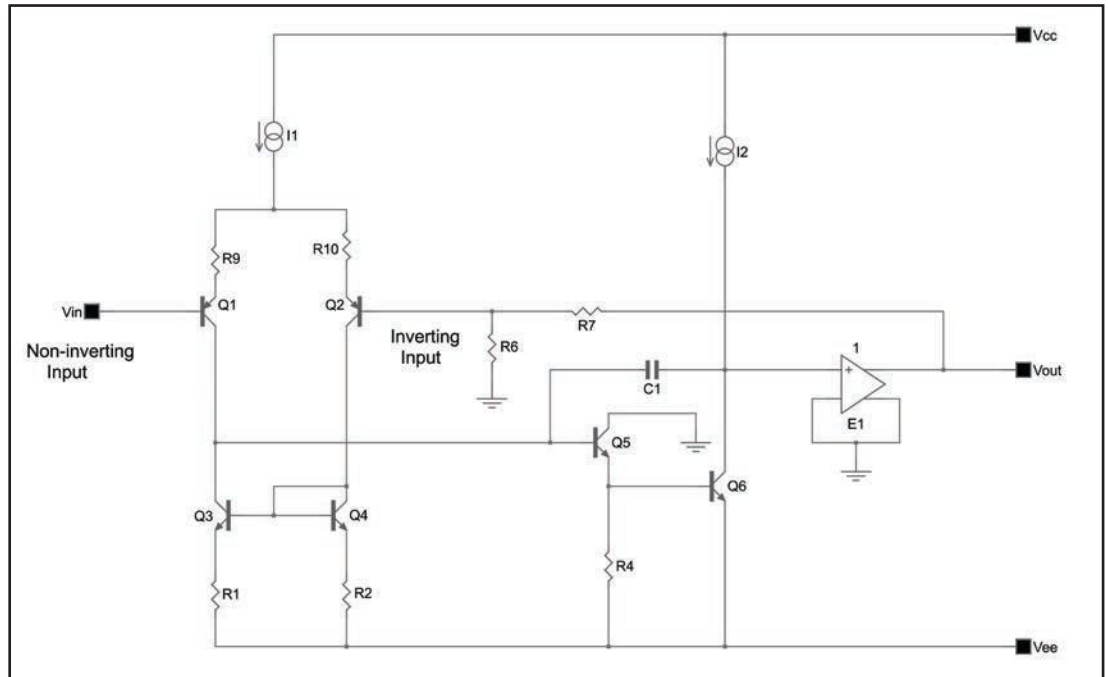
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Figure 6: The rudiments of a voltage amplifier of the Thompson topology; the input stage is a differential transadmittance stage (TAS) while the second stage is a transimpedance stage (TIS). The unity-gain voltage-controlled voltage source E1 represents the output stage of the amplifier, and this is usually a complementary Class-B emitter-follower arrangement.



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[1] S. Franco, *Design with Operational Amplifiers and Analog Integrated Circuits*, pp. 315-321, 4th edition, McGraw-Hill Education, 2014.

[2] T. M. Frederiksen, *Intuitive Operational Amplifiers: From Basics to Useful Applications*, pg. 80-85, McGraw-Hill, 1988.

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M. Kiwanuka, "Variations on the Complementary Folded Cascode Transimpedance Stage in Discrete Audio Frequency Power Amplifiers," *Electronics World*, November 2013.

A. C. Russell, "CFA vs VFA: A Short Primer for the Uninitiated," hifisonix.com, January 2014, <http://hifisonix.com/wordpress/wp-content/uploads/2014/01/CFA-vs-classic-Lin-VFA-topology.pdf>.

amplifier of **Figure 6**—the feedback network is buffered from the feedback summing node by the emitter follower Q2 that is part of the input differential pair. Thus, the forward-path gain and forward-path dominant pole of the circuit shown in **Figure 6** do not vary significantly with changes in the values of the feedback network. The workings of the circuit shown in **Figure 6** are further elucidated in an article I wrote called "Variations on the Complementary Folded Cascode Transimpedance State in Discrete Audio Frequency Power Amplifiers" (*Electronics World*, 2013).

In contrast, and in accordance with **Equation 2**, the forward-path dominant pole and forward-path gain of the so-called "current feedback" amplifier of **Figure 1** vary appreciably with changes in the values of the components comprising the feedback network. This is due to the voltage coupling factor between the feedback network and the conjoined emitters of the complementary common-emitter input stage being much less than unity. The poor voltage coupling factor is simply another way of saying the amplifier's inverting input severely loads the feedback network. The compromised voltage coupling factor also severely reduces major-loop gain in the voltage amplifier of **Figure 1** compared to that obtainable with the conventional circuit shown in **Figure 6** for the same feedback network component values. Major-loop transmission is not helped by the fact that the execrable circuit shown in **Figure 1** possesses only one common-emitter gain-generating stage

in its forward-path compared to two stages in the arrangement shown in **Figure 6**, which consists of a transadmittance stage driving a transimpedance stage.

Therefore, contrary to the comments by Andrew C. Russell in his article "CFA vs VFA: A Short Primer for the Uninitiated," the voltage amplifier of **Figure 1** is of no use whatsoever in audio frequency applications because its significantly low major loop gain (compared with that generated by the circuit shown in **Figure 6**) is insufficient to satisfactorily mitigate the non-linearity generated by its forward path. The performance of the circuit shown in **Figure 1** is further degraded by the fact that its complementary common-emitter input stage operates in Class-B. It's bad enough that crossover distortion arising from Class-B operation has to be tolerated in the output stage, but extending it to the input stage as well is downright perverse, at least as far as audio frequency applications are concerned. Although the push-pull Class-B operation of the complementary common-emitter input stage of

the circuit shown in **Figure 1** makes very high slew rates possible by supplying a relatively large amount of current to charge and discharge the capacitance at the output of the current mirrors, this is not a significant advantage in audio frequency applications.

Conclusion

The terms "current feedback amplifier" and "voltage feedback amplifier" as presently used are wholly unfounded. It has been demonstrated that the correct application of these terms is entirely dependent on the manner in which major loop feedback is applied around the forward-path of an amplifier, irrespective of the workings of its internal circuitry. 

About the Author

Michael Kiwanuka has had an abiding fascination for audio electronics since he was a teenager. This interest eventually led to the pursuit of a BTEC HNC in Electronic Engineering and a BSc (Honors) degree in Electronic Engineering. He is presently involved in the research and development of novel feedback error correction strategies for audio frequency power amplifiers.



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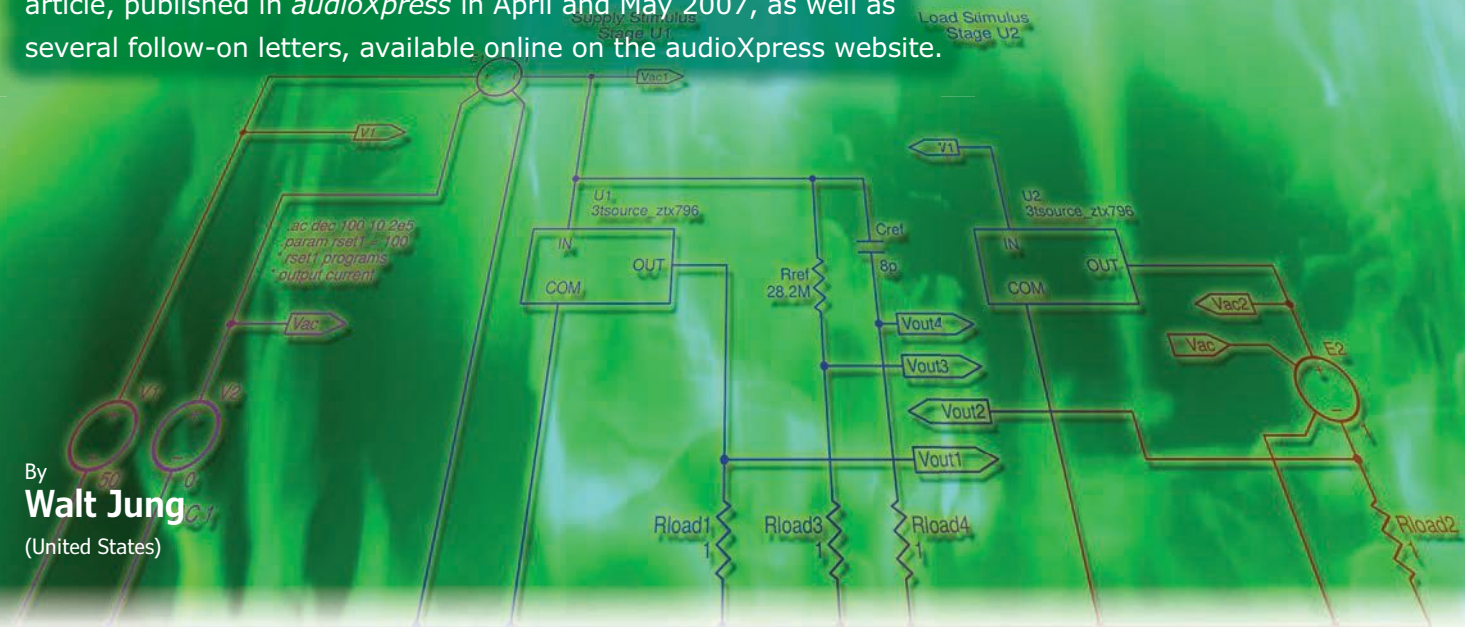
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A Sources 101 Update

This article by the author of the *IC Op Amp Cookbook* and *Op Amp Applications Handbook* is an update and revisit of his "Sources 101" article, published in *audioXpress* in April and May 2007, as well as several follow-on letters, available online on the audioXpress website.

By
Walt Jung
(United States)



This article is a update and revisit of my "Sources 101" *audioXpress* article of April 2007^[1] as well as several follow-on letters from September of that year.^[2, 3] It deals with the basic test methodology and the relative validity for some specific current source types. For readers new to these discussions, a re-reading of the above articles will be necessary.

But first, some semantic terms need resolution. "The Amplified Negative Feedback Current Source" (Figure 4 of [4]) has been used as a name for the same basic circuit as what I call a "One Vbe current source," (i.e., Figure 3A of [1]). In these discussions, we continue with that original name, abbreviated as 1VbeCS.

Two-Terminal and Three-Terminal Current Sources

These articles have dealt with *three-terminal* (3T) type current source circuits (see **Figure 1**) which have IN, COM, and OUT terminals. The

measurements are taken across a test load, Rload1 connected to OUT, with a DC bias plus an AC stimulus Vac applied to IN (see [1] again).

A *two-terminal* (2T) current source would not use a COM terminal, but would otherwise be the same (i.e., an N-channel JFET with G-S short, as in the J500 series).

As should be noted, this test setup can be reduced in simplest form to a three-element series circuit, as noted:

- (1) is the AC stimulus, Vac (with or without DC bias),
- (2) is the circuit under test or unknown impedance,
- (3) is the load, Rload1

Note also that a 1-2-3 element sequence arrangement vs. a 2-1-3 sequence will conduct the same current. The order of elements doesn't matter, since the current is the same in all elements.

At this point, Kirchoff's laws could be reviewed, should you so wish.

These 2T series circuit principles allow a measured voltage across (3) to deduce the relative Z of an unknown (2), with Vac being fixed. This may be more clear by referring back to the original **Figure 1** test circuit.^[1] The basic functionality should also be evident from **Figure 3**, a 3T SPICE test circuit with a source of (1) Vac (superimposed on a 50V DC bias V1), (2) a test current source circuit U1, and (3) Rload1. A 2T circuit would be connected between IN and OUT, with no connection to COM.

For a 2T type current source being tested, the measured voltage Vout can be related inversely to impedance. That is, a reading of -140 dB re a Vac level of 1 V implies an impedance of 10 meg Ω , -120 dB ==> 1 meg Ω , etc. For example, from the data of **Figure 2**, the lowest frequencies realize measured levels around -140dB, equivalent to around 10 meg Ω . While this is a bit fuzzy due to the setup noise floor, the basic idea holds.

Note that this data is for an actual real 3T circuit, that of **Figure 1** (which is *not* 2T), and there is a dramatic difference between the case of C1 being present, or not. What is going on here, and why?

Before answering this, it should be acknowledged that the above measurement techniques have been challenged, specifically for such 1VbeCS types.^[4] The stated objection is that a measured power supply rejection (PSRR) of X dB cannot be used to infer equivalent current source output impedance for a circuit such as **Figure 1**. The next few graphics will shed some very useful light on this point.

But first, it surely can be said the inverse relationship between relative decibel level and Z is specifically so, *for 2T unknown types*. Examples here are simple JFET current sources, or passive parts such as resistors and capacitors. In point of fact, the lab setup has been calibrated with resistances up to 1 gig Ω and capacitances down to 2 pF.

It has been consistently observed that 2T Rs and Cs measure just what they should. So this suggests that true 2T active current sources in general should also measure well, as shown previously.^[4] This assumes proper active-circuit biasing, of course.

But, a potential discrepancy here lies within the nature of exactly how a 3T source behaves, versus a 2T type, for otherwise similar measurement conditions. As was noted, a 3T source has a COM pin, by definition. This basic fact has a tricky potential of making the current flow from IN to OUT depart from that of a 2T source. This can occur in some instances, *if the 3T COM pin current isn't appropriately accounted for within the design*.

As a result, 3T current source type impedance

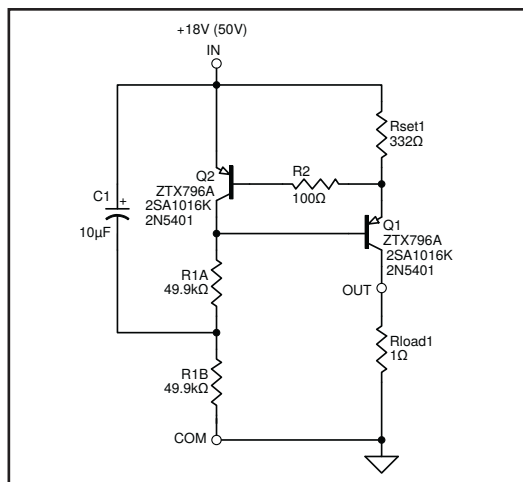


Figure 1: This one Vbe current source uses a bootstrapped capacitor C1. (Subcircuit Nodes IN, COM, OUT)

measurements, unlike 2T measurements, should not be casually related between relative decibel levels and an inversely related impedance. While a reading of "X dB" may be useful as a power supply rejection ratio (PSRR) figure-of-merit for a given circuit, that X dB won't necessarily be related to the circuit's true output impedance (i.e., the impedance as measured load-side).

