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A New Frontier for Audio

With this edition of *audioXpress* scheduled for distribution at important events such as the 143rd International Audio Engineering Society Convention in New York; the Loudspeaker Sourcing Show 2017, an excellent gathering of loudspeaker component and turnkey system suppliers in Guangzhou, China; and the 14th edition of the Rocky Mountain Audio Fest (RMAF) in Denver, CO, I feel that I must address the trending topic in the audio industry: Smart Speakers.

As I wrote in my previous *audioXpress* editorial, I believe we should look at this new class of audio products as "connected speakers." But because of the wide implications for the consumer technology space, the marketing departments have already decided to call it "smart speakers" and there will be little we can do to change that. Still, I believe it is important to highlight the "connected" part because that's what will distinguish a completely new generation of audio products—not just speakers. Connectivity is what makes voice control possible and it is essential for artificial intelligence and machine-learning engines.

Those are key aspects to distinguish what we have seen up to now in speakers that combine a powerful DSP, microphone arrays, all types of sensors, and wireless transmission, making it possible to have not only whole-home audio but also room acoustics compensation and real-time adaptable processing for different types of audio signals and entertainment experiences. Add to this the connection to the Internet, cloud services, and to Voice Personal Assistants (VPAs) and suddenly there's an entirely new exciting "connected" frontier to explore.

"We are the beginning of a wider market transformation, with VPAs at the heart of a new CE land grab," says market research company Futuresource Consulting (www.futuresource-consulting.com) in its recently released report about smart speakers. In the US, smart speakers now account for nearly one out of every three shipments in the home audio hardware category. Globally, in the first quarter of 2017 alone, 3.4 million smart speakers were sold, with a retail value of more than $440 million, the study indicates. And even though the main impact of smart speakers is being felt for now primarily in the US and the UK, Futuresource forecasts that the potential for VPAs to be the new go-to consumer interface, particularly in the burgeoning smart home sector, "is probably the biggest opportunity of the decade." According to Futuresource, "Worldwide home audio revenues grew by 20% in 2016 to $11.7 billion (at trade) and are on course to grow a further 17% in 2017, benefiting from the success of smart speakers."

Global Market Insights also projects the global smart speaker market to register a massive Compound Annual Growth Rate (CAGR) of 50% over 2017-2024 (https://www.gminsights.com/industry-analysis/smart-speaker-market), and forecasts the rushing in of Chinese competitors in smart speakers to diversify the product landscape immensely, with India and Japan to also witness substantial gains in the years ahead. "Of late, technological interventions have allowed manufacturers to introduce products that are not confined only to playing music, but have extended their applications to problem-solving, setting reminders, streaming of news and music with a single voice command. Integration of virtual assistant software has played a pivotal role in reshaping the business model." The market research firm projects that smart speakers will represent a revenue of USD $13 billion by 2024.

Another important study, Qualcomm's "The State of Play 2017" report (https://assets.qualcomm.com/the-state-of-play.html), based on a survey of 4,000 adult smartphone users in the US and the UK indicates that almost half (49%) of all consumers surveyed already own a wireless speaker, and more than 40% of respondents, indicated that they would like to have several wireless speakers networked around their home. The study finds that there is significantly more interest in voice control and personal assistants in 2017 compared to 2016, and 46% agreed that in the future most people will own a smart speaker.

This new frontier will, in my opinion, represent a major opportunity for the audio industry, considering that this new level of integration and services is still in its infancy. Most importantly, there are huge career opportunities in the audio industry. With all the semiconductor and digital signal processing companies currently working on new "smart speaker" projects for companies that have never developed an audio product, and all sorts of consumer electronic companies showing ambitious plans in this domain, there's a never-ending world of opportunities for recently graduated engineers, seasoned veterans, and even external consultants.

João Martins
Editor-in-Chief

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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.

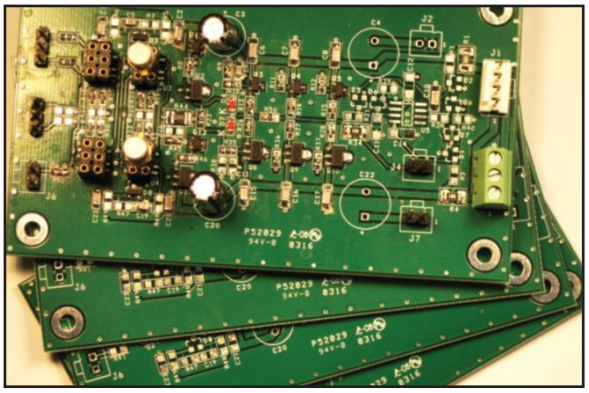
Steve Tatarunis has been active in the loudspeaker industry since the late 1970s. His areas of interest include product development and test engineering. He is currently a support engineer at Listen, in Boston, MA, where he provides front-line technical support to the SoundCheck test system's global 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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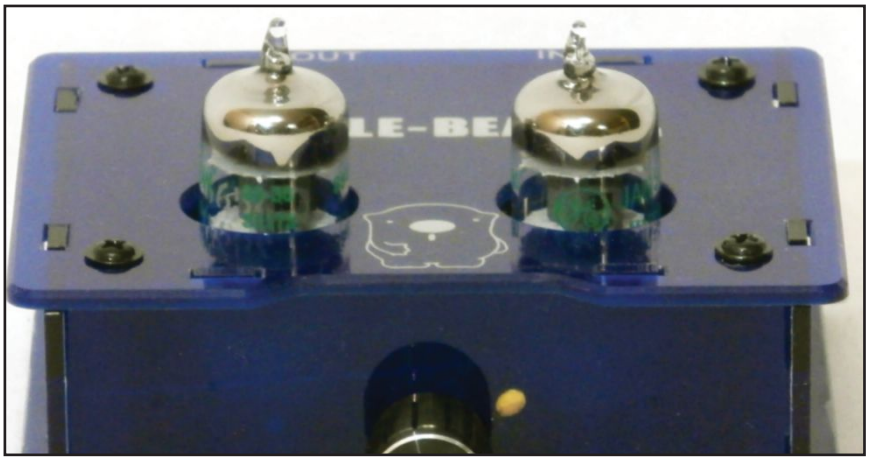
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Many audio power and preamplifier designs benefit from the use of voltage regulators. This article focuses on linear regulators and how feedback networks affect their performances, particularly in the higher voltage applications that vacuum tube amplifiers demand.

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Vinyl vs. CD (Part 1)

Measuring the Sound Difference

Several people ask themselves the question: "If a vinyl LP record sounds different, can I make a CD sound like a vinyl LP?" The answer is no, but there are ways to get rather close. But the basic question is: "Why do they sound different?" In this article, we will explore those differences.

By
Ron Tipton
(United States)

The conjecture is the higher second-harmonic distortion from the LP recording adds a "richness" not heard from the CD. Although the CD is technically superior with its better frequency response, dynamic range, and signal-to-noise ratio (SNR), many people enjoy, the LP's "mellowness." This preference is probably the primary force driving the resurgence of LP production and sales. Subjective listening tests are fine, but can this difference be measured? I'll start by describing a measurement experiment I devised to try to answer this question and then apply my measurement method to several CD play back setups. Maybe we will learn something interesting.

A Measurement Experiment

Figure 1 shows my experimental setup's block diagram. I inserted amplifiers as shown to set the recorded voltages well above the noise level. Music from an LP or CD is directed to a bank of bandpass filters with outputs at three fundamental and three second harmonic frequencies. The filter outputs are connected a pair at a time, fundamental and second harmonic, to a pair of full-wave detectors followed by active low-pass filters with a 20 Hz cut-off frequency. A pair of identical true RMS voltmeters (Tenma 72-1015), set to measure DC voltage, is connected to each detector output. Bandpass filters 1, 2, and 3 are used

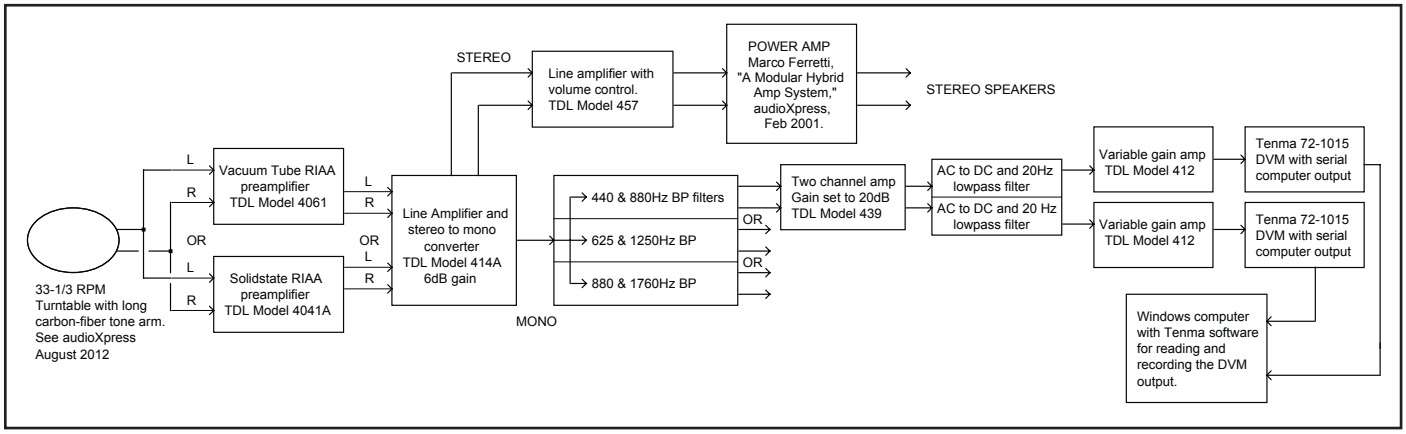
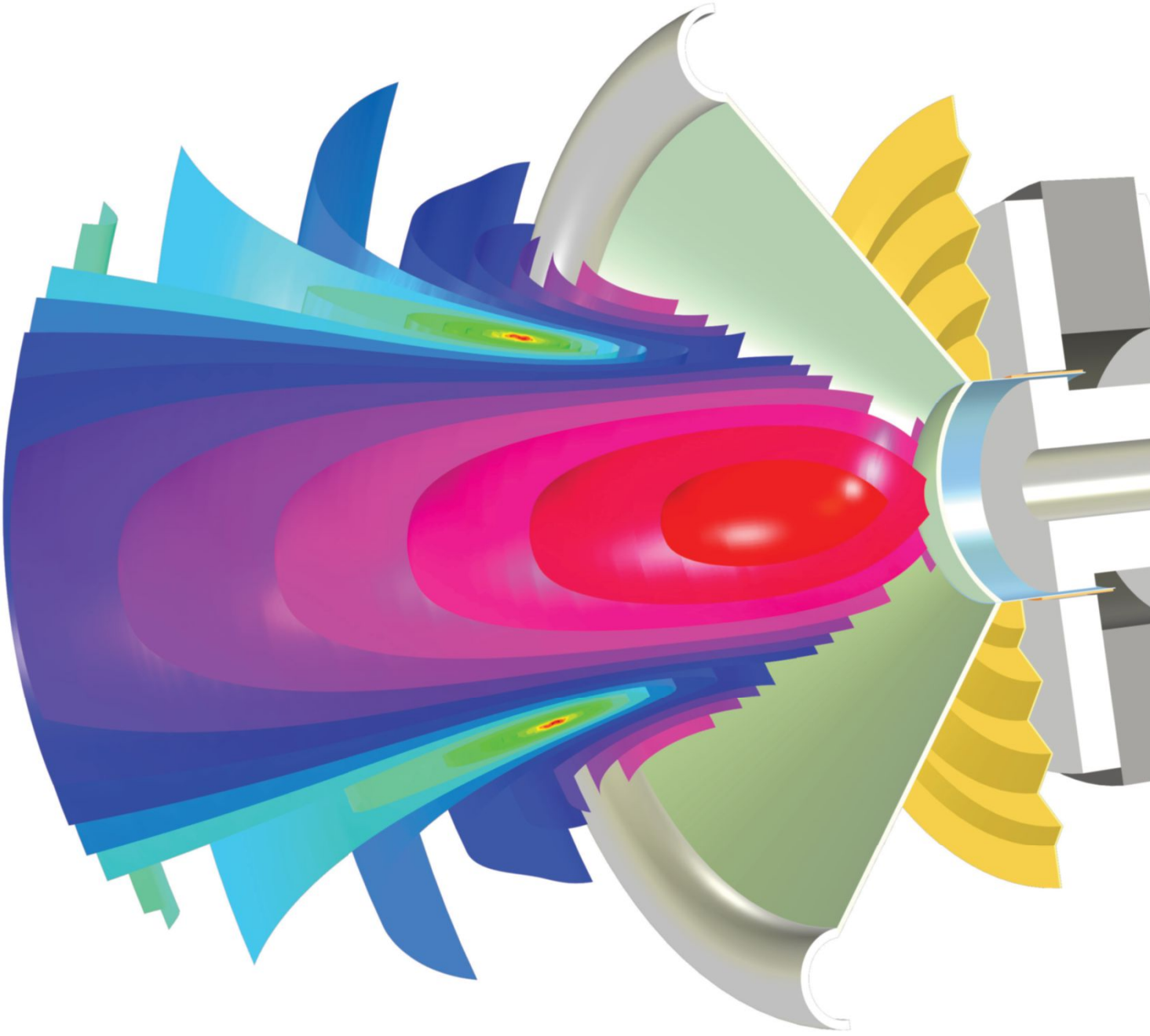


Figure 1: This block diagram shows the setup for measuring the LP record harmonic ratios. The turntable and RIAA preamps are replaced by a CD player for the CD measurements. The stereo to mono converter sums the left and right channels, both with unity gain.



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Music Source	440 & 880 Hz Filters RMS Energy Ratio	625 & 1250 Hz Filters RMS Energy Ratio	880 & 1760 Hz Filters RMS Energy Ratio
LP with TDL 4061B vacuum tube preamplifier	0.794	0.554	0.690
LP with TDL 4041A solid-state preamplifier	0.777	0.507	0.668
CD with Sony DVP-NS75H DVD-CD player	0.798	0.510	0.626
CD with Sony player into "Little Bear" vacuum tube line amplifier, unity gain	0.729	0.403	0.648
CD with Sony player into "Little Bear" vacuum tube line amplifier, 7 dB gain	0.849	0.443	0.608

Table 1: Five music sources are shown with their RMS energy ratios for the three sets of fundamental and second harmonic bandpass filters used in this experimental measurement. The high values shown in red indicate a higher amount of second harmonic energy compared to the fundamental energy.

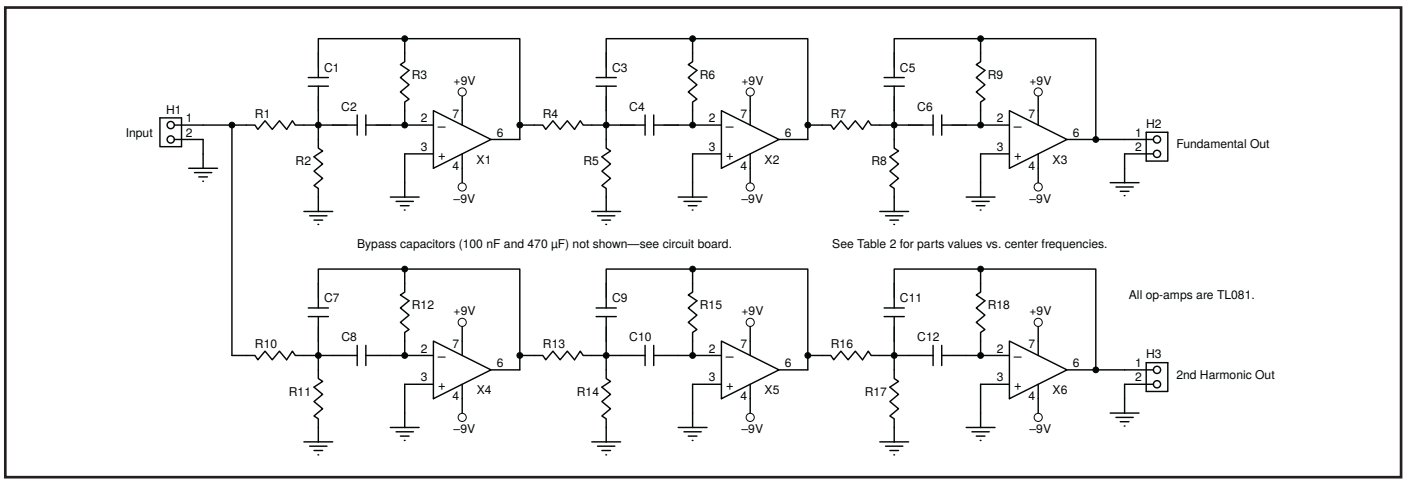


Figure 2: This circuit is repeated three times on the BP6-Filter board. The input header, H1, feeds all three filters. The parts values are listed as Table 2 in the Supplementary Material section of the audioXpress website.