This point was not covered originally, and I apologize for any confusion this may have caused. In fact, the working relationship between dB of rejection and a 3T source's true output impedance between is actually *conditional*. The uncertainty makes thing decidedly more complex. In fact, the

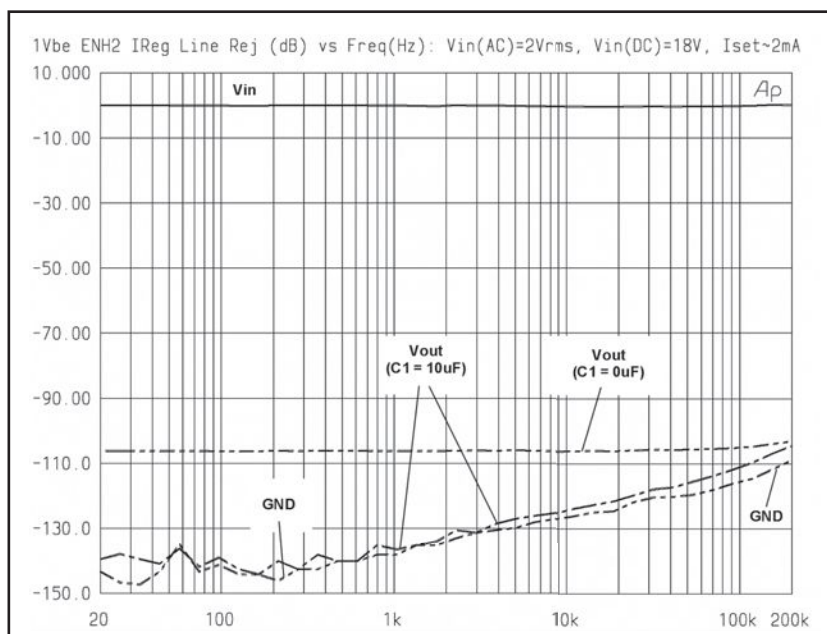
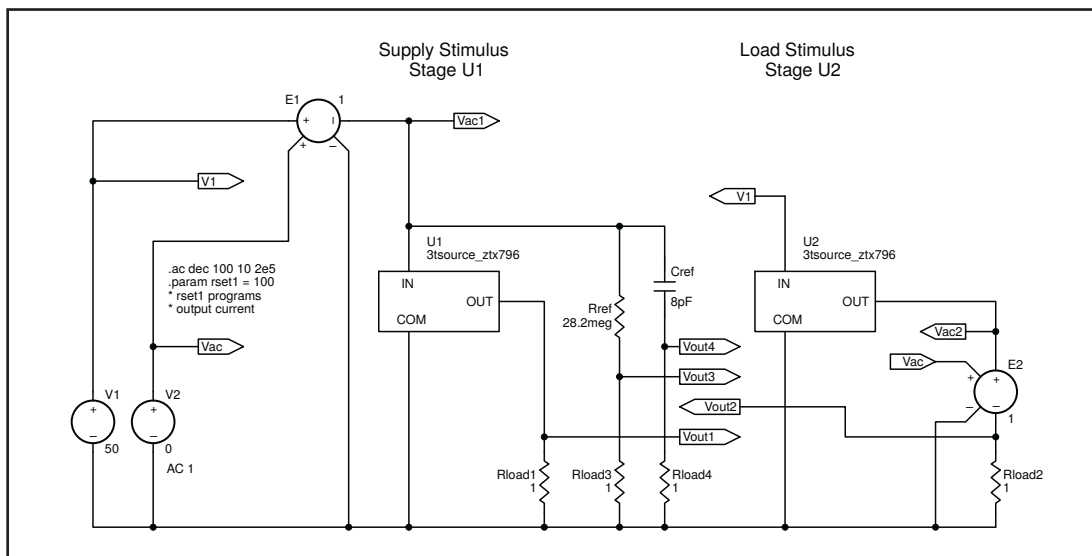


Figure 2: The lab performance of the Figure 1 circuit is greatly improved using C1. (The data was taken at 18 V/2 mA with 2SA1016Ks.)

Figure 3: A SPICE simulation circuit with 1 VAC stimulus is applied to dual Figure 1 current sources, supply side (U1) and load side (U2), with ZTX796A transistors in a one Vbe ~5 mA setup.



criticism of [4] is actually both right and wrong, with prevalence being dependent upon the exact context of the configuration.

Practically speaking, this means that one must consider the internals of a 3T current source circuit, to see exactly how it behaves. The 1VbeCS in question is a good example for explanation, as it shows both 3T and 2T behavior!

Thomas Bohley, Chris Paul and John Popelish commented on this original current source circuit and the substandard performance.^[2] I agreed, basically, but also noted a key letter from Marc Whitney.^[3] Whitney suggested looking into Doug Self's form of the 1VbeCS current source.^[5] This connection uses a bootstrap capacitor, and split

resistor biasing. This step effectively immunizes the gain stage against supply variations (more on this later).

This scheme is shown in **Figure 1** (same as the circuit in [3]), in the form of a "Tee" circuit. C1 is a bootstrap coupling capacitor, and the total R1a plus R1b $\approx 100 \text{ k}\Omega$ value provides a Q2 current just like the (value-modified) original circuit (i.e., $160 \mu\text{A}$) when operated on 18V. As the **Figure 2** performance plots indicate, this circuit really does have truly excellent line rejection, when using the $10 \mu\text{F}$ C1. The rejection at low frequencies is increased 35 dB or more, to a range of $\approx 140 \text{ dB}$, actually limited by the test setup. Above 100 kHz, more than 110 dB of rejection is available. This improved performance is had for one additional R and C, vis-à-vis the original version.

With the Tee circuit active, the 1VbeCS behaves as if the bias voltage across C1 is constant with varying input voltage. Take this point to a logical conclusion, and note that if one were to replace C1 with a $\sim 25 \text{ V}$ battery and remove R1b, the circuit actually becomes a 2T type! And, as such, it has similar properties for measurements. Understanding this chameleon-like property is a major key to using the 1VbeCS most effectively.

But, some caveats still do exist. The Tee circuit as applied to this or any of the other current source types (such as LED or diode biased) *will begin to lose effectiveness at low frequencies, or when C1 no longer exhibits low Z vis-à-vis the R1 resistances.*

For the example C1/R1 values, this degradation isn't readily apparent from the lab tests, but it could still be a potential problem for other conditions. Here, the goal was a low cap value, thus the lower -C/higher-R values. C1 can be a general purpose

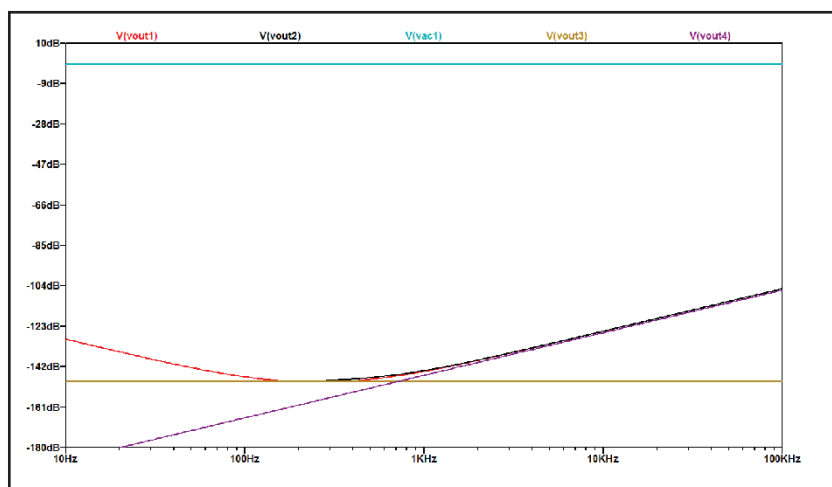


Figure 4: These are the SPICE simulation results for Figure 3. Supply Side Stimulus = Vout1, red < 200 Hz, black > 200 Hz Load Side Stimulus = Vout2, black > 200 Hz, brown < 200 Hz ~28 meg Ω Reference Resistance = Vout3, Rref = brown 8 pF Reference Capacitance = Vout4, Cref = violet

type, but it should be rated for a voltage about equal to the supply rail used. This will bias it conservatively (it sees only $\approx V_s/2$). A very important point is that the scheme also applies to other current source types, as was noted in [3]:

It is worth noting that this technique also works with other current sources of this type, among these are the “Two-diode” and “LED” variants discussed in Figure 4A and Figure 5A of [1]. The key step is to split bias resistor R1 into two equal parts, and apply the coupling cap to the midpoint.

Some performance examples of this type of source are noted within **Figure 7** and **Figure 8**.

The One Vbe Current Source: Revisited and Simulated for Both Supply-Side and Load-Side Impedance

Yogi Berra is remembered for numerous colloquialisms. Among them is: “You can observe a lot by watching.”^[6] For “observe” we take the Dictionary.com usage # 2: *To regard with attention, especially so as to see or learn something*. Of course, one definitely has to actively watch, in order to absorb/learn from a given experience.

A first case in point is the critical discussions of [4], which do not cite (and thus, do not observe) the specific points just reiterated (i.e., *suggesting use of the Tee circuit with three different current source types, from September of 2007*).^[2, 3]

A second case in point is the simulated data of [4] as associated within **Figures 2, 3, and 4**, all of which focus on impedance *as measured at the load-side of the current source*. If we take the same circuit and also apply a stimulus *supply-side*, some very useful additional information is obtained. In fact, this information is critical to a fuller understanding.

Figure 3 is an LTSpice test circuit which exercises the 1VbeCS of **Figure 1**, using the ZTX796A transistors with the Tee circuit values shown, at a ~ 5 mA current, on a 50 V supply. The SPICE circuit uses test conditions shown in **Sidebar 1**. The simulations use U1 and U2 as a common subcircuit, noted in **Sidebar 2** (see References).

Note that for stage U1, the AC stimulus is applied *supply-side*, as Vac1, while for U2 it is applied *load-side* as Vac2 (as in [4]). Both simulation results are shown in **Figure 4**, and they are displayed similar to the lab test results for the same circuit as operated on 18 V (see **Figure 2**). As will be noted, the supply-side Z data (Vout1) and load-side Z data (Vout2) are overlaid above about 200 Hz. *In other words, they are essentially identical*. Below 200 Hz, the Tee circuit values begin to lose effectiveness, and the impedance seen from Vout1 begins to fall,

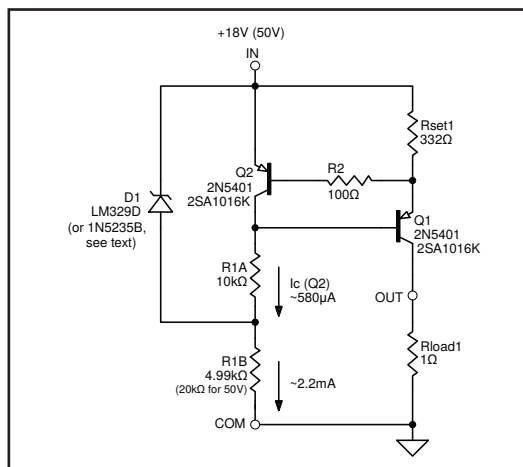


Figure 5: The Bohley one Vbe current source uses D1 Zener diode stabilization.

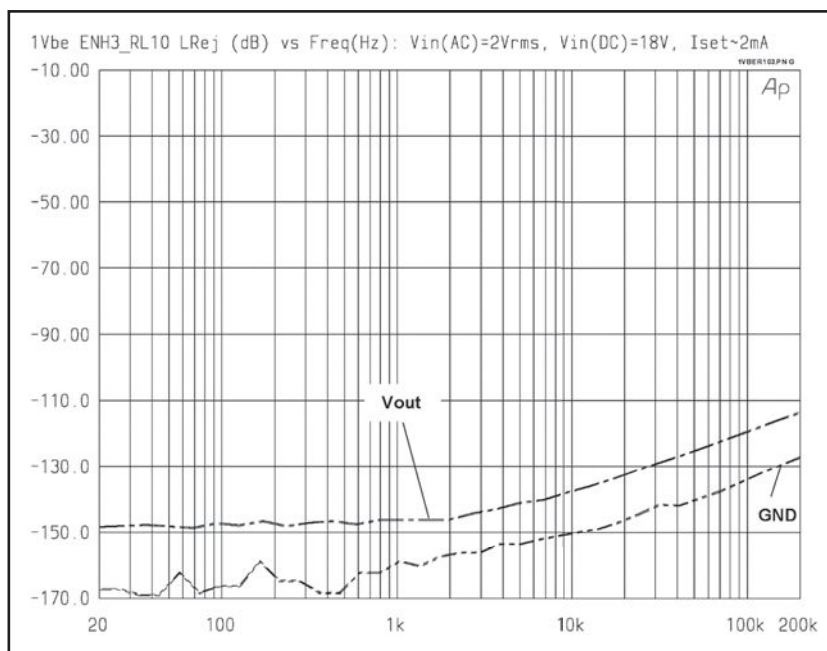


Figure 6: The lab performance of Figure 5 is essentially frequency independent below 2 kHz. The higher resolution setup allows better-performing circuits to be fully revealed. (The data was taken at 18 V/2 mA with 2SA1016ks.)

About the Author

Walt Jung has been a writer on analog design topics since the late 1960s, specializing in op-amp and audio-related applications. His first *Electronic Design* article appeared in 1968, and many others followed, including his analog-oriented column “Walt’s Tools and Tips.” He authored many audio articles for Audio Amateur publications, starting in the 1970s and continuing with *audioXpress* in more recent years. In addition to writing numerous audio-oriented articles, Walt authored several IC application books, namely three editions of *IC Op Amp Cookbook*, as well as *Audio IC Op Amp Applications*. Most recently, he edited a book for Analog Devices Incorporated (ADI). Called *Op Amp Applications*, this book was a feature of the company’s 2002 seminar tour. Walt has held senior level applications engineering positions with Linear Technology and ADI, and retired from ADI in 2002. He is a Life Fellow of the Audio Engineering Society and a Life Member of the Institute of Electrical and Electronics Engineers (IEEE).

represented by the rise of the curve. Below 200 Hz, the Vout2 curve flattens out to a -149 dB level, corresponding to around 28.2 meg Ω . This is roughly comparable to the data of [4] (but not exactly, due to slight operating differences). An additional data point is the circuit's measured capacitance of ~8 pF, as noted by the Cref curve, which is tangent to Vout2 at high frequencies.

So, what we see now is that with the incorporation of the supply-side stimulus data, we

actually now have a completely new performance dimension. In point of fact, this circuit behaves as if it is operating as a 2T type above ~200 Hz, and as a 3T type below, as evident in the rising Vout1 (red) trace. For other C1/R1 values the frequency inflection point will move. A larger C1 will move the frequency where Vout1 and Vout2 depart downward in terms of frequency.

Nevertheless, it is remarkable just how closely these two Vout1 and Vout2 traces do track, above the frequency inflection point. It would be fair to say that there is little or no practical difference between supply-side and load-side stimulus, over this range. Then it follows that either method of stimulus can be used to characterize impedance for this type of circuit (i.e., either supply-side or load-side).

What we can now learn from all of this is just what is behind the conditional nature of the circuit behavior with supply variations. With the Tee circuit active in this 1VbeCS, the performance is superior, without it, things are much worse. But to the point of the [4] criticism, as used within **Figure 1**, *the Tee circuit enables the measurement of both superior PSRR and high output impedance*. That is, the load-side measured Z is virtually indistinguishable from supply-side measured Z. This is because the use of Tee circuit makes this 1VbeCS behave as if it were a 2T source. The bottom line? *We've observed a lot here, just by watching!*

Sidebar 1: Notes Applicable to the SPICE Tests Shown in Figure 3

Vac1 from supply-stimulus buffer E1 = AC stimulus voltage for supply-stimulus stage U1 = 1v

Vac2 from load-stimulus buffer E2 = AC stimulus voltage for load-stimulus stage U2 = 1 V

Vout1 = AC output error voltage for supply-stimulus stage U1, in decibels relative 1 V

Vout2 = AC output error voltage for load-stimulus stage U2, in decibels relative 1 V

Vout3 = AC output error voltage of example; ~28.2 meg Ω Rref (-149 dB relative 1 V ==> ~28.2 meg Ω) Vout4 = AC output error voltage of example; 8 pF Cref (decibels relative to 1 V)

Sidebar 2: 3TSource_ztx796.sub

* This subcircuit produces a 1Vbe current source hookup with ZTX796A xstrs.
*

* Usage: Connect IN node to DC source, COM to ground, OUT to load.

* In top spice directive, program Iout with Rset1, as ".param rset1 = 100"

*

```
.subckt 3TSource_ZTX796 IN COM OUT
```

```
Q2 N008 N001 IN ZTX796A
```

```
Q1 OUT N008 N002 ZTX796A
```

```
R2 N002 N001 100
```

```
Rset1 IN N002 {rset1}
```

```
R1b N007 COM 49.9k
```

```
R1a N008 N007 49.9k
```

```
C1 IN N007 10µ V=63 Irms=135m Rser=1.6 Lser=0 mfg="Nichicon"  
pn="UPJ1J100" type="Al electrolytic"
```

```
.model NPN NPN
```

```
.model PNP PNP
```

```
.lib C:\Program Files (x86)\LTC\LTspiceIV\lib\cmp\standard.bjt
```

```
* insert Q1/Q2 model file here
```

```
* ZTX796A*ZETEX ZTX796A Spice Model v2.0 Last Revised 4/3/05
```

```
* available from http://www.diodes.com/search?type=0&value=ZTX796A
```

```
.ends
```

The Bohley One Vbe Current Source

In some email exchanges since his original 2007 *audioXpress* letter, Thomas Bohley and I discussed enhanced forms of this useful circuit that would have the same effect of raising the dynamic impedance of the R1a resistor, but without the use of the C1 electrolytic. The desire here is to both minimize component size and count, and to increase long-term reliability (not a feature of electrolytic caps). We agreed that a LM329 IC (a synthesized buried Zener type) functionally substituted in place of C1 would be a viable choice. Such ICs have very low dynamic impedances ($\leq 1\Omega$), well below that of standard Zeners at a similar current. This lends considerable value to the circuit here, as this impedance works against a relatively high value of R1b, thus tending to keep both terminals of D1 near the supply voltage for dynamic changes. It is this factor that enables the circuit to emulate a 2T circuit, even though it is a 3T type.

A LM329-based version is shown in the Bohley 1VbeCS, as depicted by **Figure 5**. Here R1a is selected for a Q2 current of about 570 μ A, and R1b for 2.2 mA, sufficient to keep D1 in its operating range at low voltages. R1a needn't change for other

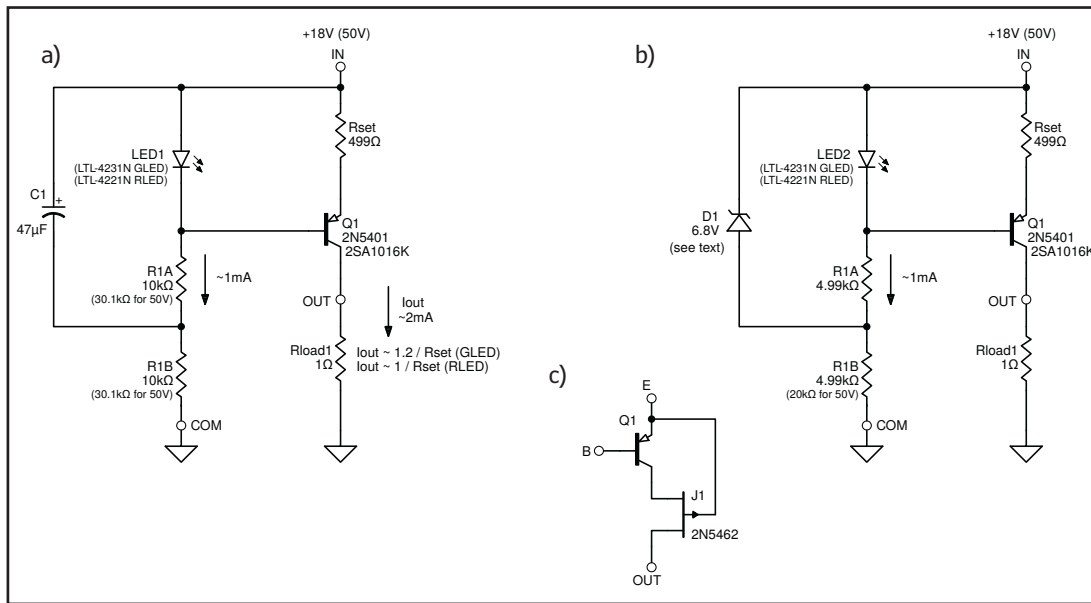


Figure 7: Two enhanced green or red LED-based current source(s)—with RC Tee circuit (a) or Bohley-style Zener Tee circuit (b). Note optional Csanky cascode (c) applies within either circuit.

supply voltages, but R1b must be changed, if the supply is substantially different. For a supply of 50 V, for example, R1b should be 20 kΩ.

This circuit was worthy of a test, to compare it to the 1VbeCS with the Tee circuit. And, it actually turned out better than expected, shown by the performance plots of **Figure 6**. These lab tests were done using 2SA1016K transistors, at 18 V and 2 mA (directly comparable to **Figure 1**), and D1 as an LM329. *Note the extended ~170 dB dynamic range, by virtue of an improved setup, allowing greater clarity for low levels.*^[8]

The low-frequency rejection of this circuit is as good or better as the **Figure 1** circuit, as there is simply no low-frequency roll off below 2 kHz, as with the Tee circuit. And, the high-frequency performance is even better. A low-frequency rejection of ~150 dB would correspond to a Z of ~30 meg Ω. More significant is the equivalent capacitance, as evident of the 50 kHz rejection of ~125 dB, corresponding to under 2 pF...or about 4 times better than the ZTX796A test circuit performance of **Figure 4**.

This Bohley 1VbeCS also clearly outperforms the **Figure 1** Tee circuit based version, using the same component count. A very low cost version could also be realized, by using a standard Zener for D1 (such as the 6.8V 1N5235B). *So, once more we've learned a lot here, just by watching.*

I do hope others will recognize the merits of Tom Bohley's improved circuit, and find it as useful as I do. Indeed, a fitting tribute to his contribution.

Some General Notes for These Enhancements

For these 1VbeCS enhancements, the choice of

transistors determines the ultimate performance, as well as the suited voltage ranges. Q1 should be a 2SA1016K or similar, for best performance. Q2 sees a much lower voltage, but, the lower capacitance 2SA1016K may be preferred here also. Performance isn't all about just voltage rating or low capacitance. In circuits where Early voltage^[7] limitations come into play, the 2SA1016K PNP or 2SC2362K NPN can be more effective. A classic higher voltage part is the 2N5401, still available and recommended. In fact

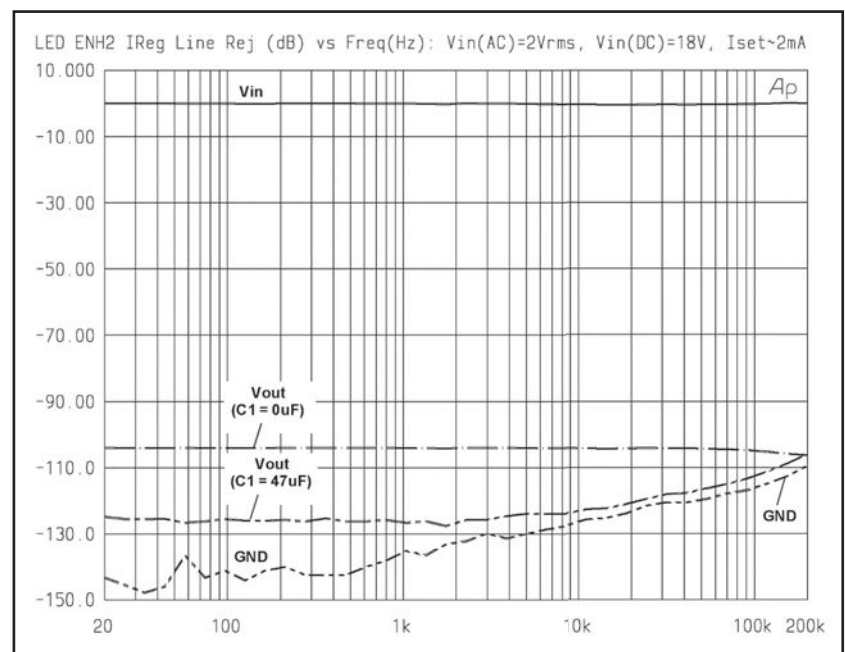


Figure 8: The performance of the circuit shown in Figure 7a enhanced LED current source shows an improved low-frequency and high frequency rejection characteristic when C1 is used. (The data was taken at 18 V/2 mA with green LED and 2SA1016ks, no cascode.)

several different PNP types were SPICE-tested in the **Figure 1** circuit, and all outperformed the relatively high capacitance ZTX796A (not recommended). A more recent device with notable specs for current source use is the PNP 2SA1579 (NPN complement of 2SC4102). These have a 120 V rating, and both low Cob and high gain. Right now these are not bench tested, but they do show high promise in SPICE.

While current sources have been shown in these PNP-based examples, analogously operated high performance current sinks are also possible. Just use NPN transistors with appropriate biasing for negative rails, reversing C1/D1 polarities as fit.

Picking a Current Source Type

After the above discussions, a logical question is this: Which current source is best for me? Here

are some details that should help decide what fork to take, or, as Yogi said, “When you come to a fork in the road ... take it”.[6]

The 1VbeCS type of circuit offers excellent performance as noted above. Unlike the LED and diode-based types, it is not subject to errors or performance limiting due to transistor Early voltage, because of the corrective feedback loop. This is a major asset. On the con side of things, it has a built in temperature sensitivity, since the reference voltage is 1 Vbe. And, the transistors need to be chosen for performance points beyond just their voltage specs. Low capacitance is a big virtue here!

By contrast, the LED and diode-based sources are subject to limits due to the Early voltage of the transistor used. Thus, even when they are stabilized for line voltage variations, this will put a final upper limit on performance. Two variants of LED-based sources are shown in **Figure 7**, using the Tee circuit in **Figure 7a**, and a Zener stabilized version in **Figure 7b**. Note that either form could also be used with the two-diode versions, by substitution of two forward-biased diodes in place of LED1.

The main feature of **Figure 7a** and **Figure 7b** is that the use of a Red or Green LED1 device allows a low-TC to be achieved. This scheme was noted years ago by Pete Lefferts.[9] The forward voltage drift of the LED matches that of the transistor, assuming they both see similar dissipations. In these two circuits R1a (and or R1b) are chosen for a 1 mA in the LED. In the Zener version of **Figure 7b**, R1a sets the 1 mA LED current, and R1b the Zener current.

Performance of the **Figure 7a** Tee circuit version is shown in **Figure 8**. Here, as was true for **Figure 2**, the use of C1 is clearly beneficial. This example uses the 2SA1016k transistors, which are a good choice for minimizing Early voltage errors. As a result, performance is still quite good, just not nearly as good as the Bohley 1VbeCS of **Figure 5**.

As noted back in the original “Sources”[1], errors due to Early voltage limits can be minimized by the use of a cascode connection for Q1. Usually this would require extra diode(s) and an extra transistor for the cascode, further complicating things. A very simple and attractive alternative here is a bipolar/JFET cascode, some of which have the powerful advantage of being *self-biased*, requiring no additional parts beyond the cascode transistor.

A great example to be cited here is the Geza Csanky cascode, first described in 1963.[10] This patent shows a cascode using an NPN transistor combined with a *depletion mode* N-channel FET, in **Figure 4**. As noted, the enormous virtue of this

Project Files

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

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approach is that it is entirely self-biased, making it easy to implement given suitable parts.

The Csanky cascode can also be applied using a PNP bipolar and a depletion mode P-channel FET, as is noted in **Figure 7c**. For example, a low voltage PNP such as a 2N3906 could be used as Q1, with J1 a 2N5462. *Note that the FET must have an I_{dss} higher than the desired I_{out} , and the voltage limits of the FET must be observed.* Thus, the 40 V and 4 mA limits of the 2N5462 can restrict things. *Caution: Don't use Figure 7c on 50 V, unless you have a higher voltage FET!* But alas, higher voltage P-channel JFETs are quite scarce.

However, similar availability limits are *not true* for N-channel FETs, and in reality Csanky types of cascodes were shown in [1, 8]. In fact, the test setup changes of [8] mentioned earlier were necessary to resolve just how good a cascode of DN2540 MOSFETs can perform. So, this "all-weather" cascode would be a very useful fork to choose—the highest performance with just five parts! Not many current sources achieve combined

impedances of 100 meg Ω and ~ 2 pF. And, being a 2T circuit type, it is indeed an all-weather choice. A fork worth taking!

A useful bibliography of higher performance current source references is included in [11].

Summary

Hopefully these discussions will enable users to select better performing current source circuits. 2T circuits are immune to the types of errors potentially seen with the 3T 1VbeCS described earlier and other circuit variants. If considering any 3T type of source, placing it within a SPICE simulation such as **Figure 3** should quickly show whether it can be useful. The files used in the simulations of **Figure 3** are included within the kit of [8]. 

Editor's Notes: At the author's request, the article content was not edited. 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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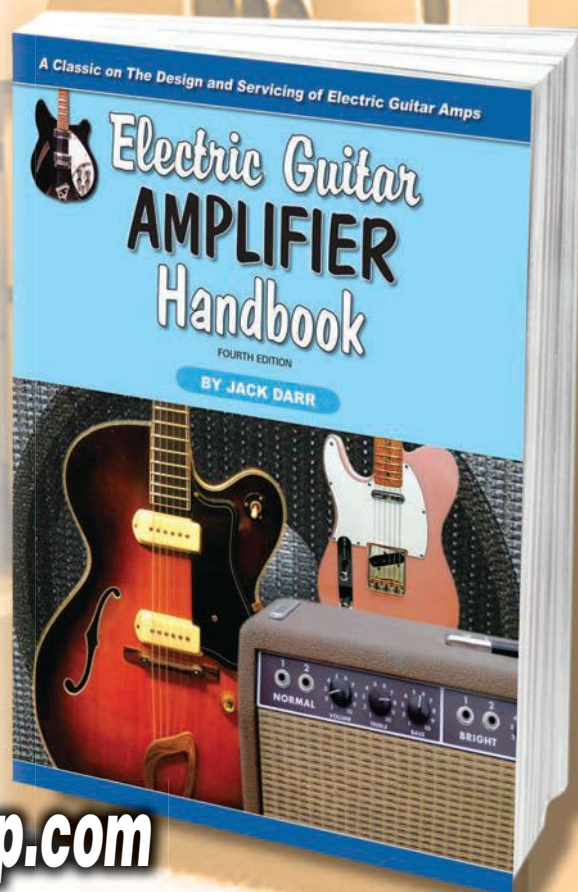
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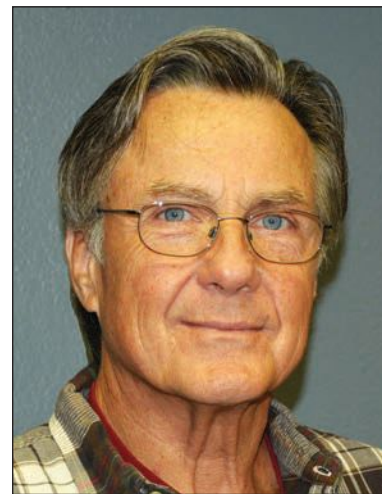
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Music Takes Flight with Raven Audio Sound Systems

An Interview with Dave Thomson, CEO of Raven Audio

By
Shannon Becker
(United States)



Dave Thomson turned his passion for music and vacuum-tube sound into a thriving amplifier company called Raven Audio.