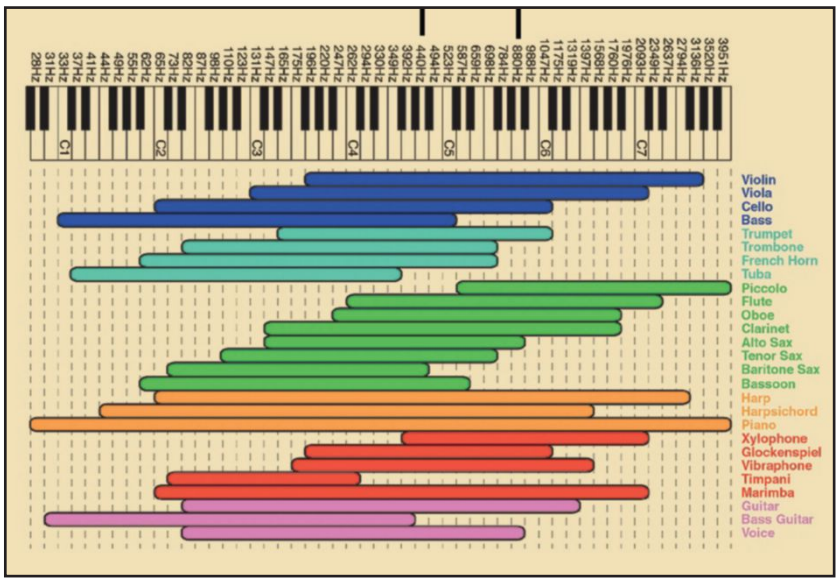


Figure 3: This drawing shows the frequency range of a variety of musical instruments and the human voice, which seems to be a combination of male and female ranges. The two black bars at the top indicate the low and high fundamental center frequencies of my bandpass filters.

in turn and an LP or CD track is played. Of course, the LP and the CD must contain the same music preferably made from the same master recording. I believe this was the case for my choice: the Diana Krall recording "Quiet Nights." I already had the CD so I purchased a virgin copy of the LP. I used the first two tracks: "Where or When" and "Too Marvelous for Words," for a playing time of about 8.1 minutes.

The Tenma meter is useful because it has a data recording feature: 0 to 6 VDC with 1 mV resolution. By using a USB cable and the supplied computer software, the output voltage can be sampled and saved as a Microsoft Excel spreadsheet file or as a text file. Calculations can be easily done in the spreadsheet.

The Tenma DVM can sample and record as fast as 1 sample per second for about a 487 sample file for each playback. The next question to answer: What to measure? The average value of each file is easy to calculate but is it the best metric? After some consideration, it seemed energy was better so I used the average RMS value. That is, find the sum of the squares, divide by the number of samples, and then take the square root. The Microsoft Excel

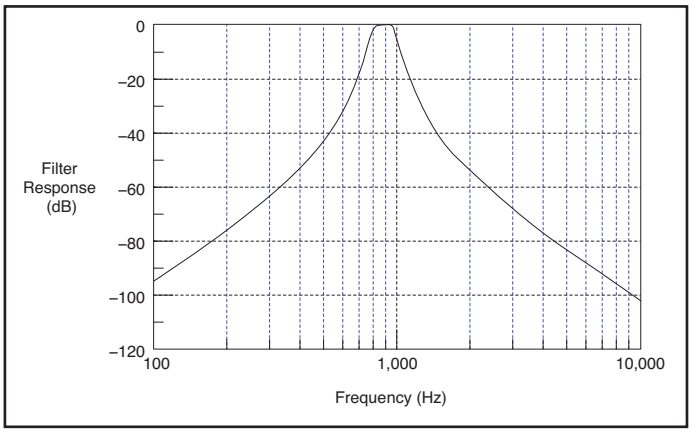


Figure 4: This simulated frequency response curve was produced by TopSpice from a Figure 2 SPICE net list. A copy of the list is included in bp6-filter.zip in the Supplementary Material. A six-pole filter was chosen as a compromise between bandwidth and circuit complexity.

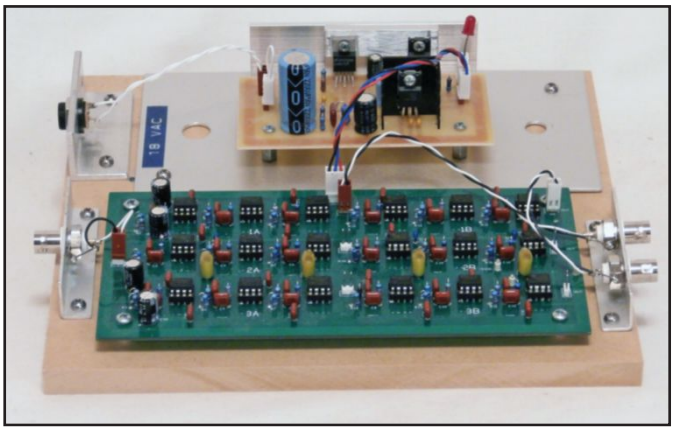


Photo 1: The six bandpass filters I used for my ratio measurements easily fit this 7.75" × 3.9" circuit board. I used op-amp sockets to make it easy to change the IC type if needed, but the TL081s work fine. The smaller circuit board supplies the regulated ±9 V.

spreadsheet makes this easy because of its built-in =SUMSQ function: It squares each data value and sums them.

The numbers in **Table 1** were calculated by finding the RMS value for the fundamental and second harmonic filter outputs and the dividing the second harmonic RMS by the fundamental RMS. Thus, the Table 1 unit of RMS Energy Ratio, with higher values indicating more second harmonic energy.

The Bandpass Filters

Each bandpass filter is a six-pole Butterworth response, realized as an active multiple-feedback design as shown in **Figure 2** with the parts values listed in the Supplementary Material (as Table 2), which can be found on the audioXpress website (www.audioxpress.com).

The fundamental filter frequencies (440 Hz, 625 Hz, and 880 Hz) were chosen because they are prominent in many musical instruments and the human voice (see **Figure 3**). And, according to the Fletcher-Munson curves, they are also tones we hear very well. The -3 dB bandwidths vary from 100 Hz for the lowest frequency filters to 400 Hz for the highest. For all filters the upper -3 dB frequency divided by the lower -3 dB frequency is between 1.23 and 1.26. **Figure 4** shows a simulated 880 Hz center frequency response.

I designed a circuit board for the six filters (see **Photo 1**). I have included details about the BP6 Filter (bp6-filter.zip) in the Supplementary Material. Built with ±1% metal film resistors and ±5% film capacitors, the response curves differ from each other and from the simulation shown in Figure 4. But this doesn't matter because the information we seek is the ratio of the fundamental to the second harmonic outputs and the filter responses remain the same for all the measurements.

The Detector (AC to DC Converter)

The FullWave detector circuit was published in a National Semiconductor application note. I designed a circuit board with the detector's output going to an active four-pole, Butterworth low-pass filter with a 20 Hz cut-off frequency (see **Photo 2**). The circuit diagram is shown in **Figure 5** and the PCB design file (fullwave.zip) is included in the Supplementary Material.

The board has space for jumper-selected averaging capacitors at the detector output but the settling time is faster with the low-pass filter so I used its output for all the measurements. The output DC voltages are all positive.

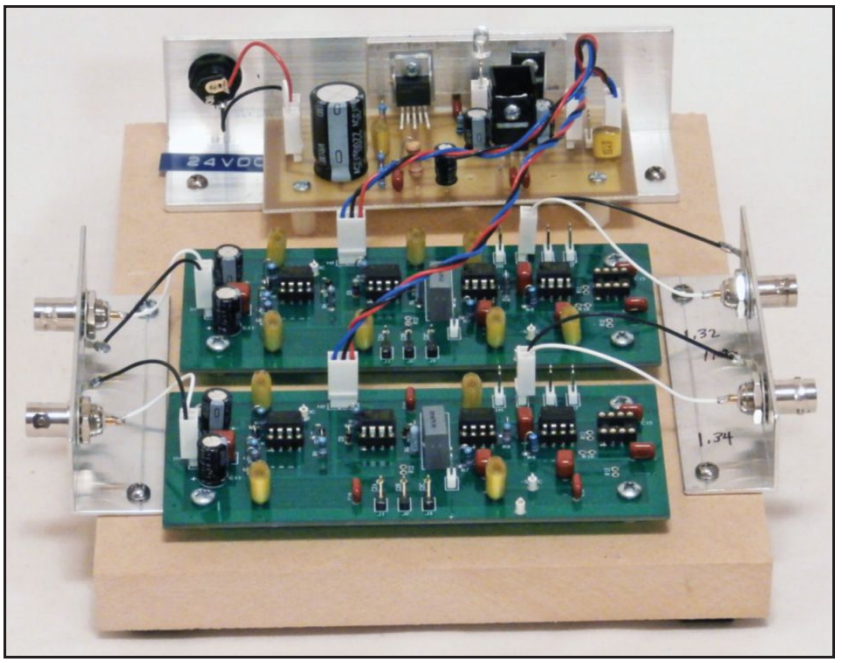


Photo 2: I used a pair of full-wave detector boards to simplify making the ratio measurements. The two left-side op-amps provide the rectification and the other two op-amps, the low-pass filtering. The last DIP socket is unpopulated because I didn't need the optional output amplifier.

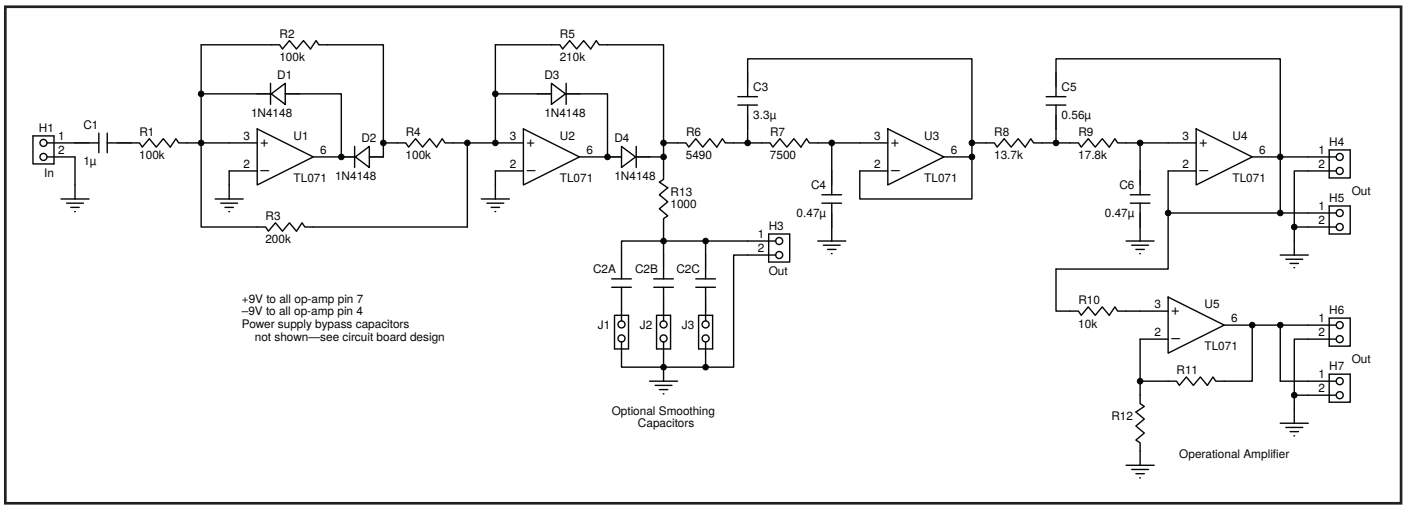


Figure 5: This full-wave precision rectifier has a positive DC output and its settling time is much faster using the 20 Hz low-pass filter output as compared to using smoothing capacitors (single-pole filter). All resistors are ±1% metal film and capacitors are ±5% film except for C3. This 3.3 µF is a ±10% film capacitor currently available from Digi-Key Corp.

Finding the LP BaseLine

The first measurement set is for the LP record to establish a baseline or "goal" for comparison with various CD playback setups. The LP record values are in Table 1. For Row 1, I used an RIAA response vacuum tube preamp (TDL model 4061B) and for Row 2, an RIAA response solid-state preamp, (TDL model 4041A). The results are interesting. The vacuum tube ratios are all higher than those from the solid-state preamp.

As you can see in Figure 1, the stereo listening volume is independently adjustable from the mono level going to the bandpass filters. This made level setup much easier. For the CD Comparisons, I'll start with a CD player and measure its stereo analog output converted to mono as shown in Figure 1.

Project Files

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

Resources

"How to Make my CDs Sound Like Vinyl?" AudioCircle, www.audiocircle.com/index.php?topic=49926.0.

Sources

Little Bear P5-1 Blue line amplifier
Shenzheng Cavins Technology Co., Ltd. - Douk Audio | www.doukaudio.com

Tenma 72-1015 DVM
MCM Electronics | www.mcmelectronics.com

DVP-NS75H CD/DVD player (obsolete model but available used from a variety of dealers)
Sony | www.sony.com

The CD Player

When the CD was "invented," its music content was technically superior to all other media except the original magnetic tape masters. But early players left a lot to be improved. The music was often described as "brittle" or "harsh" because many players had digital-to-analog converters (DACs) with only 12 bits of resolution so the upper four bits were not reproduced.

Players have been greatly improved so even the lower priced models have DACs with a true 16-bit output. I used a Sony model DVP-NS75H CD/DVD player with its analog output going to the model 414A input as shown in Figure 1. These numbers are in Row 3 and are somewhat puzzling with the 440 to 880 Hz ratio larger than the LP ratio in Row 1. Perhaps the third decimal place isn't warranted, but they would still be equal. I'm compelled to mention that several of the measurements were repeated at least once for verification.

I didn't know what this experiment would show but some of the ratios do seem inconsistent. Perhaps its explanation is in my choice of filter frequencies. Maybe adding an additional three, six, or nine filter boards would clarify the results—but maybe not.

Now, what happens if I add a vacuum tube amplifier between the CD player and bandpass filters?

The "Little Bear"

This twin-tube stereo line amplifier (see **Photo 3**) is inexpensive and has a wide bandwidth, low distortion, and low noise. As recommended by more than one online forum, I replaced the supplied tubes with a pair of closely matched JAN 5654Ws. For technical details, see littlebear.zip in the Supplementary Material.

With its volume control set to unity gain, I placed the Little Bear between the CD player output and the model 414A input. The ratios are shown in Table 1, Row 4. The 880 to 1760 Hz ratio is now larger than the un-buffered CD ratio shown in Row 3, but the other ratios are smaller.

Increasing the Little Bear gain to its maximum of 7 dB (volume control fully CW) resulted in the ratios shown in Row 5 of Table 1. (I did have to readjust the setup to compensate for the added 7 dB.) Now the 440 to 880 Hz ratio is significantly larger than the LP ratio shown in Row 1. (I repeated both of these measurements with the same results.)

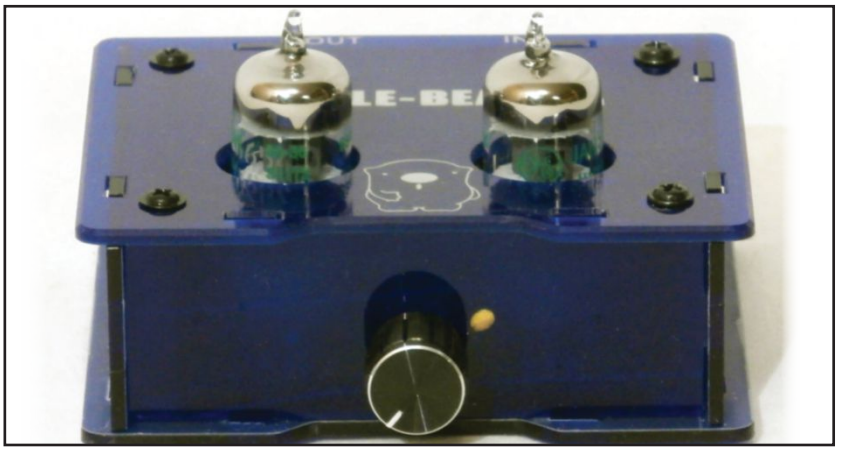


Photo 3: The Little Bear amplifier is just 3.9" wide by 3.5" deep. The front knob is the volume control with an on-off switch. I added the yellow dot to indicate the unity gain position. The stereo RCA input and output connectors and the power in connector are on the rear panel. The amplifier is furnished with a 12 VAC wall power supply.

Some Final Words

As I mentioned earlier, maybe additional filter frequencies would clarify these results. Nevertheless, these ratios strongly suggests that increased second harmonic energy probably accounts for an LP's—or a CD's played through vacuum tubes—mellowness.

Next time, I'll be reporting on playing digital music (CD quality) from a computer into combinations of DACs and vacuum tubes and solid-state preamps. (I'm planning to add another filter board with fundamental center frequencies at: 362 Hz, 535 Hz, and 765 Hz.) And then I'll have some information to share on using VST plug-ins (e.g., the PSP Vintage Warmer) with media players (e.g., JRiver Media Center) that support VST.

It will probably require some experimentation, but I think with these tools at hand you will be able to tailor your music to the sound you prefer. Please stay tuned.



Good Memories! Essential Knowledge?



After browsing through this book's content and extensive index (almost 13 pages!), I took it with me for a brief vacation. I was curious to read the chapter about the Compact Cassette decks, because there aren't many books covering equipment, which was in fact being extensively used just 20 years ago—and often needs maintenance. I was also curious about why it would be called "Retro Audio."

Review By
João Martins
(Editor-in-Chief)

Repairing analog and earlier digital audio equipment is increasingly a necessity as there are less experienced technicians to repair it, and unfortunately many of the manufacturers are no longer in business. While the market for hi-fi is decreasing globally, there is also an increased interest in collecting and restoring audio equipment, sometimes available cheap in non-working condition and worth considerable money when well preserved.

Fortunately, many electronics enthusiasts are discovering the pleasure of restoring those audio jewels, while the revival of vinyl also renewed the interest for restoring existing turntables. As the appreciation for older well-preserved working pieces increases (e.g., tube radios and tube amplifiers, turntables, amplifiers, and tape recorders/players from the 1970s and 1980s), they are also increasingly difficult to maintain. Indeed, sometimes it is easier to find a schematic or manual from the 1950s or 1960s, than it is to find any literature on relatively recent products from the 1980s. So, the idea of a Service Guide for consumer audio equipment seems to be timely.

As Paul Hetrelezis writes in the book's preface and is highlighted in the back cover, the book was written from the perspective of a workbench technician and electronics hobbyist, including many tables of useful data, servicing tips, and supplementary notes of the author's knowledge—accumulated over decades of servicing cassette, record players, and CD players. As Hetrelezis explains, the book intends to cover electronic servicing and also mechanical issues.

The book is structured into three areas, starting with the workshop, equipment, and working environment (Chapters 1–3), followed by basic diagnostic approaches and the repair process (Chapters 4–5)—taking practically the first half of the book. Finally, Hetrelezis dedicates the remaining five chapters to "specific items of repair" with tables of diagnostic faults and common likely causes, on tuners, amplifiers, cassette decks, turntables, CD players, speakers, and microphones, respectively.

The book even includes some supposedly useful appendixes, including common audio connector pinouts, resistors, and capacitors charts, semiconductor outlines, Japanese transistor codes,

Retro Audio: A Good Service Guide
By Paul Hetrelezis
Publication Date: April 2017
Format: Soft cover
Pages: 198
ISBN 978-1-907920-58-5
Elektor Publishing: www.elektor.com/retro-audio

precautions, safety tips, and more.

I found the section on components in Chapter 2, one of the most useful, providing a great quick reference guide. There are descriptions of passive components, resistors, capacitors, inductors, diodes, transistors, and so forth written in the perspective of how to identify them, what they are used for, and how to look parts for if they need to be replaced.

As with everything else in the book, don't expect in-depth technical information about anything. In fact, probably the most complete and relevant section in the book are the initial diagnostic testing procedures in Chapter 4, because they truly reflect the author's experience and provide good basic test and measurement advice (including calibration of classic test and measurement equipment).

This is a book dedicated to some of the most common and fairly "modern" items, not covering older "vintage" categories and tube equipment—in that regard the title is slightly misleading. Readers interested in vintage electronics, should look instead for the excellent *Retronics* book, a compilation of articles by Elektor editor Jan Buiting, published between 2004 and 2012 (www.elektor.com/retronics), or some of the many books on tube amplifiers that Elektor also published.

OVERALL IMPRESSION

As Elektor describes it, *Retro Audio* is supposed to be a "hands-on" service guide. Yet, the result turns out to be slightly different from the author's intention.

This is not a very technical book, and any electronics professional or audio enthusiast familiar with audio equipment in general will be disappointed. I certainly didn't learn anything new about cassette decks.

What this book is good for is to serve as an introduction and reference work about consumer audio equipment and a sort of "condensed guide" to basic electronics knowledge—for anyone who knows very little about the technical subjects.

As an introductory guide to essential test equipment and tools, with basic information about repair and obtaining spare parts, and a little of DIY experience—this book can be useful. For anyone starting with audio electronics, the general diagnosis and testing tips, understanding components and electrical properties, how to set up a workbench, and advice on soldering/desoldering work, will be a perfect introduction.

Unfortunately, the final five chapters on tuner, amplifiers, turntables, and CD players provide just basic guidance and will be of little use for any specific practical work. The very brief section on speakers and microphones—almost Wikipedia-like—is particularly disappointing.

For any experienced audio and electronics technician—as it is likely with anyone who decides to embrace repairing analog audio equipment—I don't think this book is going to be of much use, except for confirming the obvious. For students and general audio enthusiasts starting out, the book contains an interesting combination of valuable knowledge and condensed facts.



Markaudio-Sota Viotti

A Speaker Design Statement



Formally introduced to the US market in the first quarter of 2017, Markaudio-Sota says we can hear the difference in their designs featuring custom all-range symmetric driver technology. Stuart Yaniger reviews one of the company's new loudspeakers, the Viotti One, the new flagship two-way, stand-mount loudspeaker.

Photo 1:The Markaudio-Sota Viotti One speakers are available in multiple-color finely finished cabinets.

By
Stuart Yaniger
(United States)

MarkAudio is well known in the DIY community as a provider of high-quality loudspeaker drivers, specializing in single-driver and wide-band speakers. The drivers have been the basis for dozens of published projects, and they're easy to use, generally not requiring complex multi-way and crossover designs, and using relatively simple cabinets. They are designed by British-based Mark Fenlon and his team and sourced from factories in China. MarkAudio provides well-documented plans for speaker systems made from their drivers, and thus have become popular with hobbyists and small shops as an easy and relatively economical way to achieve good-sounding speaker systems.

Viotti One
Markaudio-Sota
SotaAcoustics Ltd.
Unit 609-6106/F
Bio-Informatics Centre
No. 2 Science Park West Ave.
Hong Kong Science Park, Hong Kong
info@markaudio-sota.com
Price: $2,995/pair

Noting their success in the DIY market, MarkAudio and SOTA Acoustics teamed up to design and market a line of high-end loudspeaker systems under the Markaudio-Sota brand. The range includes single driver and multiway systems in a variety of form factors, from mini-monitors to floor-standing towers. The speaker under review here, the Viotti One, falls in the middle—it's a two-way stand-mount monitor, featuring a 50 mm cone tweeter and a 110 mm cone woofer loaded in a ported cabinet of approximately 30 ltr. Both cones appear to be made of a gold-colored alloy. The stands are included and are finished in a luxurious-looking black lacquer. The price is a little less than $3,000 for a stereo pair.

The drivers are described as having similar cone profiles, but scaled according to their size. Markaudio-Sota describes this as a "symmetrical soundfield," the idea being that both units having a similar cone profile will be advantageous for maintaining a smooth polar pattern through the crossover region. Cone tweeters are a bit of a rarity. I think the last speaker I had that used them was the ancient Fulton 80s I used when I was in grad school during the late 1970s.

Crossover is stated to be at 2.4 kHz, with asymmetrical second-order electric and acoustic slopes. This is a rather remarkable claim, since having electrical and acoustic slopes being equal requires the drivers to have bandwidths extending several octaves beyond the crossover point. Interestingly, the crossover designer, Scott Lindgren,



Photo 2: The acoustically transparent grille of the Viottis is held in place magnetically. (Cynthia Wenslow photo)



Photo 3: I removed the grille before taking the near-field woofer and port measurement, where the cloth would be a hindrance and the waveguides would not have an effect anyway. (Cynthia Wenslow photo)

was given the desired crossover point and slopes as a design target, rather than the more usual practice of choosing a crossover point and slope to fit the drivers and the desired polar pattern.

Visually, the Viottis are, in a word, gorgeous (see **Photo 1**). The high gloss finish gives them the appearance of designer furniture, and the elegant curves bespeak some serious industrial design. Fit and finish are immaculate, and the speaker's feel and appearance are consistent with their price. The cabinets feel solid, and rapping on them furthers that impression—no hollowness or distinct "tune." One clever feature is the use of small magnets to hold the grill-cloth in place (see **Photo 2**). Since the molded plastic grill cloth assembly is intended to be part of the acoustic design (there are small waveguides molded around the driver and port apertures), all listening and measurement was done with them in place. The only exception was near-field woofer and port measurement, where the cloth would be a hindrance and the waveguides would not have an effect anyway (see **Photo 3**).

One small complaint—the binding posts seem to be of high quality, but they have a non-standard spacing. This prevents the usual 0.75" spaced dual banana plugs from being used (see **Photo 4**).

The manufacturer-provided spec sheet was a bit curious. The amplifier requirements gave different power level ranges depending on whether the amplifier was Class A, AB, or D. I haven't seen anything like that before, and I am frankly puzzled by this. One annoyance in the specs is the sensitivity rating, which is given as 88 dB SPL at 1 m and 1 W. Besides the lack of indication of signal (pink noise? white noise? sine wave?), sensitivity ratings are based on voltage, not power. A valid sensitivity rating would be given at a reference voltage (usually 2.83 V). Sound pressure level (SPL) at a specified power is efficiency, not sensitivity, and is very much signal dependent as well because of the variation of impedance with frequency. Markaudio-Sota is not alone in this, it's a common error, but I'd like to see this corrected.



Photo 4: The Viottis have high-quality binding posts, which are unfortunately not standard spaced. (Cynthia Wenslow photo)

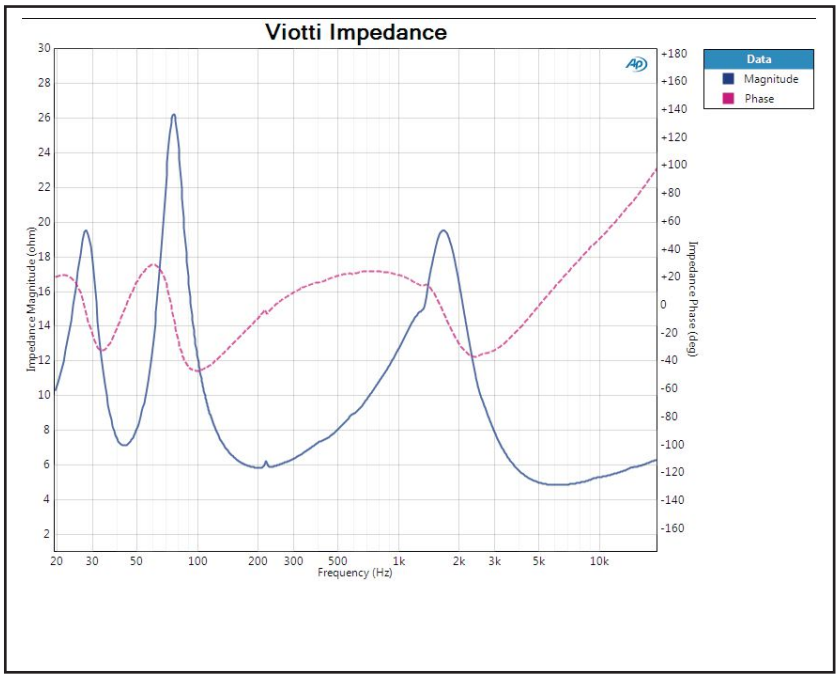


Figure 1: Impedance measurements. The double peaks in the bass are characteristic of a ported system—the saddle between them, at 43 Hz, indicates the box tuning frequency.

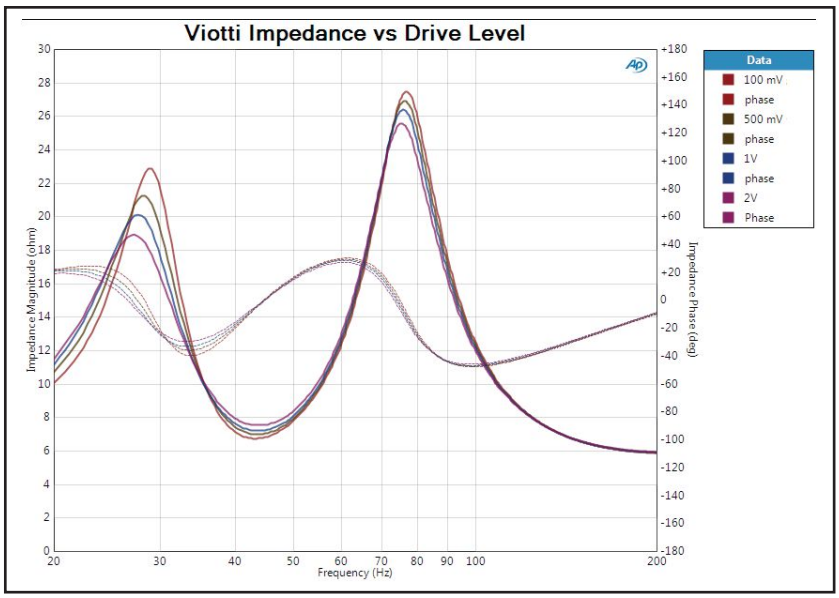


Figure 2: Low-frequency impedance measurements suggest a clean-sounding bass region.

About the Author

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

How Does It Sound?

Markaudio-Sota recommends 100 hours of break-in of the Viottis before doing any serious listening. I'm curious what in the mechanical system would take that long to settle into equilibrium, but I was a dutiful reviewer, so placed them face to face, and ran mono pink noise into each of them, with the channels out of phase. This greatly reduces the irritation factor from the continual noise! I let this run at 2.83 VRMS for about a week, then started the music appreciation phase.