SHANNON BECKER: Tell us a little about your background and how you became interested in audio.

DAVE THOMSON: I have always been interested in music. I made my living as a professional touring musician and singer/songwriter for about a decade. I also have owned many different tube guitar amplifiers and loved upgrading old historic home amplifiers and receivers (e.g., Dynaco, McIntosh, Eico, Scott, Fisher and several others).

SHANNON: Why are you such a huge fan of tubes?

DAVE: I have always preferred the sound of tubes, but I do even more now that new technology enables it to be so much more precise, accurate, and clean. There is something very human about a great vacuum-tube-based audio system.

And, there is so much more being written about them now that they are growing more popular at such a dramatic pace. Many of the negative responses you used to read were either partly or completely untrue, unimportant, or written from a clearly engineering standpoint where they completely discount or do not understand that music itself is a very imperfect, human, and emotional thing. You should always approach vacuum tube amplification with that knowledge

People get caught up talking about perfection, precision, exactness, distortion, and a whole litany of other things that from an engineering standpoint they think is important. They relate to how precise and

perfect digital and solid state can be when compared to vacuum tube audio devices. They will rant and rave about distortion until they are blue in the face, but those guys are missing the real truth about music. They just don't get it...very few of those guys are musicians. Not many of them are good musicians for sure! Why do you think there is a knob on every guitar amplifier that says "distortion?" There is something inherently human about slightly imperfect sound. It gives what we hear depth and realism.

You can squeeze the soul right out of the music if you try to make it too perfect. It will turn sterile and the more perfect you make it, the more boring it can get. The same goes for the device playing the music.

If the sonic signature of the power amplifier is too clean and perfect—the listening experience can be boring and unattractive even irritating. That is not what music is all about. Emotion is. It is so much easier to get up and walk away from most digital or solid-state systems than it is to get up when listening to a really great tube system playing great music. It is also why the marketing of those devices is usually filled with claims and comments like "tube sounding" or "sounds like a tube system" and so on.

So why not just get a tube system? Yeah, they can sometimes be a little troublesome. But that is precisely what tube audio is all about. Vacuum tube systems usually have more than one tube in them. And each one of those tubes can be swapped with another brand of the same or similar type. That is where tube audio really gets interesting. This is also why it can also be so darn much fun!

You can find some great old tubes to install here and there that will slightly change the amplifier to better match the room or the speakers or the preamp or the cartridge or the... well you get where I am going here. That is just another thing that makes vacuum tube systems so much fun. But tubes are made of glass. Glass is imperfect—and there that word is again.

They are not for everyone nor for every situation. Tube audio takes a little work. But, I have always found that it is well worth it. You will never beat the real “human” sound of a really great tube audio system. There is no answer as to why people react to music the way they do, but striving to make it perfect and completely without distortion and imperfection will rip the soul out of the art of the performance. It would be nice if more people understood this. And I am here to try to help with that!

SHANNON: Is that why you decided to open Raven Audio (www.ravenaudio.com)?

DAVE: It all started at about 3 a.m. one sleepless night in 2010 when, while cruising the Internet looking at audio gear, I encountered the website of a well-known (in that hemisphere anyway) vacuum tube audio engineer/designer from Seoul, South Korea. I did not know who he was at the time, I just saw some pictures of his tube amplifiers. I sent him an email saying “hello from Texas, USA, nice looking amplifiers.” He immediately responded telling me that he once worked at Texas Instruments in Dallas and really missed Texas, well, let’s just say that I didn’t get much sleep that night.

A few months after that initial email exchange he was sitting in my Texas living room. When I first listened to one of his tube amplifiers, I was enthralled with the sound. I knew it was a little different than anything I had ever heard, and I had by then owned many different brands of high-end audiophile-grade products. Here in my living room was a man who had more than 48 years of experience, gained mostly through 40 years of owning one of the largest repair shops in a country where old US and European audio products are highly revered and avidly collected. He had repaired and upgraded almost every existing brand in the world, which gave him a unique window into the way all of these different designers had built their products. He also knew all the tricks and methods they used and then came up with a few of his own.

By November 2010, Raven Audio was born. I sub-contracted him to design all of the products in what is now Raven Audio’s current line-up. He has since retired and now it’s time for someone else to take over where he left off. I want to bring a US engineer



Raven Audio’s first product was the Reflection Integrated Stereo Amplifier—a single-chassis, push-pull, KT-88 through KT-150 and 6550-based integrated amplifier.

into the mix. Eventually, the right person will find his way to Raven Audio and will continue our work.

SHANNON: Where do you get your tubes? Do you have a favorite?

DAVE: I actually get my tubes from many sources, because I prefer old stock tubes. But for power tubes we have to go with what is available and the few companies that manufacture tubes today. I source our power tubes from New Sensor in New York. I will never buy Chinese tubes again.

SHANNON: What was your first product?

DAVE: Our first product was a version of the Reflection Integrated Stereo Amplifier. The second product was a Shadow Stereo Amplifier. Then, we developed the



The second product, the Shadow Stereo Amplifier, is tube-rectified with a solid-state power supply. The phono stage is loaded at 47 kΩ for high-output MC (moving-coil) or MM (moving-magnet) cartridges.



The Nighthawk is Raven Audio's entry-level amplifier.

Shadow Preamplifier to go with it. Soon after that I asked our sub-contracted product designer to please develop a lower-priced but great-sounding alternative lineup of consumer level products that we could offer for less than \$5,000. So, we came out with the Avian Series, the first model of which was the little Nighthawk. It sold for \$1,995, had a manual volume, and was just a fabulous folded-steel chassis with automotive paint. That price limit was finally breached this year with the introduction of new Eagle and Golden Eagle integrated amplifiers. So now the rule is to keep it under \$6,000.

SHANNON: Can you describe the main differences between your Elite Series and your Avian Series?

DAVE: The Elite Series lineup are top-quality products, heavy, and expensive to build. They are all housed in thick CNC-machined anodized aluminum chassis and hand-wired using the absolute best sounding and highest quality parts in the world. Not every single part is costly, but they have to be dependable and sound great. That is where experience came into play. It's not about which parts you use, but where you use each one of them within the circuit.

SHANNON: Talk about your Goldfinch Tabletop System. What are some of the challenges you faced? And, is it now everything you envisioned it could be?

DAVE: The Goldfinch Tabletop Multicomponent Audio System is exactly what we designed it to be. It's the ultimate high-quality, yet affordable, fantastic-sounding multi-chassis system of tiny vacuum tube components. It took a long while to come up with something small that sounded like the premium top-of-the-line products out on the market. This was our goal. It just took some time for us to get it right.

It's built for small spaces, but it will also work well enough with just about any sensitive speaker

system. Speakers with a sensitivity of 95 dB or higher will work very well with this system. As time goes on, there are going to be more higher-sensitivity speakers coming out as people start to get into low wattage tube amplifiers. Low-power tube amps with sensitive speakers can be a very special combination. We have plans for a single-ended power amplifier for this system. The 8" x 12" footprint of the chassis takes up very little room, so they are perfect for small residences.

We aren't really finished with that lineup yet. But, we already have spin-off components in the same chassis, the Cirrus Line Stage Preamplifier and the Stratus MM Phono Preamplifier. They will both have separate power supplies, and since they are more reasonably priced we hope they will become popular. The Goldfinch Series is an entirely different world of products for small listening environments. And as cities get more crowded and space becomes a premium, and more expensive, these components will become even more important and useful in those environments.

SHANNON: In addition to the preamplifiers, amplifiers, and the tabletop system, I see you also have your own line of capacitors. Can you explain why you offer Raven Audio Capacitors?

DAVE: We had to make a large minimum purchase because we were in development with this contractor for such a long time trying to come up with something really special. That's just the way it works when you ask a company to build something for you that's new, different, hopefully special.

We approached and later hired the same company and the engineers that developed the famous "V-Cap" and some of the other more famous film capacitors, and told them we wanted to develop the best audio capacitor in the world. At first they laughed. But eventually we agreed to try, and during that quest we hit on something different that worked to make the RavenCap audio capacitor special. Along with some other technology discovered along the way, they increased the level of Teflon used in the matrix, and the addition of two other poly-components along with the use of a high-pressure injection method to force the bubbles out during the process instead of a normal "pouring" method. That, among a few other things, turned out to be the key to the difference.

The result is an almost shocking new realism that suddenly appeared when the caps built using the new method were first tested in prototype form. We knew we had something different, something really unique, something really very special. It was exciting.

Once we had them, I thought it might be fun to

offer some of them for sale into the DIY community. So we put them up on our website just to get some reaction from people in the hands-on community, mainly to see if they thought it was as special as we do. But, we will only sell them for a limited time. After all... they are a large part of our unique sound.

The RavenCap is also the source of the incredible punchy dampening power that all of our products possess. You can almost feel this huge deep wave when the music is played at higher levels. It makes the bass sound so clean and well-defined yet powerful. It almost clings to the floor forming a deep layer that flows across the room at lightning speed... it's really something.

The music literally jumps out of the speakers—and so effortlessly. It also cleans up the bloat that can plague some tube audio designs allowing the music to flow so “free and easy,” almost completely without restriction, haze or veils. It's kind of amazing. I cannot say enough about those really wonderful capacitors, except that they are a game-changer.

SHANNON: I understand you have a lot of exciting changes happening in 2017. How did the changes come about?

DAVE: Yes we have a lot of changes coming. We were already in the process of moving into a new much larger facility due to increased demand for our products when things started to change even more. During the move, demand kept increasing, so we had to stand back and make a few decisions or it looked like we might outgrow our new facility before we even get fully moved in.

We had originally planned to move in and set up our own manufacturing shop and have two separate manufacturing areas in the same building. One side would be set up to build the Avian Series products, which are our PC board based integrated amplifiers, and the other side would house the Elite Series manufacturing area, where all of the CNC-machined aluminum, hand-wired point-to-point would be built.

The Avian Series started selling so fast that we decided to take the Avian Series to an already existing highly specialized electronics manufacturing facility in College Station, TX. This move will enable Raven Audio to build more than 100 units per month.

The main thing is that our customers will never again have to place an order and wait for their amplifier. The plan is that we will keep several of each model in stock, and each purchase will trigger an immediate replacement to be built. Every week stock will be sent to over and stored at the shipping facility in Trinity, TX, ready to send out at a moment's notice.



The Goldfinch Tabletop Audio System is designed for small spaces (e.g., as a build-in unit for RVs, motor coaches, private planes, yachts, and limousines). It is also perfect as a second system in the bedroom, hunting cabin or lake house.

The Avian Series line are all similarly housed in folded steel chassis with aluminum face plates and have a separate raised aluminum preamp plate on top. The existing Nighthawk MK2.1, the Blackhawk LE, and the Osprey all have handles and weigh in at about 40 lb. and are in the same basic steel chassis.

The newest members of the Avian Series—the Eagle and Golden Eagle—are coming out in 2017 and are much larger, about 60 lb. each. They are heavy and must be handled from the bottom sides of the chassis, and for now the Eagles look really nice as they are, and they are safe to carry this way.

Now, the new Onalaska shop will be dedicated completely to the manufacture of the Elite Series, our all machined-aluminum, point-to-point, hand-wired product line. So, expect some really exciting changes in the coming years from Raven Audio.



For a limited time, Raven Audio is selling a select number of proprietary Raven Audio PTFE and foil capacitors—the same capacitors you'll find in all Raven Audio products.



Mark Trotter, one of Raven Audio's new partners, checks an Osprey amplifier, which is sitting on a shelf in the new manufacturing shop.

SHANNON: You also took on some new business partners. Can you elaborate?

DAVE: The new partners are all previous customers of either us or Legacy Audio. Tim and Connie Qualls are old friends... Tim is a fabulous drummer of some note and has had a long love affair with vacuum tube audio, mainly guitar amps. Connie is a long-time small business owner.

Marc and Kyle Trotter are a father and son active investment team, and Marc just retired from a long history in sales at Bristol Myers Squibb. Marc will work



The Golden Eagle and the Eagle are the last two products in Raven Audio's Avian Series integrated amplifiers.

with Raven Audio and help us at the store and the audio shows. Kyle will help us with the business growth planning and documentation. Our new partners came to us through osmosis and offered their help due to their extreme love of high-end audio and a desire to see Raven Audio succeed!

SHANNON: Can you tell us about your new facility?

DAVE: The new store is in Onalaska on Lake Livingston, which is about 100 miles north of Houston. We are distributors for Legacy Audio, Mytek Digital DACs, Douglas connection cables, and we are also bringing in Emotiva home-theater amplifiers and JVC projectors in the near future.

We have a perfect home-theater room, three two-channel listening spaces, a nice little office, a decent-sized manufacturing room, and an additional room for box storage, burn-in and packaging. There are also two smaller satellite rooms that currently house a little photography studio and a bedroom. For the short future we plan to stay here because it is kind of a perfect location. It is only 15 miles from the original home of Raven Audio "La RanChita" in Chita, TX.

SHANNON: Since you parted company with your original designer, can you tell us your plans for the future?

DAVE: For now, we are employing two part-time electronics engineers. We also have access to several in-house engineers in College Station and have an open position for a full-time vacuum tube audio designer. We are good for now, but are taking applications for the right person.

SHANNON: What can we expect to see from the Raven Audio this year?

DAVE: For 2017, we are officially rolling out the Eagle and Golden Eagle, which are the final members of our Avian Series integrated amplifiers. The Eagle will be able to use any power tube in the 6550/KT88 lineup (e.g., the KT88-90-100-150) and each different power tube choice can really change the sonic signature of the amplifier in both the sound and power output. The Eagle will be formally rated at 50 W per channel. Of course, this will vary slightly according to each tube quad chosen. The Golden Eagle will use only the KT150 tubes and is rated at 75 W per channel, but the KT150 is able to bring up to and even over 83 W per channel when using its full potential. It's the most powerful octal base beam pentode power tube out there currently.

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You Can DIY!

Cherry Bomb— A VFET SET Amplifier (Part 1)

The Concept, the Design, and the Plan Modifications

One of the great things about doing a DIY project is that the only one you have to please is yourself. Does it have to be practical? No. Does it have to be cost effective? No. Do you have to worry about what you call it? No. The list of limiting things you don't have to consider goes on. In fact, we should coin a new acronym, DIFY, meaning Do It For Yourself. This project is the epitome of that philosophy.

By
Thomas Perazella
(United States)



Photo 1: Yamaha's 2SK77 case is huge compared to other common power transistors.

Although I build a fair amount of electronics, I don't usually build amplifiers. With the state of electronic technology today, if you are just looking for a good performing amplifier it is usually more cost effective to purchase a commercial unit. As a DIYer, you cannot even come close to the component and chassis costs that commercial builders enjoy. So why build an amplifier? In this case, it was both a technical and an endurance challenge.

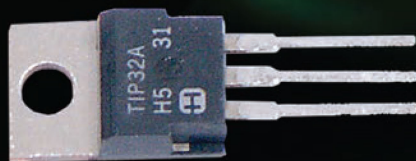
This project actually started because of a discovery I made while going through a parts bin. I found two transistors that looked like TO3-cased devices on steroids. The nomenclature was Yamaha 2SK77 with what looked like a date code of 6107. Getting out a large broom to clear the cobwebs from my brain I seemed to remember getting two of these transistors that were samples from a friend who worked at an audio magazine about 40 years ago. They had been sitting in my bin ever since. At first glance, one of the most impressive

things about this transistor is its size. Compared to the industry standard TO3 and TO220 cases, it is a giant. **Photo 1** shows the 2SK77 next to the TO3 and the TO220 cased transistors.



The 2SK77 devices were used by Yamaha in its B2 amplifier. The goal was to have a transistor amplifier with the same characteristics as a triode vacuum tube amplifier. Yamaha called the transistors vertical field-effect transistors (VFET). Another designation for this type of transistor is a Static Induction Transistor (SIT).

In recent times, Nelson Pass has written papers on amplifiers that use this type of device and he references the 2SK77 as desirable, while acknowledging they are no longer obtainable. For reference, these are a few manufacturer specifications for the 2SK77 transistor: The maximum operating voltage is 200 V, the drain current is 20 A, the transconductance is 1,500 mU, the output impedance is 5 Ω , and the power dissipation is 200 W.



Before building my VFET SET Cherry Bomb amplifier (see **Photo 2**), I brought them over to a friend, David Berning, to determine if the devices were functional and had triode-like characteristics. David builds very innovative and high-quality tube amplifiers and has a curve tracer. So, we measured the characteristic curves of the transistors. To our surprise, they looked similar to the 300B and the 845 triodes—with one notable exception. With the tubes, the curves are further apart at lower voltages, getting progressively closer together as the voltage increases (see **Photo 3**).

With the VFETs the opposite is true—the curves got closer together at lower voltages and further apart at higher voltages. If you compare the graph of the first VFET shown in **Photo 4** with **Photo 5**, which is the curve of the second VFET, you can see that they are certainly not a matched pair with significant gain differences. These gain differences would have to be addressed later if we decided to build an amplifier. David and I were excited about building an amplifier that could utilize these devices so we decided to forge ahead on a team project.

The Concept

It was clear from the beginning that since we had only two devices, we would have to build a stereo amplifier with a single-ended topology. When I mentioned to David that we would be designing and building a SET amplifier he reminded me that I could not use that term because it referenced a Single-Ended Triode configuration. Not to be outdone, I said the acronym was still correct because it would be a Single-Ended Transistor amplifier. Begrudgingly he agreed. Isn't DIY fun! We also decided that we would like to keep design elements similar to what it would have been in the early days of tube designs (e.g., single-ended Class A 300B amps). This included a main power supply that was choke filtered.

Since the devices were no longer available, it was critical to protect them through the design and ultimate operating phases. No transistor killing mistakes could be allowed or the project would immediately be relegated to the scrap heap. That can be quite a challenge with such an unusual device in what turned out to be a very non-standard if not unique design.

Some Single-Ended Basics

With single-ended designs, the active device has to be connected in series with a constant current source (CCS). When the active device changes impedance, the CCS will increase or decrease the voltage across the active device in order to maintain the same current. When a load is connected to the



Photo 2: When we complete the project, our Cherry Bomb VFET SET preamplifier should look like this. Here is a front view with the VU meters illuminated.

junction of the CCS and the active device, the load will see that varying voltage, which in fact is the audio signal. For efficiency reasons, we decided to use chokes as the constant current sources. For more information on that decision and other technical discussions and testing, please see the information provided in the Supplementary Material on the audioXpress website.

Other Design Considerations

To increase performance, we also decided to make the amplifier output transformerless. Building output transformers with good performance is always difficult, and especially so for Class-A designs where there is significant DC current in the core. Typically, they have poor high-frequency characteristics. Having no output transformers eliminates that problem. David has been designing output transformerless tube amplifiers for many years, using his patented ZOTL configuration (see Resources). Since we were not working with high impedance devices with high voltage power supplies, direct coupling without a ZOTL impedance converter was possible.

For that to work required a very unorthodox floating power supply design combined with a servo that adjusts the DC bias current of the output FETs working against the CCS to maintain an essentially constant steady state current through the chokes. Since the speaker load is referenced to ground and the floating power supply only references ground through six large electrolytic power capacitors, DC is blocked at the speakers while allowing passage of AC signals.

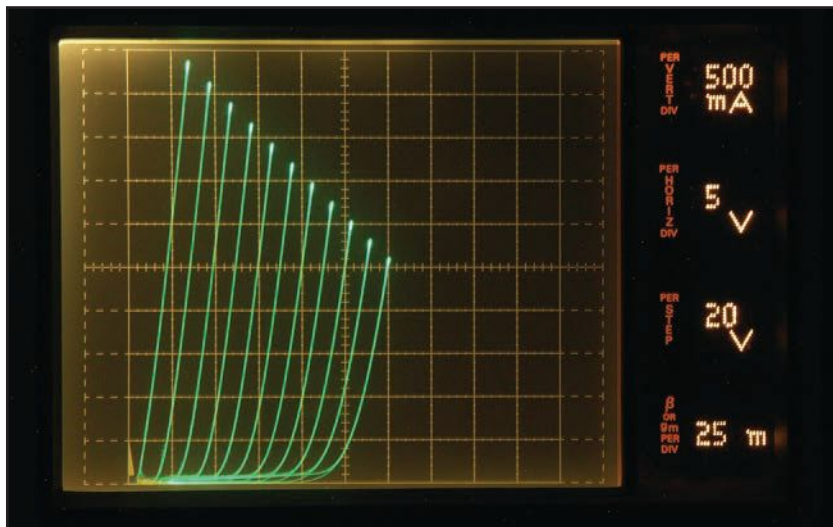


Photo 3: These are the characteristic curves for a 300B triode.

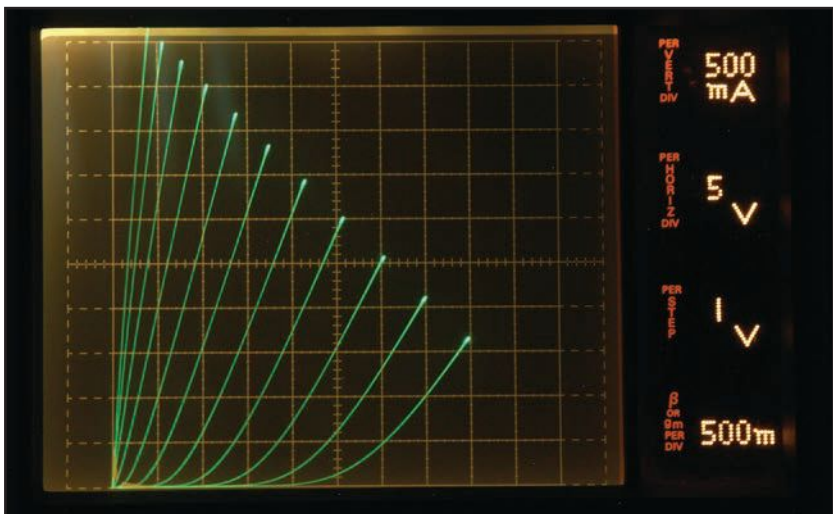


Photo 4: Here are the characteristic curves for first VFET.

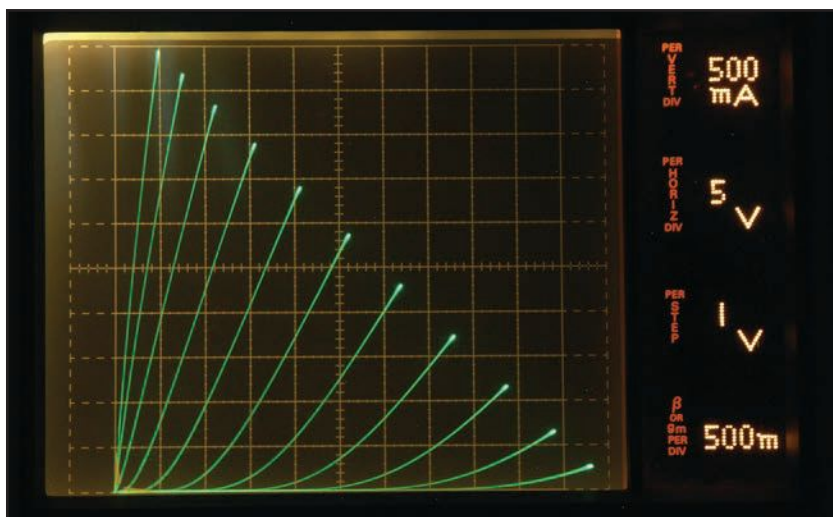


Photo 5: For comparison, these are the characteristic curves for second VFET.

The Best Laid Plans

To say that the design changed many times over the course of this project would be an understatement. I am not going to show the various routes taken to get to the final results, but in the early stages of the design, there were three sections of the power supply all working off one power transformer. The main output power came directly off the secondaries of the power transformer with no direct connection to ground. The other two voltages came from voltage multipliers referenced to different parts of the power transformer and sections of the main DC power supply. Also included was a control power MOSFET to decouple the supply if certain operating conditions were not met. To say that this was overly complicated and troublesome is a gross understatement. Problems cropped up all over the place including timing, power dissipation, and current capability issues.

To be successful in DIY, one of the primary traits you must possess is a good sense of humor. During the initial phases of design, we substituted a vacuum tube ZOTL circuit that was on an open breadboard with the tube acting as an analog in place of the VFETs. As the testing progressed, we managed to destroy a number of control MOSFETs due to several different reasons. If the VFETs were in the circuit, they could also have been destroyed by these failures.

We also sometimes checked the control MOSFET by putting a finger on the heatsink tab. Often that was accompanied by a loud ouch and several unprintable expletives. The MOSFETs were inexpensive and we suffered no permanent scars from the burns. In another instance, after disconnecting the tube emulator circuit to check a problem, David picked it up before the storage capacitors were discharged. He received a significant jolt from the capacitors but since there were a number of expensive components on the board David was loathe to drop it. As a result, there ensued a period of the board alternately being tossed into the air and caught with another shock to him each time it hit his hand. If I had a video of that scene it would have been a great hit on YouTube, but again the expletives would have had to be deleted.

To shorten an otherwise long tale of the power supply design, we finally eliminated the voltage multipliers and substituted two small separate transformer-isolated regulated supplies. Timing issues were eliminated because those supplies would stabilize long before the main supply reached operating voltage. In addition, noise decreased and performance improved.

The Final Design

After the power supply was completed, the

driver and gain stages were tested. Input gain was provided by op-amp sections in combination with balanced to unbalanced converters as we wanted to be able to handle both balanced and unbalanced input signals. The input stages had different gains to compensate for the gain differences between the two VFETs as they contribute to a portion of the amplifier's overall gain.

Originally, the amplifier was designed to be either a no-feedback or low-feedback design using a selector switch. After completing the amplifier and testing it at high power, we discovered that the output chokes did not work well with the 5 Ω impedance of the VFETs at low frequencies below 20 Hz. We tested an organ recording with pedal notes at 18 Hz that did not sound right. Therefore, we changed the design to always have a small amount of negative feedback and eliminated the switched feedback capability.

The driver stages were pretty conventional except that we needed to provide higher currents than what would commonly be required with bipolar output devices. The VFETs have significant levels

of input and Miller capacitance. If high-frequency response is not to suffer, the driver stages must be able to source and sink enough current to charge and discharge those capacitances in a timeframe small enough that they become irrelevant. Those driver stages, therefore, operate at quiescent levels around 100 mA. **Figure 1** shows the final circuit configuration of the complete amplifier.

Circuit Description

Before we go any further, it is important to disclose that the current version of the amplifier is "second generation." The first was built with no output meters. After it was constructed and in use for a considerable period, I found some cool looking analog VU meters that I felt would enhance the amplifier's retro feel. I completely disassembled the amplifier, drilled the holes for the meters, and mounted them. I then reassembled the amplifier.