I initially set the Viottis up in my living room. They seem to be rather critical of placement, with both spectral balance and imaging changing significantly as the speakers are moved. After experimenting a bit with room placement to get a good balance between bass smoothness and imaging, I ended up with the speakers about 60 cm (2′) from the back wall and about 110 cm (3.5′) from the side walls, with no toe-in. This placed me about 20° off axis, which yielded the smoothest treble and a solid center image.

One thing I noted was that my listening height was a bit too high (or the speakers were a bit too low) for optimum. I had to scrunch down in my seat to get the sound to balance well. So I ended up putting about 8 cm (3″) of wood between the stands and the floor to raise the speakers up a bit. I should note that at 180 cm (6′), I'm not particularly tall, and my couch is quite typical in size—it would be nice if the stands could raise the speakers just a bit higher.

I was surprised by the speakers' strong point, the bass. I'm not usually a fan of bass reflex (ported) speakers, but the tuning resulted in a tight and well-defined bass sound, verging on lean. There was never a hint of boominess, and the word that kept coming to mind was "articulate." Now, the lowest bass is clearly missing, and indeed, it would be unfair to expect high SPL at 20 Hz from a 11 cm woofer, but from about 50 Hz on up, there is little more to be desired. The kick drum and string bass never blurred together, and dreadnaught acoustic guitars had a natural richness without exaggeration. If I hadn't seen the port, I would have guess that this was a well-damped sealed box or an aperiodic, using a larger woofer. Realistically, this won't be satisfying for rock enthusiasts or fans of pipe organs, but they probably aren't looking at many speakers with an 11 cm woofer!

Since, for me at least, the lowest octave adds a better sense of realism to the overall sound of acoustic music, it occurred to me that I might want to try adding a good subwoofer, and I asked Mark Fenlon about it. He was less than enthusiastic about the idea, and was especially unhappy with the idea

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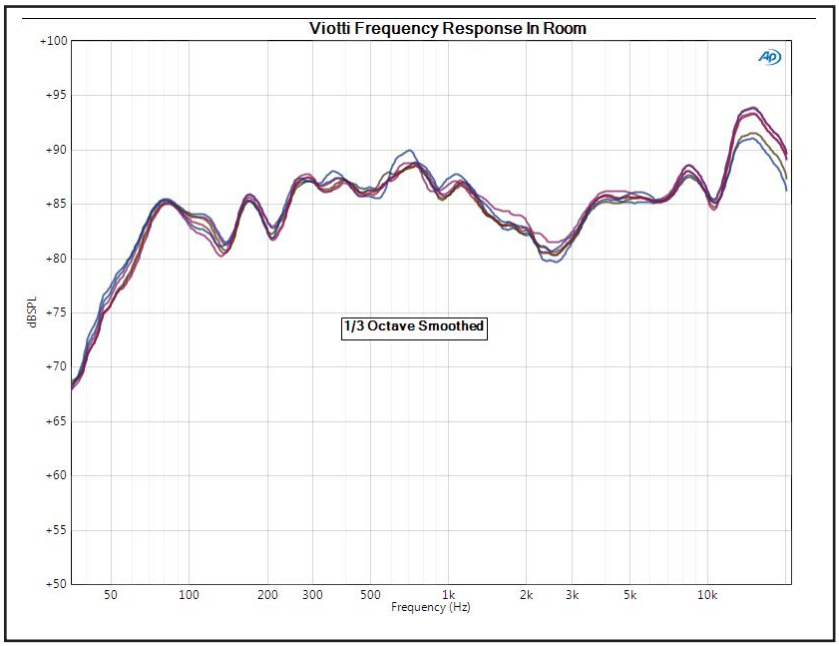


Figure 3: This is the 1/3 octave on-axis in-room response at 1 m microphone-to-speaker distance, with each curve representing an average of a cluster of five positions, spaced about 7 to 10 cm apart.

of using a woofer with a high moving mass for reasons of bandwidth (which would seem to defy the Fourier theorem). Fenlon says, "We often use the term 'AGs' (Atmosphere Generators). Typical heavy coned, large mass, high power AGs make noise, not music. Their cones are so mass-dense and the profiling prioritizes structural stability for heavy load conditions that there's little (if any) physical potential for typical AGs to emit the micro-resonant wave pattern that allows listeners to distinguish fine harmonic structures generated by various musical instruments."

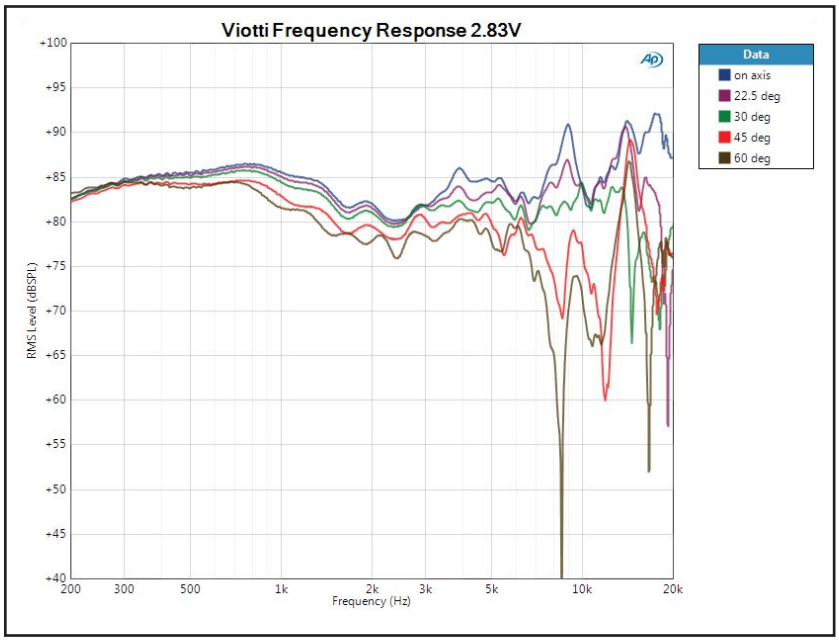


Figure 4: Here are the unsmoothed frequency response measurements at varying angles.

When boiled down and the Fourier theorem taken into account, Fenlon's main objection is that it would be too easy to set the sub level too high relative to the main speakers. This is true, but easily corrected with the subwoofer volume control. So at the end of the review period (all of my other observations in this review are for the Viottis as stand-alone units), I wheeled out my subwoofers (a pair of 10 ft^3 sealed enclosures with JBL 2245H drivers, equalized to be flat to 20 Hz) and paired them with the Viottis. With the levels and slopes set properly (second-order high-pass on the Viottis, sixth-order low-pass on the subwoofers, 50 Hz corner frequency), the subwoofers fleshed out the sound and added solidity and a better sense of the space of the rooms and halls where the recordings were done. No bass articulation was hurt or injured in the performance of this experiment.

Imaging and soundstage were very dependent on the distance between the speakers and my listening position. If I moved back too far, the sound was easier to localize as coming from the two speaker positions. When I moved in closer, the speakers sonically disappeared and I was immersed in a well-defined soundstage. If the Viottis are sited in a large room, it's worthwhile setting up the listening position accordingly.

Overall, the Viottis had what I describe as a "high-end audio sound." This is partially complimentary, partially not. On the good side, the distortion was low and it was easy to hear fine details without blurring. One of my favorite tests is a recording I made of Lee Barber playing "Tin Foil Satellite," a deeply sad and moving song about the change and loss as life progresses. At the end, there was a pause as the swept-away audience took in the intense emotion of the performance before bursting into applause. In the midst of that brief silence, you can hear a soft sob from one of our friends—if the speakers can resolve that subtlety. The Viottis did.

The downside of "high-end audio sound" is a coloration in the midrange, which has been nicknamed the "BBC dip." It's a broad and shallow recess in the midrange response that often occurs at about the crossover point chosen for the Viottis (2.4 kHz). Once you hear its effect using a speaker with a flat frequency response and an equalizer, it's easy to remember. Some claim that it adds realism to the sound, but I am not one of them. Here, this sonic tailoring was a deliberate design choice rationalized by the claim that it can prettify less-than-perfect recordings. I used the parametric equalizer function in the ADI-2 Pro DAC to reduce the dip. This helped, but was not a totally satisfactory answer.

If you like the sound of most high-end multiway box speakers or fondly remember the glory days of British monitors, this will not bother you as much as it bothered me. On the upside, the Viottis made recordings of my Ovation guitar sound more like an expensive wood-bodied guitar. But on the downside, the midrange coloration somewhat suppressed the great projection and forwardness of Peter Mulvey's 1957 vintage Martin on one of my home recordings of him. My personal preference is for midrange neutrality, enabling the user to choose the equalization as recordings demand.

I was also bothered a bit by the treble. On axis, the speakers sounded a bit fierce, but that's a deliberate design feature. Off axis at my listening position, the treble was better balanced, but not at all what I'd call "smooth." It didn't strike me as harsh per se, but it wasn't realistic sounding. Moving the speakers around and elevating the stands a bit helped, but the top octaves never really gelled in the way that they do in my reference system. As a quick test, I ran mono pink noise through the speakers—in my reference system, the pink noise sounds like it's coming from a single fairly sharp point between the speakers. With the Viottis, the pink noise sounded more diffuse and the tweeters spoke with their own voice, separately from the woofers.

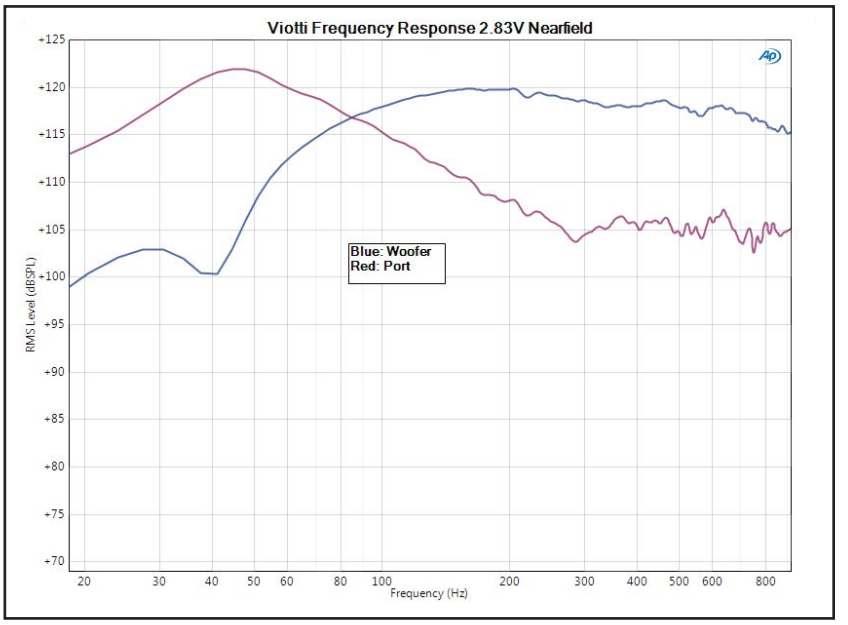


Figure 5: This is the near-field frequency response measurement for the woofer and the port.

How Does It Measure?

After spending about six weeks listening, I started the measurements. As usual, my measurement system for electrical and





Photo 5: The near-field measurement of the Viotti's woofer and port confirmed the listening impression of tight and well-defined bass. (Cynthia Wenslow photo)

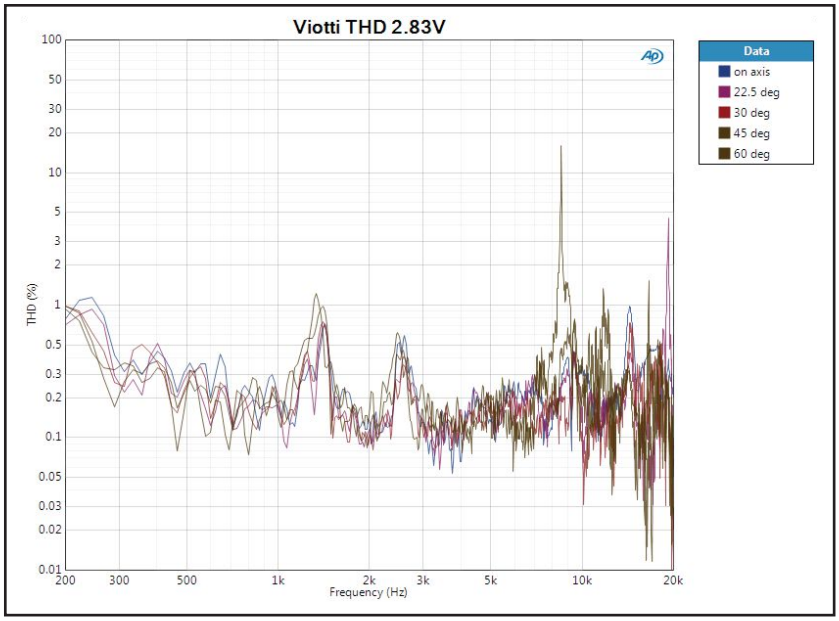


Figure 6: The distortion measurements at 2.83 V drive at a variety of angles.

Resources

D. Janszen, "Speaker Setup—Minimizing the Allison Effect," The Janszen Blog, http://david-janszen.squarespace.com/blog/speaker-setup-minimizing-the-allison-effect2015321

Markaudio-Sota, www.markaudio-sota.com

S. Yaniger, SYclotron Audio, www.syclotron.com.

acoustic response comprised an Audio Precision APx515 analyzer and an APx1701 Transducer Interface. For acoustic measurement, I used the new PCB Piezotronics 376A33 (0.5") and 376A31 (0.25") condenser microphones. These micorphones feature very low distortion at high SPL, high stability, and accurate calibration. At times, I also used a CLIO Pocket measurement system, just to check replicability of a few unusual-looking measurements.

First stop, impedance. The results are shown in **Figure 1**. The double peaks in the bass are characteristic of a ported system; the saddle between them, at 43 Hz, indicates the box tuning frequency. The higher amplitude of the double peaks is the one at higher frequency, which indicates that the box tuning frequency is below that of the driver's free-air resonance. In ported speaker design, that translates to a higher bass rolloff frequency, but a more gradual rolloff, which correlates with the speakers' very tight and well-defined bass.

There's a little blip at about 220 Hz, which suggests a mechanical or acoustical resonance of the box or driver and a similar one at about 1.4 kHz. The impedance minimum is about 5 Ω at 5 to 6 kHz, but the phase angle there is close to zero. Impedance stays at 6 Ω or above just about everywhere else, and the phase angle is within ± 40° below 10 kHz. This is a relatively easy load for a competent amplifier to drive, and indeed the speakers sounded relatively consistent with any amp I paired it with, within power limitations.

Recently, I've been looking at how woofer impedance (especially resonance points) shift with drive level, and have found some correlation between distortion and shifts in resonance point as drive voltage is changed. **Figure 2** shows how the Viotti's low-frequency impedance varies with drive. Relative to other systems I've measured, the change is very moderate, suggesting a clean-sounding bass region, which correlates with my listening impressions.

Figure 3 shows the 1/3 octave on-axis in-room response at 1 m microphone-to-speaker distance, with each curve representing an average of a cluster of five positions, spaced about 7 to 10 cm apart. The clusters were spaced about 15 cm apart from one another in a rough circle around the midpoint between tweeter and woofer. The midrange reticence I talked about earlier shows as a broad dip from 1 to 4 kHz. The bass wiggles are a function of the position of the woofer with respect to the floor and walls (the so-called Allison effect) rather than being intrinsic to the speaker. You can also see the designed-in treble elevation in the 10 to 20 kHz octave, as well as something of a roller coaster in the treble response above 7 kHz.

Quasi-anechoic frequency response at 1 m was taken on the same axis, but with a gated impulse response to remove room reflections. The unsmoothed frequency responses at varying angles are shown in **Figure 4**. Below about 7 kHz, the response rolls off smoothly with increasing angle, and the treble is most in balance at about 20° to 30° off axis, which I had found to be my preferred listening position. The midrange dip is also evident and fairly consistent with angle, suggesting that it's likely part of the crossover design rather than a diffraction phenomenon.

By contrast, the region above 7 kHz is a hot mess, with enormous swings in response. To an extent, some of the peaks fill in the dips as angles change, but this will only be perceived by a listener in a highly reverberant field (that is, farther away from the speaker in a live room). The dramatic changes with angle point to diffraction being the culprit, which may be the result of using a large diameter tweeter with a cone. I understand that having cone-and-cone for the woofer and tweeter is part of the company's design philosophy, but at some point, philosophy might have to take a back seat to physics if accurate performance is the goal.

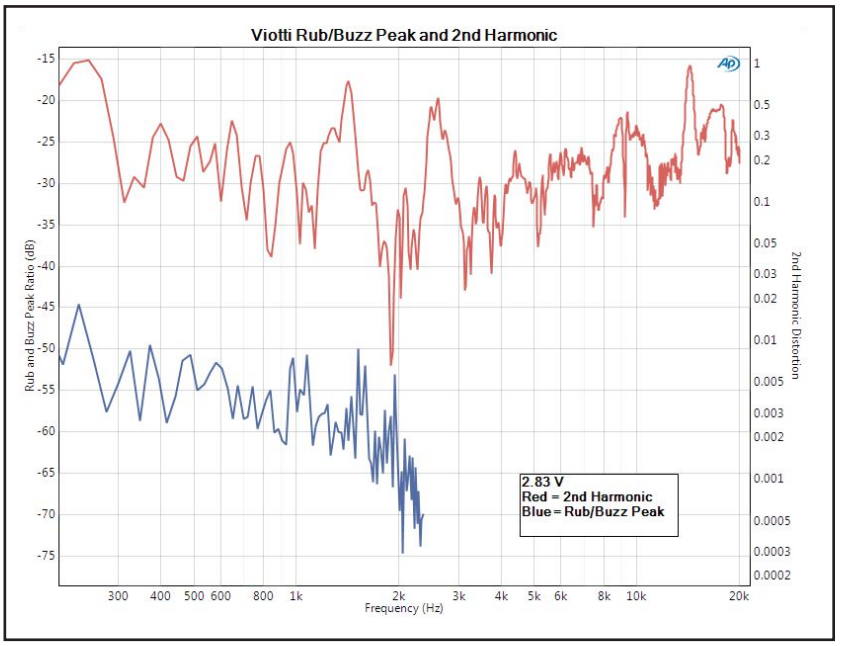


Figure 7: I measured the distortion via Audio Precision's "Rub and Buzz" algorithm.

Figure 5 shows the near-field frequency response measurement for the woofer and the port. Because of the difference in area between the driver and the port, the port's output will be reduced by 6 dB relative to the driver when the responses are summed. Overall response is rolling off by 60 Hz, and because of the tuning, there's useful output to 40 Hz or so. Note

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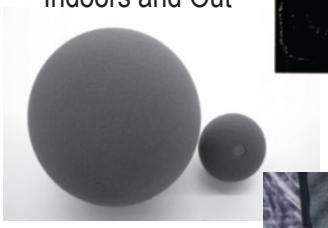
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the absence of pipe resonances in the port response. These data confirm the well-thought-out box/driver design (see **Photo 5**).

Figure 6 shows the distortion at 2.83 V drive at a variety of angles. The sharp spike at 8.5 kHz is likely a measurement artifact because of the sharp dip in frequency response at that angle. However, the two distortion humps in the midrange persisted with all measurements. Nonetheless, they are relatively minor, and spectral analysis showed that they were almost entirely second order, which is sonically benign at these levels. A more annoying form of distortion is measured via Audio Precision's "Rub and Buzz" algorithm, and those results are shown in **Figure 7**, along with the second harmonic levels. Overall, the distortion of the Viottis is commendably low. Having previously noted the very clean and well-defined sound, this came as no surprise to me.

Wrapping Up

The Viottis have a definite personality, which may or may not appeal to you. For your money, you get a superbly constructed and finished loudspeaker with low distortion and tight, though not deep, bass. The midrange is somewhat recessed and the treble is uneven, though not at all rough or harsh. The speakers are easy loads for most amplifiers. Since the manufacturer offers a 30-day money back guarantee, your risk is minimal. What they do well, they do very well, and if the strong points fit with your priorities, the price is very reasonable for the evident high-quality of construction and components.

Reference System and Music

I will confess to not being a typical audio enthusiast. I don't use much commercial equipment, don't have the typical "audiophile" albums, and I try to optimize factors with demonstrated audible significance in my design and setup. I probably am on the right-hand side of the bell curve in terms of frequency of hearing live, unamplified music, attending numerous concerts, and regularly hosting musicians in my home. If this kills my audiophile street cred, then so be it, pay no attention to my subjective observations and make what you will of the measurements.

Rooms: I listen in two different rooms. My living room has wooden floors and typical frame-drywall construction. Dimensions are 14′ × 8.5′ × 32′ (4.3 m × 2.6 m × 9.8 m). I would love to have acoustical treatments, but I would also prefer to have a happy marriage. Thus, I use rugs on the floor and padded furniture for absorption, and bookcases for diffusion. There's no discernable slap echo or major bass resonances, so at least I avoid the biggest potential problems.

My lab has concrete floor and walls, and a wood ceiling with joists running laterally—an unintentional diffusor. Dimensions are similar to the living room but the back opens into a larger space where my lab is. The front wall is treated with absorptive material (several layers of acoustic blankets spaced a few inches from the wall), the floor has several layers of rugs, and the side walls are a mix of diffusive and absorptive (bookcases, wine racks, and spot absorption).

Speakers: My reference speakers are based on the NHT 3.3, with the original drivers and cabinets, but triamped with an active crossover (the Didden-modified DCX2496, active version) and is described in an article on my website SYclotron Audio (www.syclotron.com). Amplifiers include tubes (The Red Light District, a 25 W per channel EL84-based design) and solid state (a venerable Adcom GFA555, a home-built 70 W per channel MOSFET design, and two 500 W mono Sunfire woofer amps).

Sources: For digital, I variously use the M-Audio Delta 192 D/A card, the RME ADI-2 Pro, and the DAC in the Behringer/Didden DCX2496 crossover. Source files are WAV and FLAC. For analog, I have a vintage VPI HW-19-II with an upgraded platter, bearing, and motor, a Linn Ittok LVII tonearm, and an Audio-Technica 150MLX or a modified Denon DL103R cartridge, driving Equal Opportunity or His Master's Noise phono stages, respectively, in balanced mode.

Miscellaneous Bits: Interconnects and cables are made from non-exotic but high-quality wire and connectors. XLR and MicroXLR are my connectors of choice.

There are no "magic" accessories—if I went into a hospital where the MRI machine had quantum purifiers, Schumann resonance absorbers, silver power cables, or whatever, I'd get the heck out and find a real hospital that wasn't run by quacks.

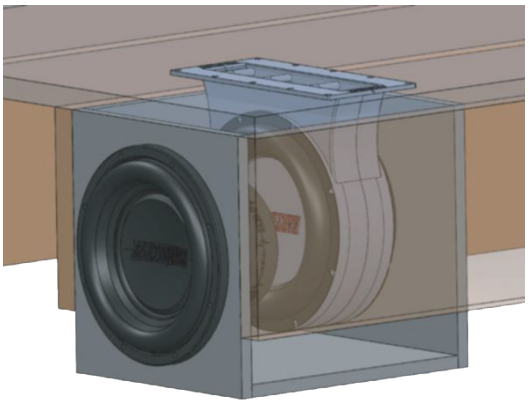
Reference Music: I use an assortment of acoustic music of various genres, with a little bit of rock mixed in just to see how that sounds. My own recordings of Lee Barber (www.leebarbermusic.com) and Southpaw Jones (www.southpawjones.com) were done with minimal miking, 24 bit /96 kHz digital, and no compression.

Reference commercial recordings include the eponymous album *Redbird*, which was recorded with a single-point stereo microphone in a living room, featuring voices and instruments I know well from live performance (including Peter Mulvey's in my living room and my library), "Art Blakey In Sweden," a wonderfully natural and unprocessed live recording on LP, *Engineer's Choice*, a sample CD of Delos classical recordings from John Eargle, Clifford Jordan's *Live At Ethell's* on Mapleshade CD, and "Old Fashioned," from John Statz, some straightforward folk featuring a voice and guitar that I've heard live and unamplified dozens of times. For fun, rock classics like *The Yes Album*, Frank Zappa's *Over-Nite Sensation*, and the Who's *Live at Leeds* exercise the high SPL end of speaker capability.

Disclaimer: I am a strong advocate of controlled subjective testing, that is, ears-only, no peeking. Unfortunately, I am not set up to do loudspeaker testing blind, so I'm certainly vulnerable to the same sort of biases and preconceptions that plague typical "subjective" reviewers.

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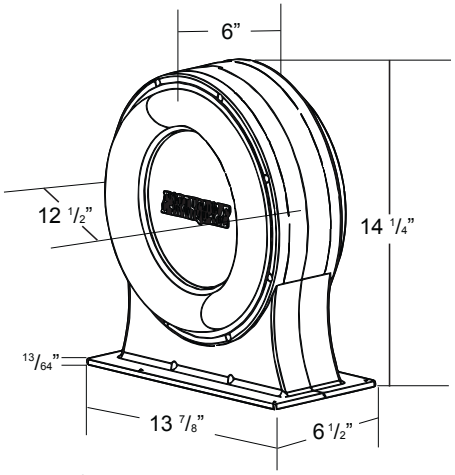
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R&D Stories

Virtually Tuning an Automotive Audio System

Experts at HARMAN are using physical experiments in conjunction with mathematical modeling and numerical simulation to improve the development process for the latest vehicle infotainment technology.

Photo 1: Here the HARMAN team experimented with loudspeaker positioning in the vehicle interior.

By
Valerio Marra
(United States)

Today's vehicles offer dazzling electronic entertainment possibilities—from smartphone connectivity to interactive displays and video screens. HARMAN is the market leader in these connected car setups, equipping more than 80% of the world's luxury cars with premium audio systems.

Each vehicle model requires a unique configuration, and HARMAN's team of acoustic and simulation specialists ensure that different components and car acoustics are accounted for in the design process. Details such as the ideal placement and orientation of speakers, speaker packaging, and driver enclosure geometry (e.g., car doors) all influence the sound quality.

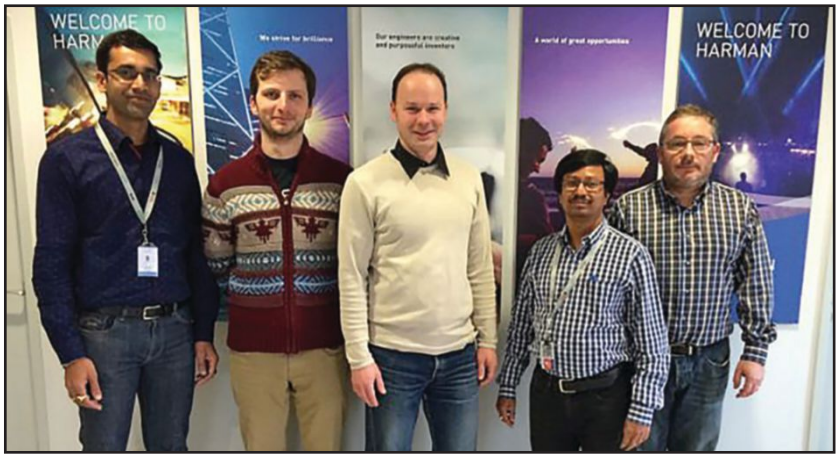


Photo 2: The HARMAN VPD team consists of Maruthi Srinivasarao Reddy, Michal Bogdański, Michael Strauss, Ninranjan Ambati, and François Malbos.

The team uses physical experiments in conjunction with numerical analysis to accelerate product development by virtually "tuning" the systems before ever creating a live prototype. This saves time on physical testing, and enables virtual tests to replace in situ listening, so that the team can design their products even before the final car designs are complete.

"We may become involved very early in the car development process, when a vehicle designer has not yet decided what is required from the audio system," explains Michael Strauss, Senior Manager of Virtual Product Development (VPD) and Tools at HARMAN. "Or we may only have basic details such as size and volume of the car cabin. Yet, frequently we need to present a concept within a few days, creating a tricky challenge to meet our clients' requirements and deliver high-quality systems."

Simulations and Experiments Create Customer Satisfaction

To provide customers with a response that is both quick and accurate, engineers at HARMAN turn

to mathematical modeling in COMSOL Multiphysics software. "We needed capabilities for mechanical, acoustic, and electrical simulations in one integrated environment, and we wanted a program that would free up the time and effort spent on creating and updating our own tools," says François Malbos, Principal Acoustics Engineer, at HARMAN.

"The multiphysics approach is one of the most important parts of the virtual product development process," says Michał Bogdański, Simulation Engineer and Leader of the Project at HARMAN. "We can explore how the acoustic behavior of a loudspeaker relates to any part of a vehicle structure—for example the stiffness of a door—and then provide door design guidelines to our customer."

In one case, they both measured and simulated the sound pressure levels (SPL) generated by a loudspeaker in the cabin of a Mercedes-Benz ML car (see **Photo 1**) in order to validate their numerical models and later use them to optimize acoustic equipment. "Car cabin simulations are among the most challenging to run because they cover many different areas of physics," explains Strauss. Fortunately, COMSOL software offers options to couple together the acoustic, mechanical, and electrical effects throughout the system.

To support companywide engineering efforts, Strauss' team (see **Photo 2**) established a library of validated models and known solutions that allows for performance predictions of a variety of loudspeaker configurations. "We are able to offer everything from a high level trend analysis to a detailed design examining the performance of a subsystem," he continues.

Analyzing Vehicle Loudspeaker Performance

In one study, engineers at HARMAN used COMSOL to create a simulation of a car cabin's sound system to optimize the speaker acoustics, specifically for low-frequency soundwaves. They then designed a series of tests to validate the model. Once validated, the model would enable the HARMAN team to deduce the best loudspeaker setup for a given car.

In validation tests, a loudspeaker was mounted on a rigid enclosure near the driver's seat of the car. Four sets of microphone arrays throughout the cabin served to measure the average SPL at each location (see **Figure 1**).

For frequencies below 1 kHz, the loudspeaker was represented as a rigid flat piston tied to a simplified lumped parameter model (LPM), taking into account the voltage at the voice coil terminals and the stiffness of the suspension and speaker membrane surface. The geometry was generated from a manual 3D scan (see **Figure 2**). Using a postprocessing algorithm implemented in MATLAB

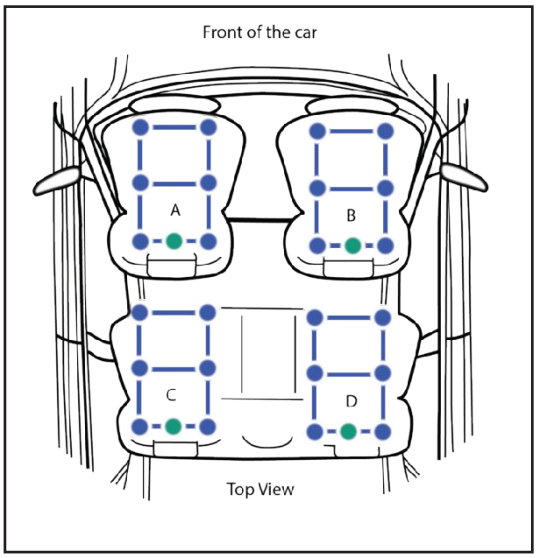


Figure 1: The top view shows the microphone arrays positioned in the four different locations.



Figure 2: HARMAN took a 3D scan of the car cabin.