To begin, I want to describe the numbering system used to identify the amplifier's various sections. On the schematic shown in **Figure 1**, the power supply and protection circuits are common to

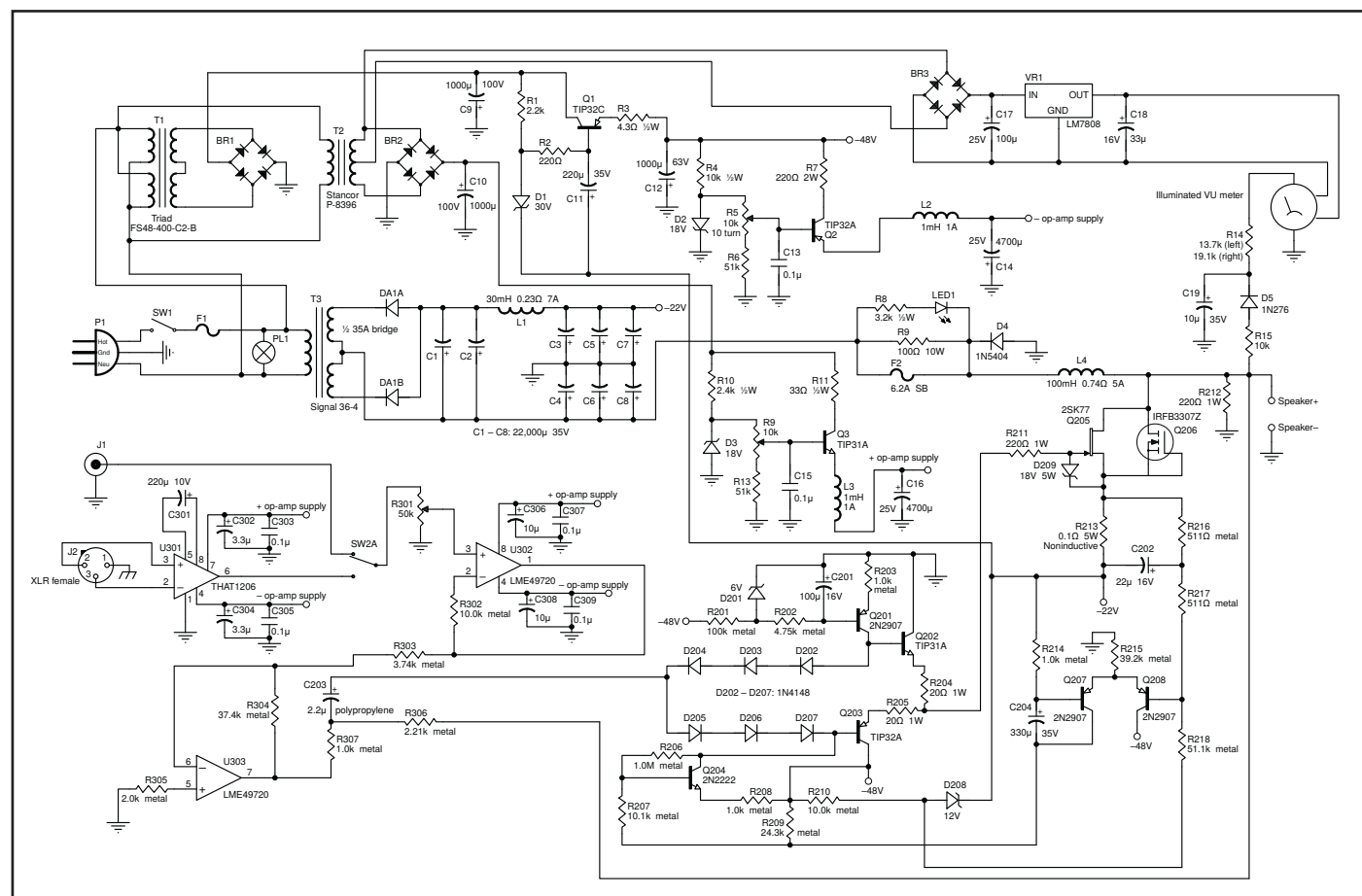


Figure 1: The schematic shows the final design for this unique amplifier.

both channels. The input gain stages, driver stages, servo stages, and output stages are separate for the left and right channels but they are identical in component values except for one feedback resistor. R303 and R403 have different values as shown in the Parts List (see the Supplementary Material) to compensate for the different gains of the VFETs. The schematic shows only the left channel of those stages.

- Components with numbers from 1 to 99 are in the various sections of the power supplies. Note that the output chokes are considered power supply elements in the numbering schema.
- Components with numbers from 100 to 199 (not shown on the schematic) are for the right channel servo, driver, and output devices.
- Components with numbers from 200 to 299 are for the left channel servo, driver, and output devices.
- Components with numbers from 300 to 399 are for the left channel input and gain stage devices.
- Components with numbers from 400 to 499 (not shown on the schematic) are for the right channel input and gain stage devices.

Looking at the schematic, you can see that there are several parts that are either custom or not commonly used in DIY amplifiers. The power supply choke and the constant current chokes are all custom designed and hand built. More about that later. The THAT 1206 balanced line receivers are also not often seen in amateur applications.

Power Supplies

There are four sections to the power supply: a positive supply for the op-amps; a negative supply for the op-amps, drivers and servos; a supply for the VU meter illumination bulbs; and a floating main supply for the output sections.

The positive supply for the op-amps is the most straightforward design. Transformer T2 provides isolated power to rectifier bridge BR2. Capacitor C10 provides the first stage of filtration. The resulting filtered DC is applied to a discrete voltage regulator with an output of +16V.

The floating supply for the meter bulbs is also simple. One end of the secondary of the transformer and the center tap feed a bridge rectifier and a filter capacitor. An 8 V integrated voltage regulator is used to limit the voltage to 8 V and is filtered by a small capacitor for regulator stability. Although the lamps can work at 12 V, we chose 8 V to give an illumination that is warm in color to enhance the retro look.

The negative supply starts with transformer T1 supplying isolated power to bridge BR1. Output from the filtered DC goes to a first stage regulator using Q1 as the active pass device. The voltage reference for the pass device is Zener diode D1, but the cathode of that Zener diode is not referenced to ground but rather the -22 V from the main floating supply. Therefore, the -48 V output from the first stage is the result of the 30 V Zener diode, the -22 V floating supply, and the drop across Q1 and R3. That -48 V is fed to the driver and the servo stages. The negative op-amp supply uses the -48 V supply as its input. The design of that stage is the negative analog of the positive op-amp supply.

The main supply uses T3, a centered tapped output device. Rectification is provided by half of a high power bridge that I had available in my parts bin. The first stage of filtration is provided by capacitors C1 and C2. Capacitors C3 through C8 are fed from the first stage and are in a series parallel configuration with the center of the array referenced to ground.

Unlike most configurations, the center tap of the transformer is not referenced to ground. Rather, it is connected to the top side of some of the filter capacitors and then to the tops of the constant current chokes. This configuration, although unconventional, sets the speaker voltage with no AC input signal to be approximately zero. The speaker and a small resistor for stability in case there is no speaker being connected are the only DC paths to ground. The power supply capacitors will block any DC voltage that tries to develop across the speaker, while allowing AC signals to pass through.

Chokes

Purchasing chokes with the specifications necessary for this project proved to be almost impossible, so we decided to build the chokes ourselves. Buying small batches of laminations is

About the Author

Thomas Perazella is a retired Director of IT. He received a BS from the University of California, Berkeley campus. He is a Past President of the Rockville Chapter of the Izaak Walton League of America, one of the oldest national conservation organizations in the US and currently is the Treasurer. Audio has been his passion for more than 50 years and he is a member of the Audio Engineering Society, the Boston Audio Society, the Philadelphia Area Audio Group, the DC HiFi Group, and the DC Audio DIY Group. He has written for *audioXpress* magazine and prior to that for its predecessor, *Speaker Builder*. In addition to audio, his interests include photography, cooking and competition pistol shooting. He has authored several articles in professional audio journals and taught commercial lighting at the Winona School of Photography. Recently, he received a patent on a cost-effective high-efficiency LED lighting system for commercial and residential buildings.

a pain so I found some old transformers I could disassemble to get the laminations. How hard could that be? Well, the choke building project turned out to be like going through basic training in the Army. You learn a lot but never want to do it again. For more information, check out the Supplementary Material for this article found on the audioXpress website.

Servo Stages

The DCR of the output chokes was approximately $0.75\ \Omega$. With a desired quiescent DC current of about 2.5 A that results in a voltage drop of around 1.9 V. The resulting positive output of the floating supply, therefore, is about 1.9 V above ground. The servo is a balanced amplifier and has a response time well below the audio frequencies being reproduced. It controls quiescent current through the combination of the voltage drop across the reference resistor at the source of the VFET and a divider network that compares that voltage to a Zener diode reference. I told you the power supply design was unconventional. If you don't understand it, I suggest you buy one of David's amplifiers where he has it all done for you. In any case, you cannot build an exact replica unless you find the 2SK77s, which are quite rare.

Input Sections

The most modern part of this amplifier is the input stage. There are so many high-quality audio gain and other specialty function modules that I did not want to try to duplicate the functions with discrete devices. The gain stages use Texas Instruments (TI) LME 49720 dual op-amps. If you have not used these amplifiers, you should seriously look at them. When I started using op-amps back in the 1960s, this kind of performance for audio circuits in an op-amp was beyond even dreaming. I still have a spiralbound copy of the Philbrick/Nexus applications manual for op-amps, *Applications Manual for Operational Amplifiers*, published in 1968. It is quite a read.

Good design practice indicates bypassing the power supply leads right at the op-amp supply inputs. Each supply input is bypassed by a $10\ \mu\text{f}$ electrolytic capacitor plus a $0.1\ \mu\text{f}$ monolithic ceramic capacitor. Although many audio enthusiasts eschew using ceramics, they are unsurpassed in bypass applications. For each channel, one section of the op-amp U302 as a non-inverting input buffer to minimize loading of the volume control pot. The value of that pot is $50\ \text{k}\Omega$, at David's suggestion, to allow use with tube preamps having high output impedance. Resistor R302 is used to help balance the voltage created by the input bias current to minimize offset voltages. The second section, U303, is the

actual gain section and is inverting to complement the inversion in the output section resulting in correct overall polarity.

To properly interface with preamps with balanced outputs, the THAT Corp. 1206 balanced line receiver is used and identified as U301. Supply inputs are bypassed with a $3.3\ \mu\text{f}$ electrolytic capacitor and a $0.1\ \mu\text{f}$ monolithic ceramic capacitor. In previous articles, I have mentioned the advantages of using this receiver in balanced situations. The datasheet for this device (see Resources) and other InGenius products explains the principals involved in maintaining good common-mode rejection ratio (CMRR) performance in balanced situations and provides references to papers written by Bill Whitlock on this subject. The output of the 1206 is coupled to the input of U302 through switch SW2A that allows selection of either balanced or unbalanced inputs. The input gain stage is coupled to the driver stage through a $2.2\ \mu\text{f}$ polypropylene capacitor C203. Feedback is applied at the input of that capacitor through R306.

In the second half of this article, we will discuss the rest of the design, the construction, and the testing and listening results. 

Project Files

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

Resources

D. Berning, "ZOTL Technology vs Audio Output Transformers," The David Berning Co., <http://davidberning.com/technology>.

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Source

LME 49720 dual op-amps
Texas Instruments (TI) | www.ti.com



A Range of Power Supplies for Hollow-State Guitar Amplifiers

This article explores some of the power supplies that manufacturers have used for hollow-state guitar amplifiers, dating back to their beginnings in the 1920s.



By
Richard Honeycutt
(United States)

Since the introduction of hollow-state guitar amplifiers by George Beauchamp in the late 1920s, several types of power supplies have been used. The least expensive ones had no power transformer. Somewhat better, higher-powered models used a power transformer feeding a half-wave tube rectifier.

Moving up in quality and power, we find a change to full-wave rectifiers. Beginning in the late 1950s, amplifier manufacturers began to change from tube rectifiers to solid-state diodes. Let's look at some

of the power-supply circuits that amplifier makers have used along the way.

AC/DC Power Supplies

Soon after electric power for homes and businesses first became available, there was a conflict over whether alternating current (AC) or direct current (DC) power would be standard. Thomas Edison and his company, General Electric, urged the use of DC, partially because they owned

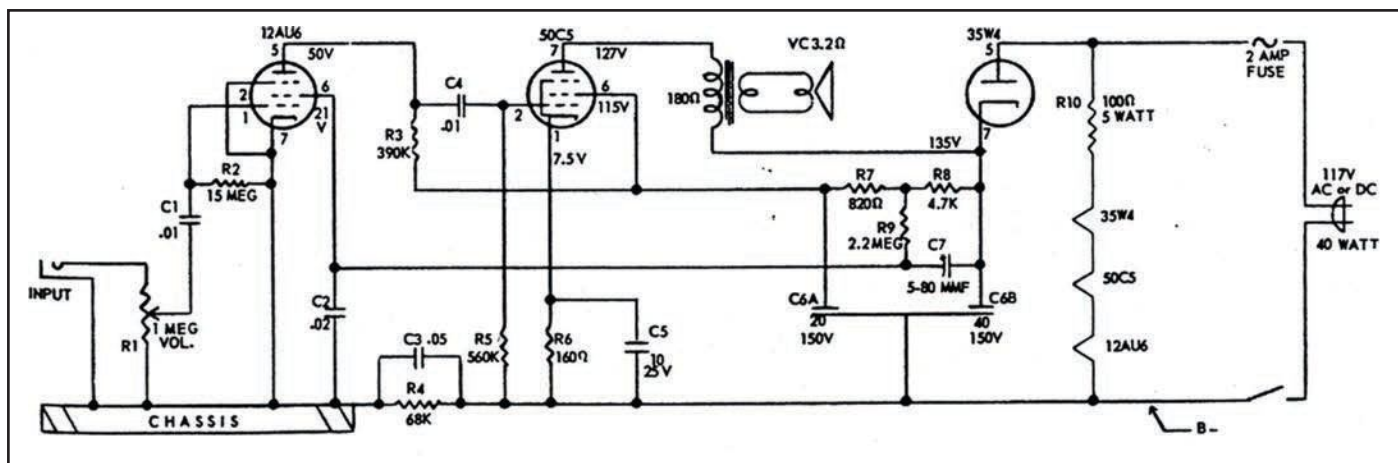


Figure 1: The Wards Airline GDR8511A amplifier had a truly transformerless power supply.

many patents for DC motors and generators. Westinghouse followed the advice of its engineer Nikola Tesla in promoting the use of AC. Ultimately, AC won the “war of the currents,” but some cities continued to use DC well into the 20th century. Parts of Boston still used 110 VDC in the 1960s.

Most customers for DC used it almost exclusively to power electric elevators, and the last DC commercial power distribution system in the United States was shut down in 2007. However, in the early 20th century, a significant number of homes were still DC-powered, so it was advantageous for any appliance or electronic device that had to be plugged into an electrical receptacle to be able to operate from either AC or DC.

To operate from DC, hollow-state devices had to have transformerless power supplies. But probably the more important reason for the manufacture of radios and guitar amps with AC/DC capability was that transformerless power supplies were less expensive to build.

In the early to mid-20th century, Sears, Roebuck, and Co. and the Montgomery Ward company were the most successful guitar and amplifier retailers. Low-end amps such as the Wards Airline GDR 8511A, whose schematic is shown in **Figure 1**, used AC/DC supplies for their universal electrical compatibility and for cost reasons.

In order to do this successfully, the tube filaments were connected in series, and their voltages should add up to about 110 V. (For many years, commercial power in the US was provided at a nominal 110 V, with permissible variation extending from 105 to 125 V. In the late 20th century, the nominal voltage was changed to 125 V. But when brownouts occur,

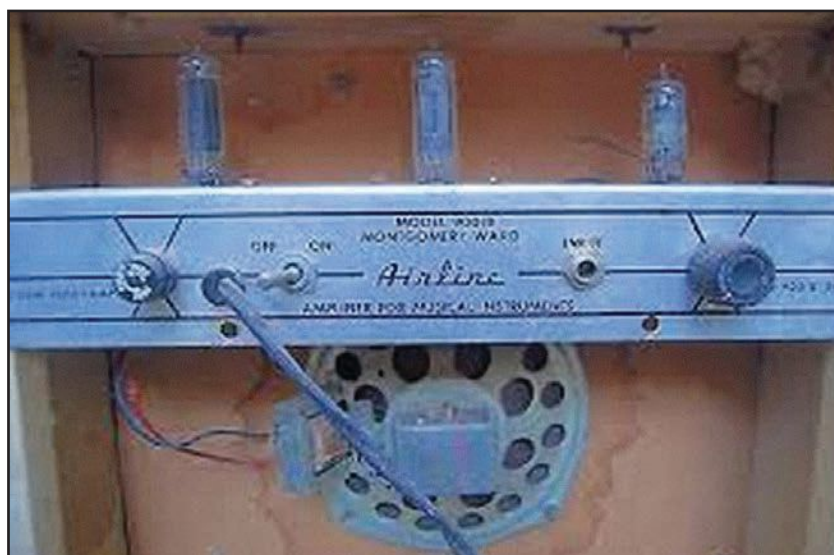


Photo 1: It appears from this rear view as if the Wards Airline GDR 9001B amplifier had a transformerless power supply, but a small filament transformer is hiding inside the chassis.

the voltage can drop to 85 V or lower.)