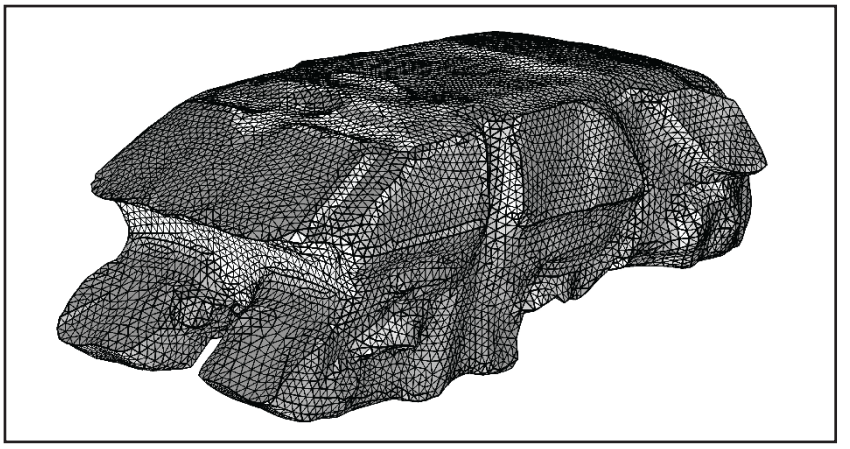


Figure 3: Using a postprocessing algorithm, the HARMAN team created a surface mesh of the car cabin.

About the Author

Valerio Marra is the marketing director at COMSOL in Burlington, MA. His previous roles include working in applications, technical support, and sales. Valerio received his PhD in fluid machines and energy systems engineering.

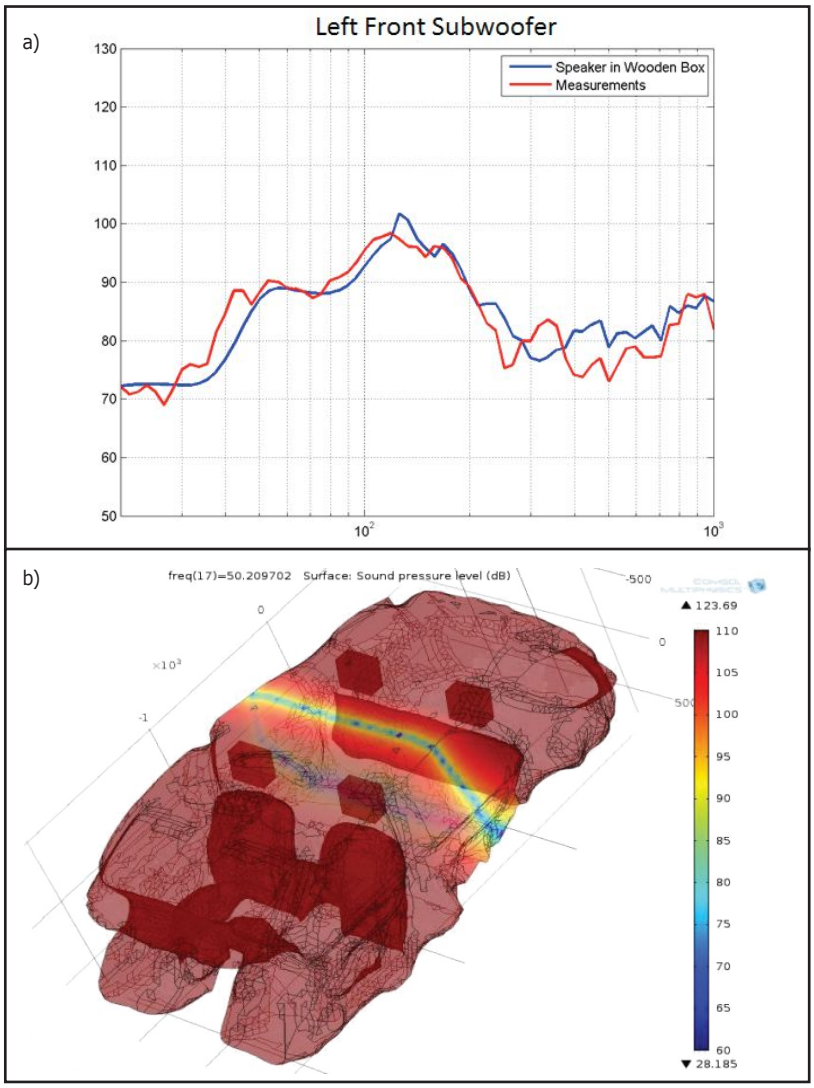


Figure 4: Sound pressure levels are shown for one microphone array (a) and throughout the cabin (b).

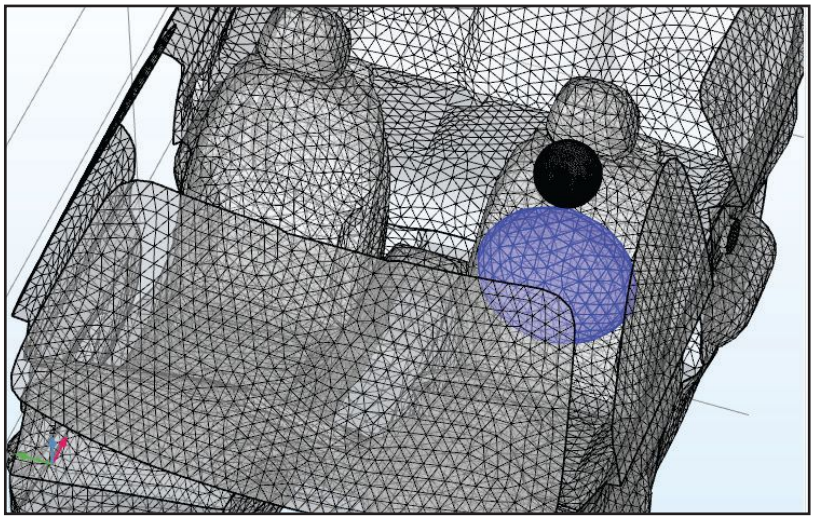


Figure 5: Mesh created using COMSOL helps predict binaural room impulse responses or how ears receive a sound.

software and an add-on product to COMSOL called LiveLink for MATLAB that creates a bidirectional link between the two programs, the team converted the point cloud created by the scan into a surface mesh of the car cabin (see **Figure 3**) and created an optimized mesh for studying acoustic pressure waves.

The simulation analyzed the interaction of the sound waves generated by a speaker with the different materials of the windshield, floor, seats, headrests, steering wheel, and other sections (e.g., the roof, the doors, and the instrument panels), each of which have different absorption properties.

Optimizing the Acoustic Model

In addition to accounting for many different materials, the team also defined speaker membrane motion and acceleration based on the volume of the enclosure using the Livelink for MATLAB and developed special MATLAB scripts to simplify the preprocessing and postprocessing activities.

"Everything is fully optimized and automatic so that we do not have to calculate the acceleration for each case; when one simulation finishes, the next launches," explains Michal Bogdanski. "This ensures that the whole process is easy and error-free; we simply let the scripts run."

The team also optimized the frequency-dependent absorption coefficients necessary to achieve a strong correlation between the measured and simulated sound pressures. The analysis then provided the SPL emanating from each microphone array (see **Figure 4**).

Objective and Subjective Evaluations in the Driver's Seat

Using their validated simulations, HARMAN is able to begin developing a sound system even as a vehicle is still being designed. The accurate prediction of the sound pressure field throughout the car cabin allows for optimization of audio system performance. Equalizers and psychoacoustic effects are also included in the tuning algorithm, allowing for design modifications without the need for a physical prototype.

Auralization, or the production of sound from virtually computed acoustics, is of interest in the pursuit of a top-notch sound system. Using a high end headphone, engineers at HARMAN have developed a playback system that allows for listening, evaluation, and comparison of audio systems comprising subwoofers, midranges, and tweeters. "All based on simulation results and signal processing," says Malbos.

HARMAN engineers include the effects of the

human head, torso, and ear canals on acoustics in predicting Binaural Room Impulse Responses (BRIR) or how ears receive a sound. To capture the full 3D sound, BRIR are computed at various head positions in the azimuth plane. The playback system uses a head position tracker to perfectly reproduce the sound experience as the listener would experience it (e.g., in the driver's seat).

Figure 5 depicts the mesh created with COMSOL software that was used in predicting the BRIR. **Figure 6** shows a comparison between predicted and simulated BRIR.

Auralization is not without its challenges. Auralization quality, a measure that is inherently subjective, must compare to real-world listening. As such, subjective measurements are made to ensure the quality of the listening experience.

At HARMAN, the ability to assess an audio system based purely on simulation has increased the quality of product and speed of product development. It has also improved customer responsiveness, and lowered the cost of design amendments, fostering a sense of design freedom among the engineers.

"The beauty of simulation is that a systems engineer can sit at a desk, put headphones on and begin to tune a system without the car," Says Strauss. "Using simulation, HARMAN engineers will be able to assess, optimize, predict and subjectively evaluate the performance of a proposed sound system, even though it does not actually exist yet."

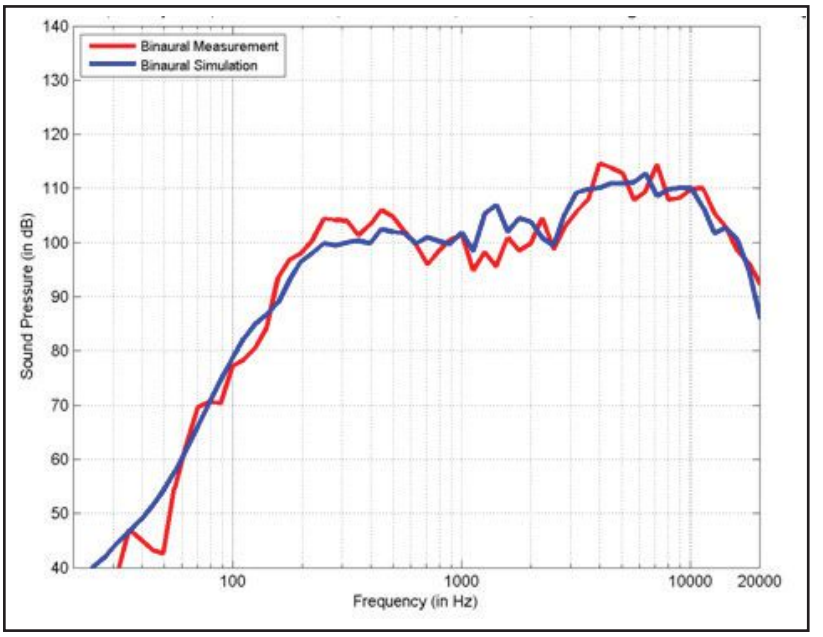


Figure 6: Here is a comparison of measured and simulated BRIR in the frequency domain.





The Denali Voltage Reference

Learn more about this small, wide range, low-power, and low-impedance design.



By
Christopher Paul
(United States)

Many audio power and preamplifier designs benefit from the use of voltage regulators. We can divide regulators into two distinctive classes: switching, and linear (although strictly speaking, there is a small group of others). Switchers are the more complex of the two and have their advantages and disadvantages. Peter Delos gave us a great discussion of switchers along with a design of one for a vacuum tube guitar amplifier ("A Switching Power Supply Design for Tube Amplifier Applications," *audioXpress,* January 2016).

In this article, though, I'll stick to linears and focus on how feedback networks affect their performances, particularly in the higher voltage applications that vacuum tube amplifiers demand. But before we jump into that, let's spend a little time talking about types of linear regulators and the circuitry that surrounds them.

Linear Voltage Regulator Configurations

All voltage regulators have references. References are usually "cleaner" and quieter, lower power versions of their regulator's output signal. In some cases, though, they can be the complete regulator itself. There are two basic classifications of regulators: series and shunt. **Figure 1a** shows generalizations of each. **Figure 1b** presents some feedback-free examples, and **Figure 1c** gives us some in which such feedback does exist. In the latter, negative feedback through the Z, R network enhances regulator voltage stability in the face of variations in load current and power supply voltage. Finally, **Figure 1d** offers designs where positive feedback through a similar network, implemented to set output voltage, actually derogates the supply rejection and load regulation of which the regulator ICs therein are capable. As we discuss these circuits further, we understand that R and Z are simply the impedances of networks that could contain resistors, capacitors, inductors, Zener diodes, or even active circuits.

Consider the Figure 1c designs, which apply the output voltage times R/(R + Z) to the regulator inverting input. Designs of this type reduce output disturbances, multiplying them by a factor of approximately (R + Z)/(A × R), where A is the loop gain provided by the relevant integrated circuit. Contrast that with the positive feedback in the Figure 1d design.

Its IC regulators act like voltage followers. A voltage divider's factor of Z/(Z + R) multiplies any disturbance at the regulator output and applies the product to its reference input. Rather reducing the disturbance, this feedback actually amplifies it by a factor of (R + Z)/R! Clearly, performance is best for both sets of circuits when Z is less than R. A closer look shows us that little is obtained by driving Z to smaller and smaller values below R; even driving it to zero yields only an improvement of a factor of two over the case where Z = R.

Keeping Z small is generally not a challenge for Figure 1c designs, but achieving output voltages much above a few tens of volts is challenging because of the operating range limits of the ICs themselves. Modifications of these designs are required to reach several hundred volts of output, and that tends to get complicated. The same high voltage problem presents for the Figure 1d designs, although additional Michael Maida-style regulator add-on components have been known to provide a solution (see Resources).

Unfortunately, the original LM317-based Maida design, and even some very modern updates thereof using the LT3080, employ networks where Z is a lot greater than R over at least some of the audio range. One way to address that issue is to make Z a string of Zener diodes. However, you can dissipate a few watts biasing these at levels to make Z adequately small, and they are typically quite noisy. A partial remediation comes from bypassing the Zener diodes with a healthy electrolytic capacitor, but that takes up a good deal of space. Finding a good Z – R network for the Figure 1d designs is not trivial.

It's worth noting that the folks at Linear Technology designed the LT3080 with the Z – R challenge in mind—they simply eliminated the need for R! A tightly controlled current of 10 µA flows out of the LT3080 pin connected to impedance Z. So let's see, for an output voltage of 400 V, all we have to do is to make Z a 40 MΩ resistor! There are some obvious concerns about a circuit with a 40 MΩ resistor. Fortunately, the LT3080 can still be used with finite values of R allowing smaller values of Z. But that doesn't solve the Z – R problem.

So where does all this leave us? Well, both the Figure 1b and 1d circuits would benefit from a low noise, low power, small and easy-to-program R – Z network with the beneficial characteristics we've been discussing. And what if such a circuit could also be used at the lower and negative voltages needed to bias the grids of power amplifier output stages? Is it possible to develop a solution that meets these needs? I believe I can answer "yes" to that question.

An Alternative to Zener Diodes

Consider the Figure 1c shunt regulator. It's compelling because for some variants of the LM431V, with under 1 mA of current and 100% feedback applied, impedances around 1 Ω can be achieved over much of the audio band. Of course, with the DC portion of Z less than that of R, we are restricted to output voltages below 2.5 VDC. And even if 1 Ω audio band performance could be obtained at higher DC voltages the device is incapable of operating over

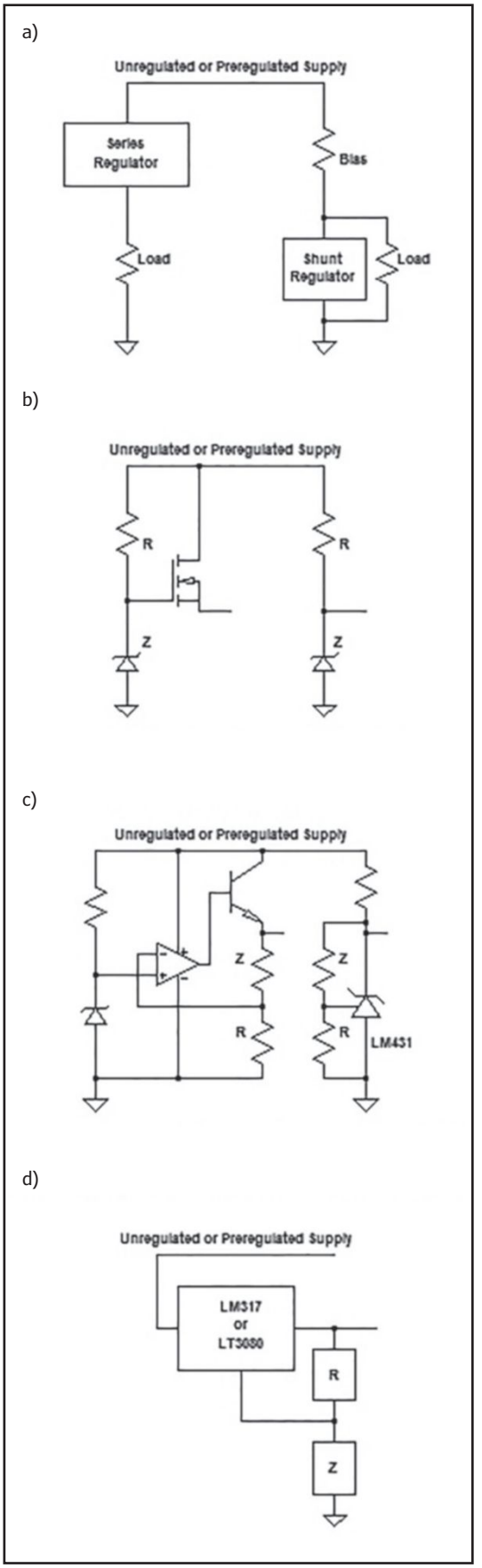


Figure 1: There are two basic classifications of regulators—series and shunt: generalized series and shunt regulators (a); series and shunt regulators lacking feedback (b); series and shunt regulators applying correctional negative feedback (c); and series regulator employing derogative positive feedback (d).

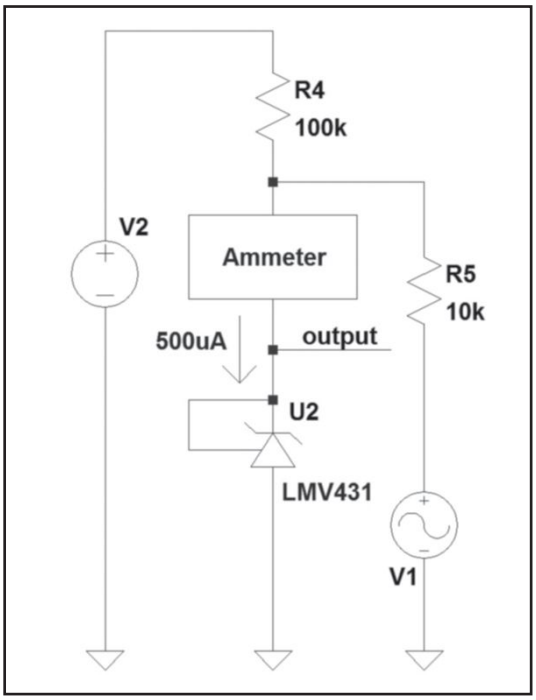


Figure 2: Testing an LMV431 reference with 100% feedback

30 VDC. So why the interest in it? Well, let's just say that these devices can get by with a little help from their friends.

Let's confirm the low value of the LM431V impedance when Z = 0 and R is infinite. Take a look at the circuit shown in **Figure 2**. V1 is a signal generator, and V2 is a variable DC supply. The supply is adjusted so that a current of 500 µADC passes through the LMV431. The output is connected to my t(rusty) Stanford Research SR-770 spectrum analyzer and an oscilloscope.

For the first set of tests, I set V1 to produce a 0 dBv sine wave at various frequencies. This revealed an impedance of less than 1 Ω from 3 to 1,000 Hz, rising to 5.7 Ω at 20 kHz. Next, I configured V1 to produce a 1 V_{PP} 1 kHz square wave. My oscilloscope showed some ringing. This was eliminated by a network of 150 Ω in series with 0.1 µF (not shown.) Finally, I disconnected the signal generator and measured the circuit's noise. From 4 Hz to 20 kHz, I saw about 250 nV/√Hz, in line with what I expected from the part's datasheet, and comparable to that typical of the LT3080. (The LM317 does not have a noise rating.) I was emboldened to proceed to the Denali prototype shown in **Figure 3**.

Why Denali? Well, Denali (formerly McKinley) is the highest mountain in North America, and this reference sustains a pretty high voltage. But since I'm sure that higher voltage ones exist, it would have been hubristic of me to dub it "Everest." Hence, "Denali."

In Figure 3, V1 and V2 are as before. M1 is a small signal MOSFET with a standoff voltage of 500 VDC. C1 was added to bypass R2 and R3 at audio frequencies, providing the LMV431 with a unity or lower Z/R ratio above 12 Hz. (I'll discuss the current source I1 later.) U1 forces a 1.24 V drop across and a 100 µA current through R1. I repeated my impedance test and was surprised to

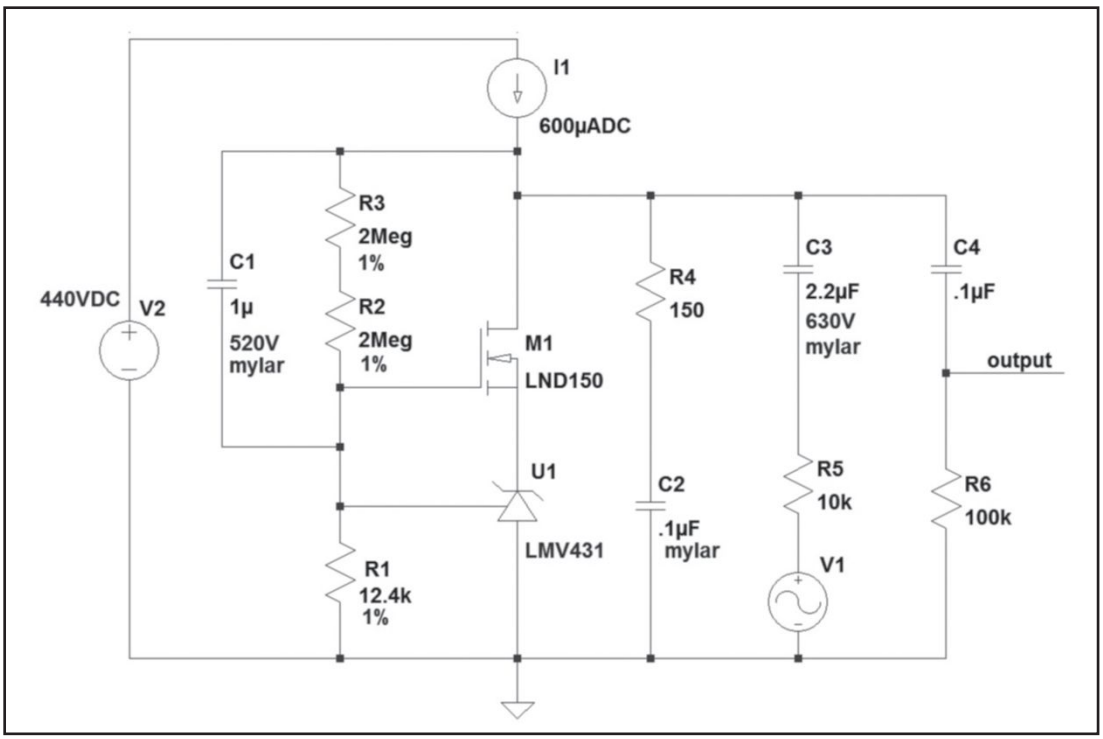


Figure 3: A prototype Denali Reference. Note that the output is a test output, not the reference system output.

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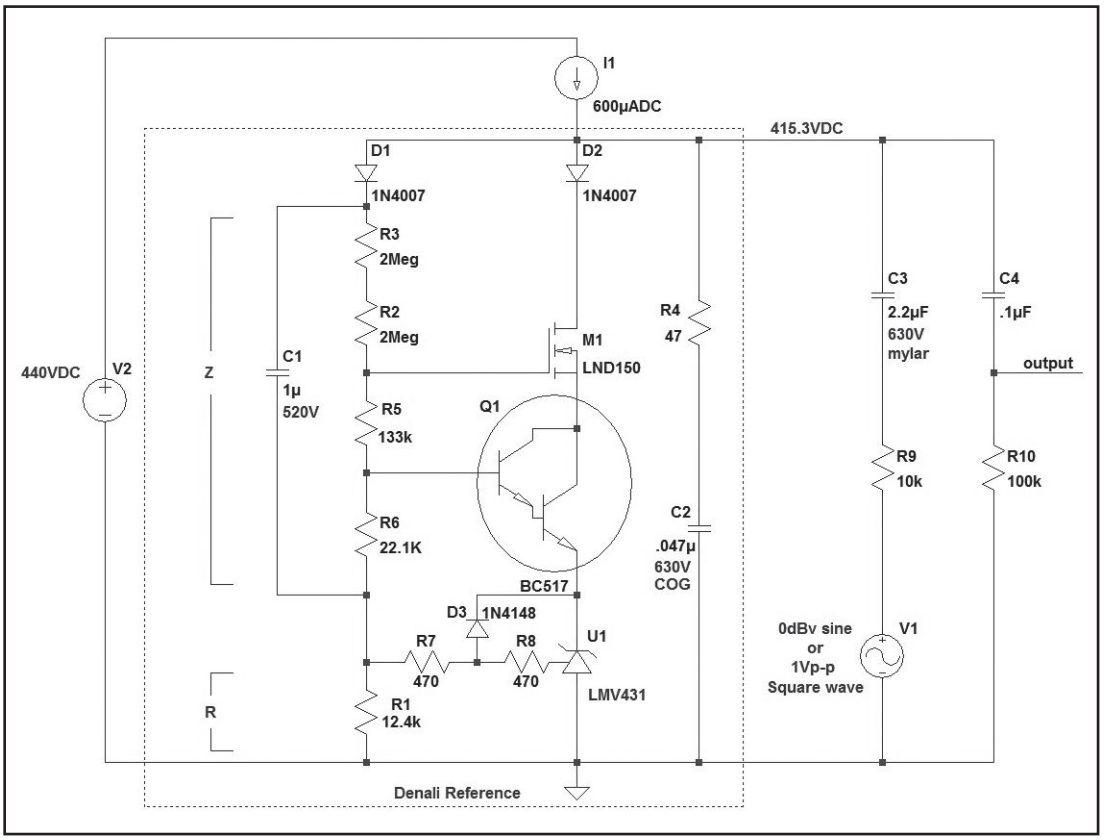


Figure 4: The completed Denali Reference. Note that the output is a test output, not the reference system output.

discover that the lowest audio band impedance I could measure had risen to 1.5 Ω, while the 20 kHz impedance had risen to 32 Ω. What went wrong?

Looking at the LND150 datasheet, I noticed that the transconductance of the part was typically 2 mS at 1 mA. This would be a little less at my 500 µA operating point, but close enough. The reciprocal of that number, 500 Ω, is the impedance looking into that MOSFET's source pin. Whatever voltage (V) that the LMV431 drove into that pin, the LND150 could only pass a current of v/500 Ω. Was there something that could be done to increase that current?

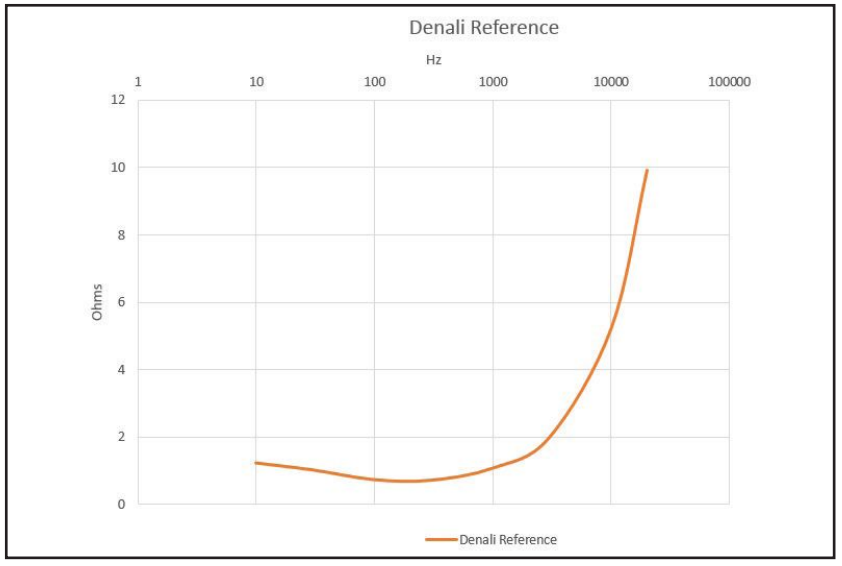


Figure 5: Audio frequency impedances of the Denali Reference

Well, the transconductance of an NPN transistor at room temperature is about I_{DC}/26 mV. I've biased the LMV431 and M1 at 500 µA. This would result in an NPN emitter resistance of 0.026/0.0005 = 52 Ω. However, that resistance would effectively be increased by the impedance seen by the transistor base divided by the transistor beta.

To try this, I connected the transistor base to the top of a 22.1 kΩ resistor added between R1 and R2, dropping 2.21 V, so that even if I needed the double base-emitter drop of a Darlington, the "top" of the LMV431 would be more positive than its 1.24 V feedback pin. That meant that the NPN base would see an audio frequency impedance of about 12.4 kΩ in parallel with the 1 µF of C1 plus 22.1 kΩ. But now I needed a beta of about 22.1 kΩ/52 Ω = 425 or higher to not derogate the 52 Ω resistance!

I decided to go for broke and use a Darlington. The BC517 is rated at a minimum beta of 30,000 at a current of 20 mA. Fortunately, its typical curves show that there is no loss in beta at any temperature down to a current of 1 mA, where the curves stopped. This part was worth a try.

The introduction of the BC517 brought me to the circuit shown in **Figure 4**. Note the addition of R5 and R6. I have already discussed R6; there is a 13.3 V drop across R5, affording the BC517 a comfortable V_{CB}. I also added D1, D2, D3, C2, R4, R7, and R8. C2 and R4 were arrived at through experimentation to tame a small amount of ringing that occurred in response to a V1 square wave test signal. A ceramic capacitor is the smallest type suited to C2. Do not use anything other than a COG or NPO ceramic. Others beginning with the letters X, Y, and Z lose significant capacitance when biased with a DC voltage. D1 and D2 protect the Reference as it discharges under normal conditions when power is removed, and even when a dead short is placed across it. D1 prevents negative voltages from appearing at the Ref (input) pin of U1, which could easily damage U1. D2 affords simple and cheap protection of the M1, Q1, and U1 in the case of a sudden short of the Reference. R7, R8, and D3 provide some protection against damaging levels of current being drawn by the U1 Ref pin in the unlikely case of a positive voltage step appearing cross the Reference.

None of these components degrades regulator performance in any way with the exception of D1. Its 26 mV/100 µA = 260 Ω resistance causes a negligible increase in Z with respect to the R of R1, but it does add to the Reference's overall temperature coefficient. At high voltages such as shown in the Figure 4, the effect is negligible (417 V falls by about 88 mV when the temperature rises 40°C.) At one tenth that output voltage while under use as a negative grid bias reference, it might begin to be objectionable, but I'll discuss how this can be ameliorated.

So How Does It Test?

The components within the dotted line box shown in Figure 4 form the Denali Reference, while those remaining power or test it. The Reference impedance over the audio frequency range is shown in **Figure 5**. It is always less than 10 Ω. Above 20 kHz as it rises further, it works with R4 and C2 to keep the impedance less than 100 Ω. Be careful about adding capacitance across the entire Reference—ringing or instability could result. As the frequency falls below 20 Hz, the Reference impedance begins to rise. At DC, capacitor C1 has no effect, and Z rises to about 4.13 MΩ, pitifully larger than R1's 12.4 kΩ. The small amount of feedback at DC causes the Reference impedance to rise to 249 Ω. This was measured by connecting a voltmeter and a switched 100 µA load across the Regulator. Noise performance remained below 300 nV/√Hz over the audio range. Tests over a temperature range from 25°C to 65°C revealed a Reference output voltage change of less than 1%.