Adding the filament voltages of the 35W4 half-wave rectifier, the 50C5 power amplifier, and the 12AU6 pentode preamp together, we get 97 V. Running the tubes from a 110-V receptacle would have applied about 13% overvoltage to the filaments. The 100 Ω , 5 W resistor connected in series with the filaments dropped about 15 V, since each of the tube filaments operated at 150 mA, bringing the actual filament voltage in line with their specifications. C6B provided capacitive filtering for the plate supply of the output tube, and the combination of C6A, R7, R8, and R9 decoupled the high B+ from the preamp tube's B+.

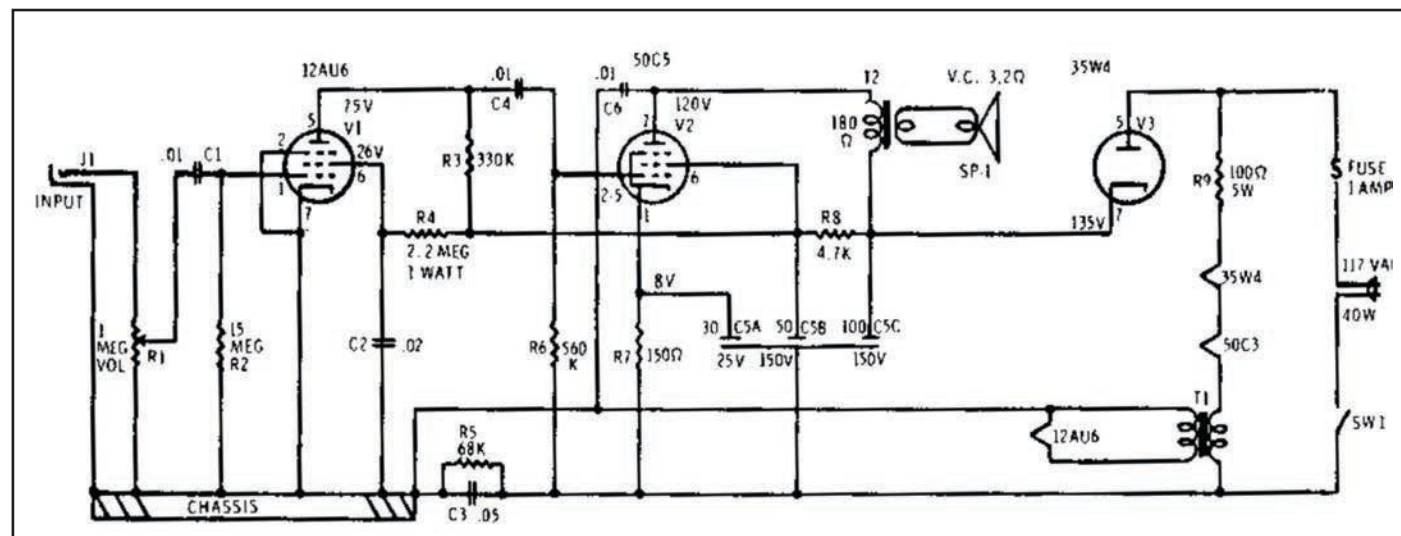


Figure 2: The GDR 9001B's circuit included a filament transformer, making the amplifier AC-only.

Photo 2: The Supro S6622 offered reverb, tremolo, more power, and a better sound than the low-priced AC/DC-supply amplifiers.



As with almost all electronic equipment using AC/DC power supplies, one side of the power line was grounded to the chassis; and thus, to the sleeve of the guitar cord, jack, and pickup covers (in guitars using metal pickup covers). This meant the guitarist had to unplug and invert the power plug if the amplifier hummed badly when powered up, and could cause an unpleasant—or even life-threatening—shock if the power plug was inserted the wrong way. When these amplifiers were made, polarized power plugs were not yet used much, and three-prong “U-grounded” receptacles were not common.

Perhaps in order to reduce hum and noise, Montgomery Ward changed from an AC/DC supply by adding a filament transformer to supply the 12AU6 preamp tube. One amplifier that used this circuit was the GDR 9001B (see **Photo 1**).

Figure 2 shows the GDR 9001B's schematic. The primary changes you'll notice in this circuit are the larger filter capacitors to reduce hum and the isolation of the power line from the chassis by the parallel RC circuit C3 and R5. This circuit was used

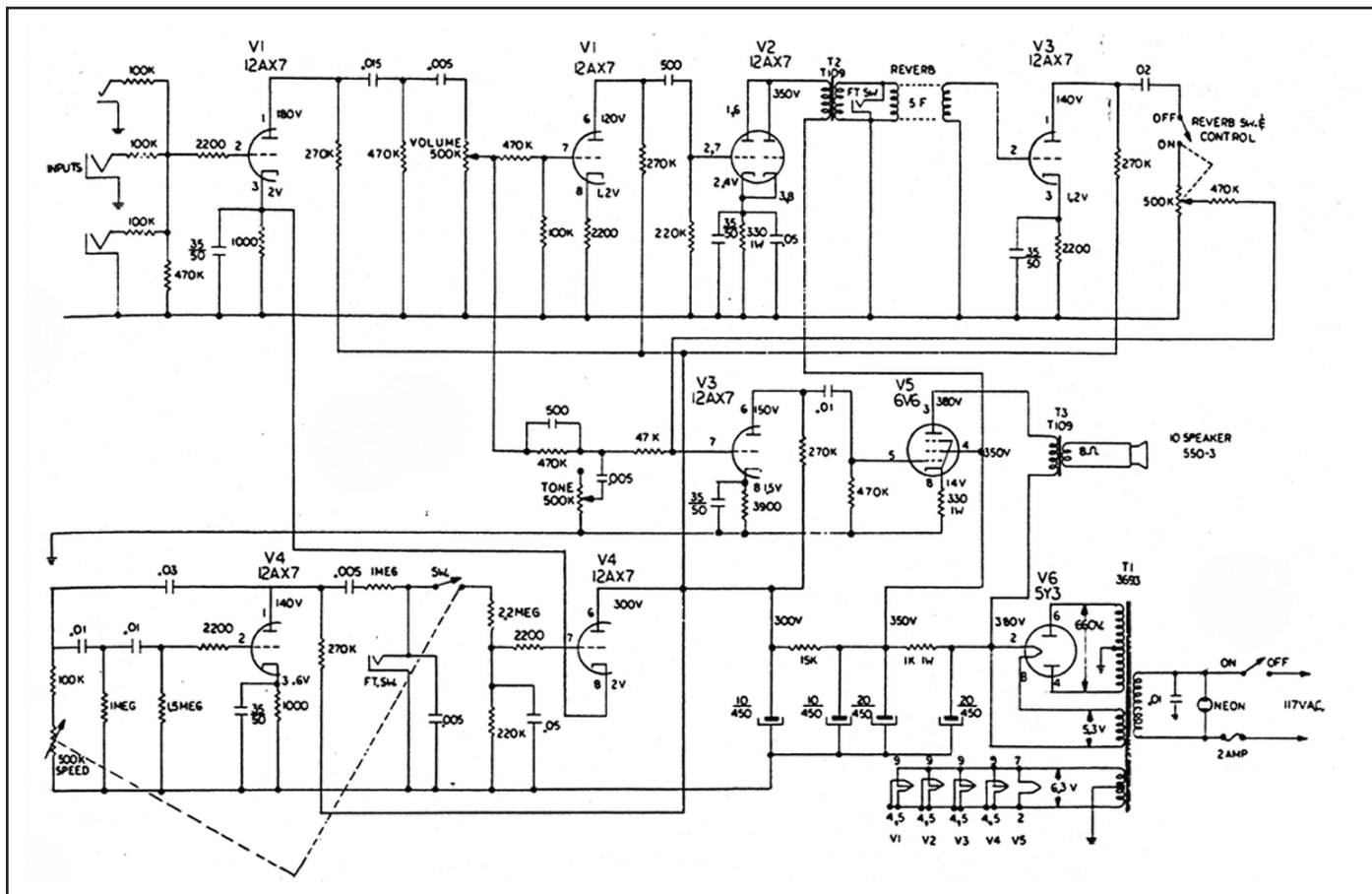


Figure 3: The Supro S6622 had a transformer-coupled supply with a full-wave rectifier.

in many low-priced guitar amplifiers through the mid-1960s, including the Sears Silvertone 1449L “case amplifier.”

AC-Only Full-Wave Supplies

The Supro S6622 shown in **Photo 2** represented a “step-up” amplifier for players who could move out of the “price basement.” As you can see from

the schematic shown in **Figure 3**, the power supply used a 5Y3 full-wave rectifier, with filtering and decoupling circuitry similar to those used in the Montgomery Ward amplifiers mentioned earlier. The filaments of the four 12AX7 dual triodes were paralleled with that of the 6V6 beam power output tube. One characteristic of half-wave-rectifier supplies like those used in AC/DC supplies is poor



Photo 3: The Silvertone model 1484 “Twin Twelve” amplifier was the first amp owned by many serious guitarists.

GETTING THE MOST OUT OF
VACUUM TUBES

By Robert B. Tomer

Vacuum Tubes—The Important 4/5ths of Tube Circuits

Over 80% of all tube equipment defects result, directly or indirectly, from tube failures. Why do tubes fail? What can be done to prevent them from failing before their time? How can you determine whether a tube is good or bad, or how well and how long it will work in a given circuit? Should tubes be replaced periodically, whether they've failed or not . . . or should they be tested every so often, and replaced if indications show them to be below par? This book supplies the answers to these profound questions . . . plus many, many more.

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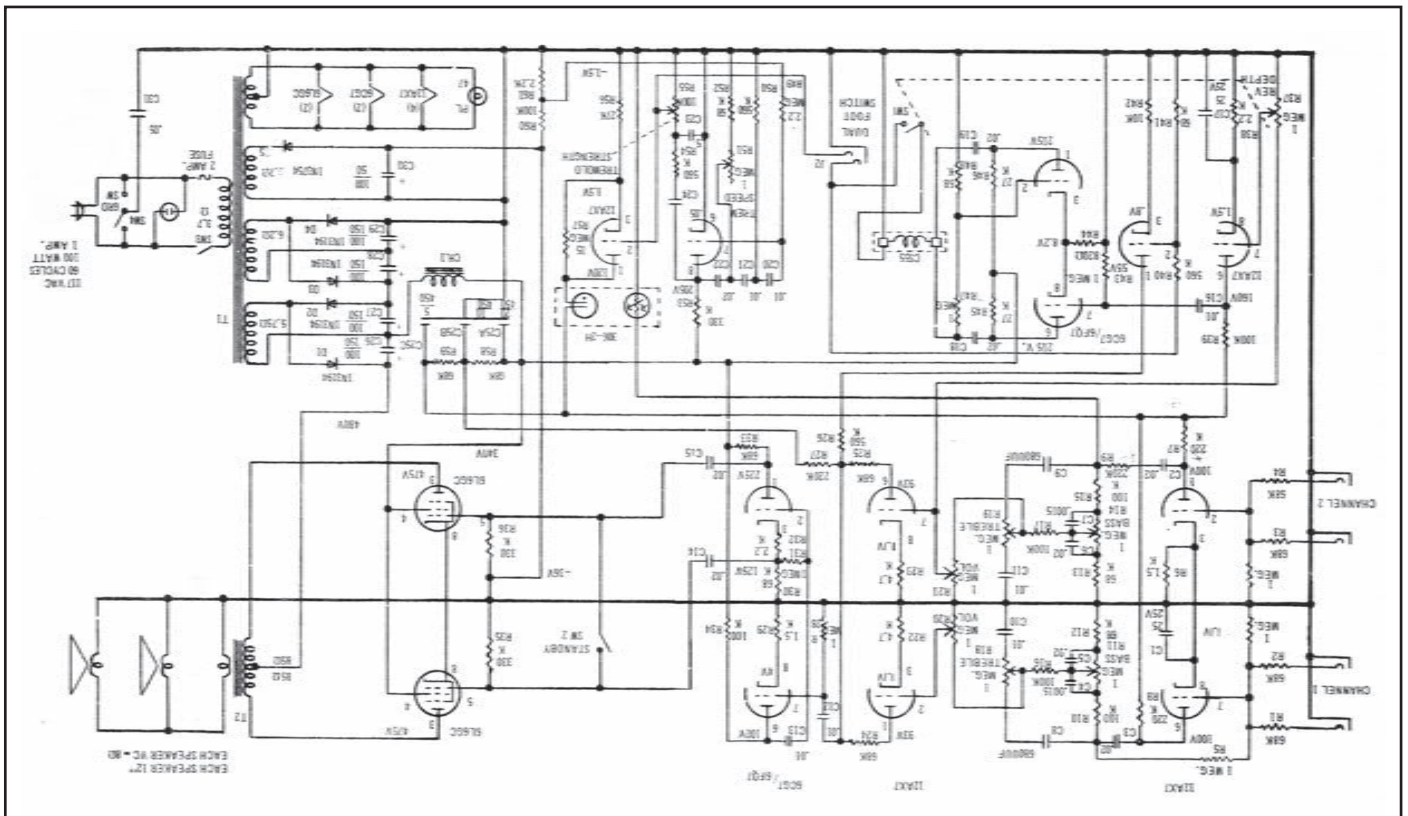


Figure 4: The Silvertone 1484 used a solid-state voltage-quadrupler rectifier.

voltage regulation. This means that although the peak power output may be high—about 5.5 W for a 6V6—the power will drop on sustained chords. This happens because the half-wave rectifier charges the filter capacitor up to the peak voltage once per cycle, which is 50 or 60 times per second, depending upon where you live: once each 16 to 20 ms. When load current is drawn from the power supply, the capacitors discharge through the tubes (mainly the

output tubes) in between charging peaks. Thus, the average B+ voltage drops, reducing available power output. Full-wave-rectifier supplies, such as the one used in this Supro amp, are “stiffer,” providing better voltage regulation.

Voltage-Quadrupler Rectifier Power Supplies

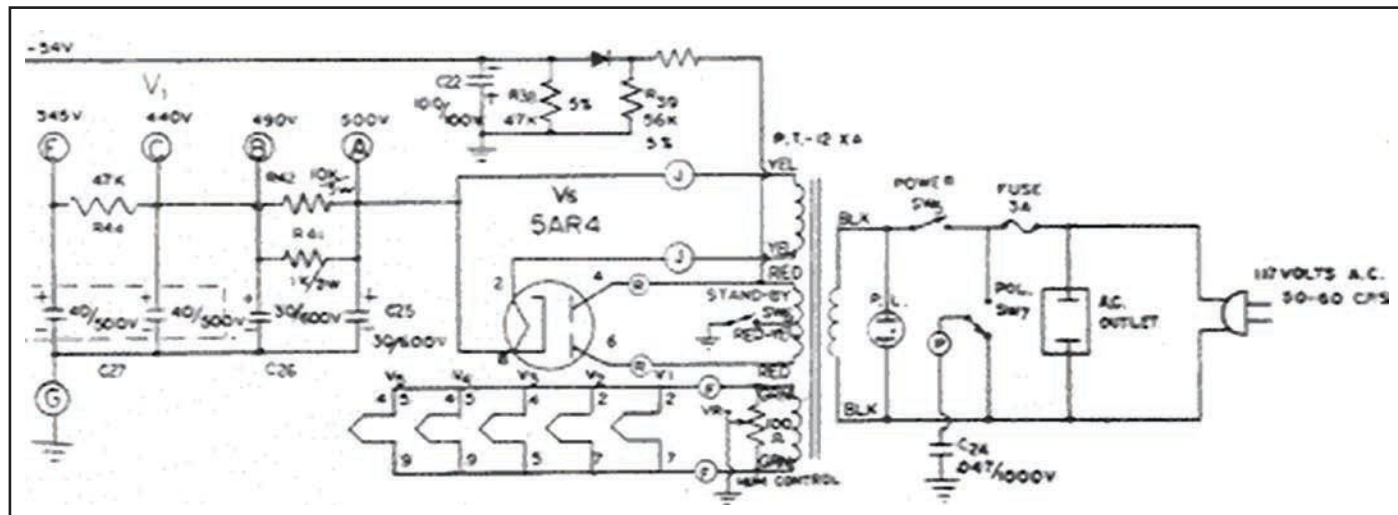
The Silvertone model 1484 “Twin Twelve” amplifier (see **Photo 3**) was a mainstay of high school rock bands of the 1960s. Using a pair of 6L6GC tubes in push-pull, the amplifier had an advertised power output of 50 W. However, possibly to hold down the cost of the power transformer and filter capacitors, this amplifier used a voltage-quadrupler rectifier (see **Figure 4**). This type of rectifier essentially acted as four half-wave rectifiers stacked in series, each charging a separate filter capacitor. The total no-load voltage across the four series capacitors was four times the peak value of the AC voltage on the power transformer’s B+ winding. Since filter capacitors were charged once in each half-cycle, this arrangement constitutes a full-wave quadrupler, and has almost as good voltage regulation as a full-wave rectifier.

In this case, each of the two B+ windings



Photo 4: The Ampeg B25 is a workhorse bass amplifier.

B+ voltage drops to 480 VDC. More than one fledgling technician has learned that if you replace the filter capacitors in the voltage quadrupler with the negative terminal grounded, as is usual in tube



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Photo 5: The Ampeg SVT is one of the most powerful tube amplifiers for electric musical instruments ever produced.

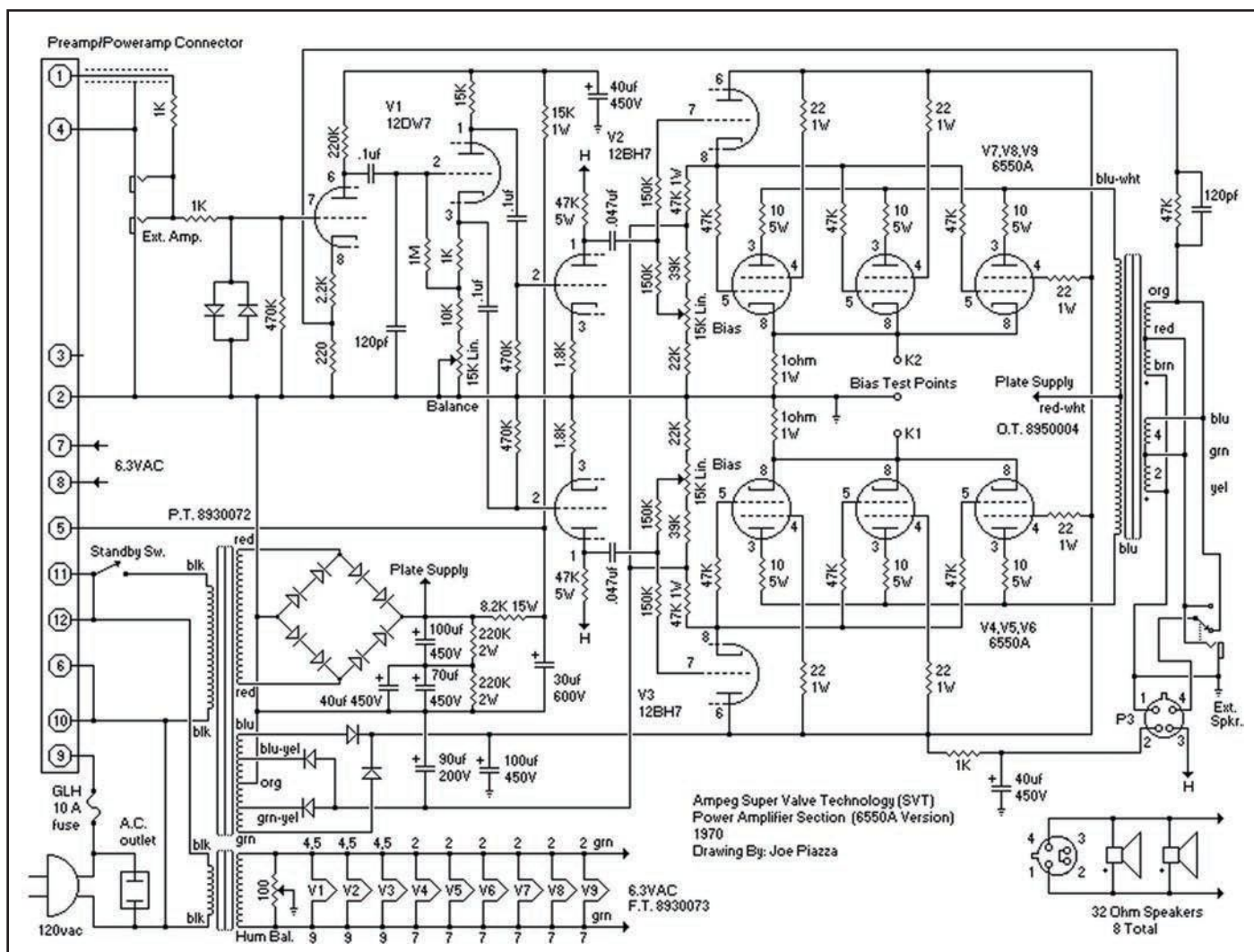


Figure 6: The SVT uses a solid-state full-wave bridge rectifier.

amplifiers, the “magic smoke” comes out of the capacitor and it won’t work anymore. It is essential to pay close attention to installing replacement capacitors in the polarity shown in the schematic.

Since the 1484 is the first amplifier we’ve looked at that uses a push-pull output connection, it is the first one to have a bias supply for the output tubes. This is made up of the bottom-most solid-state diode in the schematic, and the 100 μ F/50 V capacitor. Unlike some push-pull tube bias supplies, this one does not have a potentiometer to adjust the bias voltage after the output tubes are replaced. In the unusual event that the bias filter capacitor shorts or becomes leaky, the output tubes will get too little negative bias, possibly causing excess plate current to flow (leading to “red-plating”) and perhaps even overheating the power transformer.

The 1484’s power supply has another feature that we have not seen in any of the other supplies we have discussed: a filter choke. This choke is fed from a point about three-quarters of the way up the quadrupler capacitor stack, so it acts as a capacitor-input filter.

This type of filter has little effect on voltage regulation, but does an excellent job of cleaning up hum and buzz from the B+ for the preamp tubes. The output stage is less sensitive to hum and buzz in the B+, since the push-pull connection causes these types of noise to cancel out.

Hollow-State Full-Wave Rectifier Power Supply

The Ampeg B25 (see **Photo 4**) uses a 5AR4 full-wave rectifier tube feeding a capacitor filter (see **Figure 5**). The 500 V output feeds the output tubes: a push-pull pair of 7027As. The 490 V output feeds the output-tube screen grids. The 440 V output feeds the phase inverter, and the 345 V output feeds the preamp. The -54 V output is the grid bias supply for the 7027As, and is not adjustable.

Solid-State Full-Wave Rectifier Power Supply

There was a time when the sound systems used for vocals were not used for amplifying guitars, basses, and other

instruments. In those days, guitarists and bassists of loud bands just had to suck it up and carry very heavy amps.

One of the best of those super amplifiers was the Ampeg “Super Vacuum Tube” (SVT) amplifier (see **Photo 5**). This behemoth used six 6550 output tubes in parallel/push-pull, and was rated at 300 W continuous output into a 2 Ω or 4 Ω load, with total harmonic distortion (THD) under 3%. Frequency response was rated flat ± 1 dB from 20 Hz to 20 kHz. A complete SVT system included the head plus two speaker cabinets, each loaded with eight 10”, 32 Ω bass speakers connected in parallel for a total impedance of 4 Ω per cabinet, or 2 Ω for both cabinets paralleled.

To provide such superlative performance, an amplifier must have an excellent power supply. As you can see from the SVT schematic shown in **Figure 6**, the power supply uses a full-wave bridge rectifier. Series pairs of diodes are used in the bridge, because the B+ for the 6550s is 660 V.

Achieving a safety factor of three times the peak transformer voltage would require either two 1 kV diodes in series or single 2 kV diodes. The pairs of 1 kV diodes are easier to source and less expensive. Also, because of the 660 V high B+, a series pair of 100 μ F/450 V filter capacitors is used. The 462V supply is filtered using a 30 μ F/600 V capacitor, and more common 450 V capacitors are used for the lower-voltage outputs.

A separate full-wave rectifier is used to supply B+ to the 12BH7 bias regulator. The -150 V grid bias for the 6550 tubes is supplied via a filtered half-wave rectifier. Bias adjustment for the output tubes is performed via controls VR1 and VR2, which are accessible from outside the chassis once the cabinet baffle is removed.

Despite not using a filter choke, the SVT power supply is very clean and stiff—a must for a bass amplifier having good sustain. Although not many hobbyists understand the importance of the power supply in establishing the sound and feel of a guitar or bass amplifier, the engineers who have designed the best amplifiers have fully understood the effect of a really good power supply. 



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June 2–4, 2017

Los Angeles Audio Show (LAAS)

Sheraton Gateway Hotel, Los Angeles, CA

www.laaudioshow.com

The Los Angeles Audio Show (LAAS) is a new high-end audio and home-theater show debuting in June 2017, and every weekend each year following Memorial Day. The LAAS is presented in collaboration with the Los Angeles and Orange County Audio Society at the Sheraton Gateway across from the Los Angeles International Airport with complimentary shuttle service. The LAAS is a presentation of The Orion Group—a private investment firm—and is managed by Marine Presson, Show Director, formerly of T.H.E. Show. After the 2016 edition of T.H.E. Show in Irvine, the promoters of that show decided to move the event to the Anaheim Hilton in September (it is now called T.H.E. Show Anaheim.) The Los Angeles and Orange County Audio Society joined forces with The Orion Group to promote the new LAAS.

The Los Angeles Audio Show's three-day experience (the first day is not open to the public) includes five hotel floors of private listening rooms and booths by leading brands showcasing their newest products as well as educational seminars produced by the LAOCAS featuring highly recognized guest speakers. A central focal point for the new show will be a "HeadGear Emporium," a 7,000 ft² space dedicated exclusively to headphones and other personal listening devices.

June 7–9, 2017

CES Asia

Shanghai New International Expo Center (SNIEC)

Shanghai, China

www.cesasia.cn

The exponential growth of CES Asia is proof that it is a must-attend consumer tech event for buyers, investors, exhibitors, and the media. Exhibit space has increased more than 200% to debut groundbreaking technology and more than 250 notable companies are already slated to exhibit.

The three major themes of CES Asia 2017 are Connectivity, Innovation, and the Internet of Things (IoT)—topics that are on everybody's minds in this fast-moving industry. For 2017, four new trending topics and product categories have been introduced. Virtual Reality, Augmented Reality, The Internet of Things (IoT), and Drones.

Introduced at CES Asia 2016, the Startup Park will be also be expanded in 2017. Some 50 conference sessions and keynote addresses will provide an international platform, covering some of today's most notable topics.

June 15–17, 2017

2017 AES International Conference on Audio Forensics

Holiday Inn Arlington At Ballston, Arlington, VA

www.aes.org/conferences/2017/forensics

The Audio Engineering Society (AES) International Conference on Audio Forensics is dedicated to exploring advances in the field of Audio Forensics by providing a platform for research related to the forensic application of speech/signal processing, acoustical analyzes, audio authentication, and the examination of methodologies and best practices. This is the sixth AES conference devoted to the technical developments and practical approaches developing in the field of Audio Forensics. With increased participation at each offering, this conference has become the hub for researchers and practitioners to exchange ideas and foster new approaches to address problems that are evolving every day. The conference program will include paper presentations and discussions as well as Workshops on topics related to Forensic Audio.

June 14–16, 2017

InfoComm 2017

Exhibition: June 14–16

Conference: June 10–16

WEST building, Orange County Convention Center (OCCC), Orlando, FL

www.infocommshow.org

InfoComm International is expecting a strong 2017 edition in Orlando, FL. Working in partnership with IMCCA, InfoComm will host a day of Emerging Trends education on Tuesday, June 13, which will feature the inaugural class of Emerging Trends Fellows—experts chosen for their knowledge of a cross-section of critical technology areas.

InfoComm 2017 will have a new Center Stage, located on the exhibit floor, where AV industry thought leaders will showcase case studies and real-world AV solutions to customers' enterprise technology challenges. Also new this year is TIDE (Technology. Innovation. Design. Experience.), a one-day conference examining the strategic impact of AV technology. InfoComm 2017 will also introduce the new Immersive Technology Pavilion, an area dedicated to augmented reality, virtual reality (VR) and 360° video solutions. The new pavilion will include hands-on experiences of application-specific VR environments.

June 22–24, 2017

2017 AES International Conference on Semantic Audio Erlangen, Germany

www.aes.org/conferences/2017/semantic

Tutorial Day: June 21, 2017

The third edition in the series of AES conferences on Semantic Audio will be jointly organized by the Fraunhofer Institute for Integrated Circuits IIS, the Friedrich-Alexander-Universität Erlangen-Nürnberg (FAU), and the International Audio Laboratories Erlangen (AudioLabs), Germany. The IIS venue can host more than 100 participants, providing a large lecture room, poster areas in the foyer, additional rooms for e.g. breakout/author's room, Wi-Fi access and in-house catering.



Article Update

Our May 2017 *audioXpress* issue featured an article by Fabio Camorani titled "Kismet 5: The 20th Anniversary Edition: The Evolution of Its Design." After the publication of the Kismet 5 article, Walt Jung contacted Camorani with an important note he would like to share with our readers. Jung noticed that Camorani had not put any capacitors at the output of LT1084 IC.

Camorani used these ICs as constant current sources (CCS) and did not see any oscillation in his specific application, but this may not be the case for everyone and you may want use an output capacitor as the datasheet recommends. Without the capacitor, the regulator could oscillate. Camorani suggests using: a 150 μ F electrolytic capacitor or 22 μ F tantalum one. Of course, this will impact the sound so he recommends using a high-quality capacitor—possibly an MKT one?



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