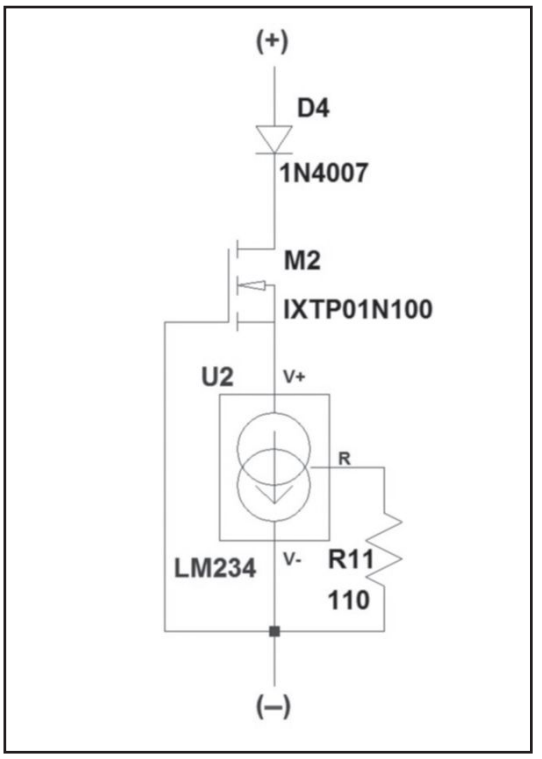


Figure 6: The Denali Current Source

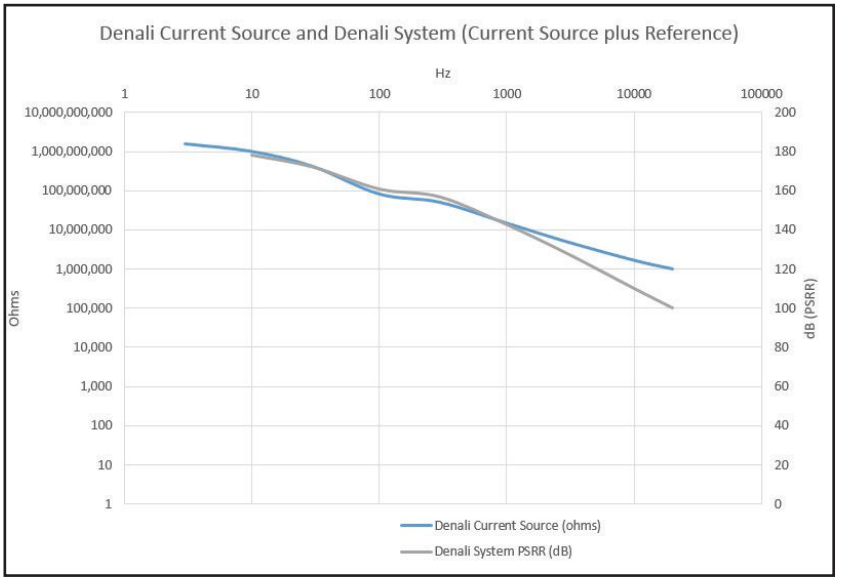


Figure 7: Impedance of the Denali Current Source and the PSRR of the Denali System (the current source supplying the Reference)

About the Author

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

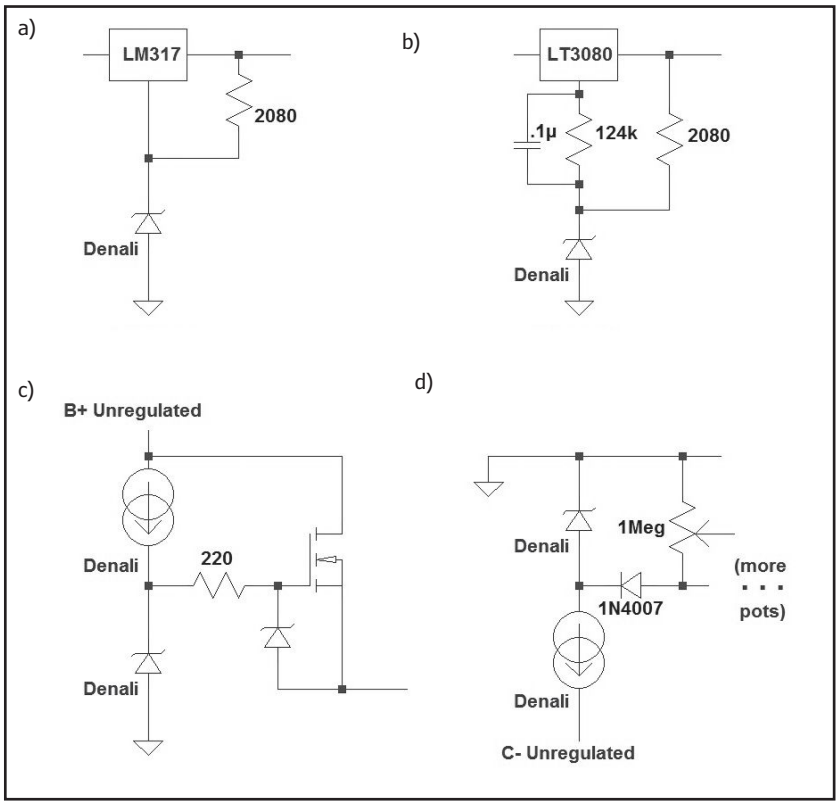


Figure 8: Some applications of the Denali Reference and the Denali System

Programming the Denali Reference Output Voltage and Start-Up Time

Both of these characteristics are controlled by selecting values of C1, R2, and R3. U1 acts to hold a voltage of 1.24 V across R1, which responds by drawing a constant 100 µA. Since the U1 pin monitoring that voltage draws no more than 0.4 µA over temperature, and the base of Q1 and the gate of M1 draw even less, virtually all of this current flows through the parallel combination of C1 and R_{total}, where R_{total} is the series combination of R2, R3, R5 and R6. It also flows through D1, which has a drop at 25°C of 450 mV. So V_{out} is simply 100 µA × (R1 + R_{total}) + 450 mV. Use resistors with temperature coefficients of 50 ppm/°C. Select R2 and R3 to achieve the desired voltage. Ensure that they are rated to handle the voltage drops across them.

The circuit's time constant T is C1 × R_{total}. So 3 × T seconds after power is applied, the circuit will have reached 95% of its final voltage. Do not reduce C1 below 1 µF. This is because below 1/(2π × C1 × R1) Hz (13 Hz when C1 is 1 µF), the Reference's Z exceeds its R.

The Denali Current Source

It's time to pull the cover off the mysterious I1 shown in Figure 4 and reveal the Denali Current Source, which is shown in **Figure 6**. This circuit owes much of its impressive performance to the LM234, which, interpolating from graphs in its datasheet, exhibits an impedance at 600 µADC of approximately 10 MΩ or more below a few hundred Hertz, and this with as little as a volt across it! But wait! There's more! Add diode D4 for reverse voltage protection and cascode the LM234 with depletion mode MOSFET M2 to increase its impedance and its operating range to 1,000 V.

Figure 7 shows the Denali Source's impedance is about 1 MΩ at 20 kHz and rises at 30 Hz to about 420 MΩ! Continuing on down in frequency to 3 Hz, we measure 1.6 GΩ—more than a thousand million ohms! I will infer a DC resistance of the same in recognition of the difficulty in measuring one. Figure 7 also shows a calculation of the power supply rejection ratio (PSRR) that results from the Denali Reference being powered by the Denali Current Source. The results are displayed in decibels. (You didn't think I was going to try and actually measure signals with a 180 dB dynamic range, did you?)

So how about the DC PSRR? The ratio of 1.6 GΩ to 249 Ω is not as impressive as that at 30 Hz. It has fallen to a mere 136 dB. But if we reduce R2 and R3 to achieve an output voltage of 40 V, we increase the DC feedback by 20 dB, and that should win the DC PSRR an additional 20 dB.

But not everything about the Denali Current Source is butterflies and lollipops. True, it has a phenomenal operating voltage range. When I quickly switched between 440 V and 5 V across the circuit, my meter didn't budge from 609.7 µA. But leave it at a higher voltage for a while, and the current climbs noticeably. Why? M2 drops 600 µA × 440 V = 260 mW, and that's enough to heat it up and pass that heat to the LM234. And the LM234 has a non-zero temperature coefficient.

How bad is this effect? I raised the ambient temperature of the current source by 40°C and saw an increase of a little less than 100 µA once things stabilized. That means that at 400 V, the Reference's output voltage would rise by 100 µA × 249 Ω = 25 mV. At 40 V, the rise would be reduced to about 2.5 mV. I'm not losing any sleep over this.

Denali Applications

We now know enough about the Denali building blocks to apply them in practical applications (see **Figure 8**). All of the designs therein employ the Denali Reference, which is depicted as a simple Zener diode. Where the Denali current source is used, it is shown as a pair of overlapping circles with an arrow inside of them.

Figure 8a and **Figure 8b** show portions of regulators can supply up to several hundred

milliamps. They assume additional Maida regulator-style components that enable the ICs within to operate at high voltages. The LT3080's 10 µA set current creates a 1.24 V drop across the 124 kΩ resistor, which does the same for the associated 2.08 kΩ resistor. Both 2.08 kΩ resistors supply 600 µA to Denali References. Note that 2.08 kΩ is greater than the impedances of the References (R > Z) from DC to beyond the audio range, even for the high DC resistance of a 415 V Denali Reference.

Figure 8c is a simpler design which can supply just as much current. But without feedback around the MOSFET, the impedance and PSRR are not nearly as impressive. The current source compliance voltage is 1,000 V, so there is plenty of room for unregulated supplies greater than the 500 V limit of the Regulator. The Denali duo appears again in the negative voltage grid bias design of **Figure 8d**. Note the 1N4007. It does an excellent job of compensating the small temperature dependence of diode D1 in the Reference. Mount these two diodes close to one another so that they operate at the similar temperatures.

Wrapping it Up

The Denali System provides a number of high quality options for a reference for a B+ regulator as well as a complete negative grid bias regulator. If you want to build one for yourself, be my guest. If you want to employ it in a commercial design, that's fine too, but only if you refer to it in the accompanying documentation as being a Denali Reference System and/or Current Source. You see, I just want you to give my design a good reference!

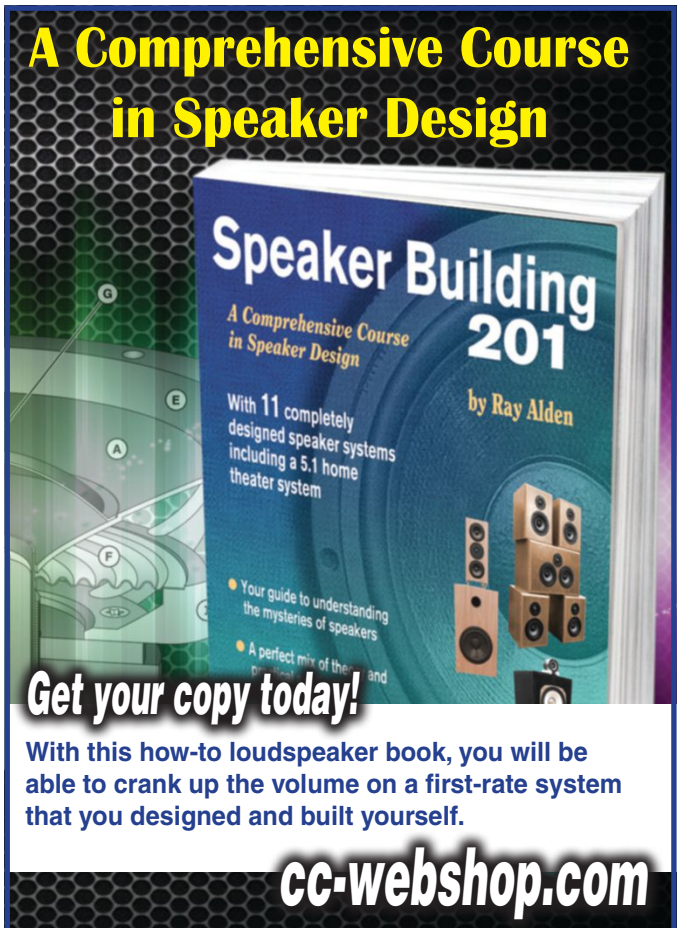
Project Files

To download additional content such as the Bill of Materials (BOM), LTspice schematics and reference tables, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Resources

P. Delos, "A Switching Power Supply Design for Tube Amplifier Applications," *audioXpress*, January 2016.

M. Maida, "High Voltage Adjustable Power Supplies," Linear Brief 47, National Semiconductor, March 1980, www.tsf-radio.org/forum/im/36310doc-national-lb-47.pdf.





Engineer Follows Dream, Opens Audio Design Company

An Interview with Mark Schneider of Linear Tube Audio



Mark Schneider founded Linear Tube Audio in 2015.

Mark Schneider founded Linear Tube Audio (LTA) in 2015. Coming from an extensive engineering career that included management, manufacturing, and product design—including being the architect behind the fastest document scanner in the world—Mark was forced into an early medical retirement. After a remarkable, multi-year recovery, he decided to commit himself to the passion that started a long time ago with a pair of three-panel Magnepans, a Thorens turntable with an SME arm, and some SAE amplifiers.

By
Shannon Becker
(United States)

SHANNON BECKER: Tell us a little about your background.

MARK SCHNEIDER: I had a friend who was an audio store salesman and I got my first high-end stereo at 17—Magnepan Tympanies, SAE amps, and a Thorens 124 with a Shure SME. I went to school for electronic engineering and eventually became director of engineering at a company where I developed document scanners for banks. I retired from my corporate career at 56 due to health.

SHANNON: How did you become interested in audio electronics?

MARK: My dad's CIA friend gave us a Halicrafters SW receiver, which sparked my interest, and my brothers and I would also make big speaker cabinets to use with amps we would find in old consoles.

SHANNON: Tell us about your first audio project.

MARK: My first serious audio project was several years ago when I took an Eico HF-81 amp schematic and made one using modern parts. I used that for two years until I received David Berning's prototype for the first of the LTA amplifiers. Once I heard the ZOTL amp, there was no going back.

SHANNON: Can you provide a bit of background about ZOTL technology?

MARK: David Berning had been making amplifiers for 20 years and was frustrated that he could not get the

sound he wanted using output transformers. At the time, he was designing switching power supplies and for the office holiday party he decided to make a lightweight amp using the same kind of resonant inductor technology, assuming it wouldn't sound great but would be good enough for the partygoers. After building it, however, he realized that it finally accomplished what he had been trying to do.

ZOTL (a name coined by Harvey Rosenberg) manages to get around listening to a transformer's inherent tone without requiring the system-specificity that other OTL designs often require. ZOTL gives a very low output impedance (less than 2 Ω) and easy matching with speakers of any impedance above that, it isn't damaged without a load, and it can give a frequency response that is flat from 10 Hz to 60 kHz. It also allows the tubes to be DC coupled, which makes possible an effective protection circuit Berning also designed that shuts off the power supply.

While the ZOTL amps have been bunched together with OTL designs because of the basic fact of eliminating traditional output transformers, it would be much better for people looking for an amp and more accurate to simply call the ZOTL amp a unique topology, or simply call it a tube amplifier, as it exhibits a flexibility and ease of use, when constructed well (the hard part), that will rival any amplifier using tubes.



The LTA MZ2 is shown in black and in silver. The MZ2 is the latest version of the ever-evolving successor to the Berning micro-ZOTL.

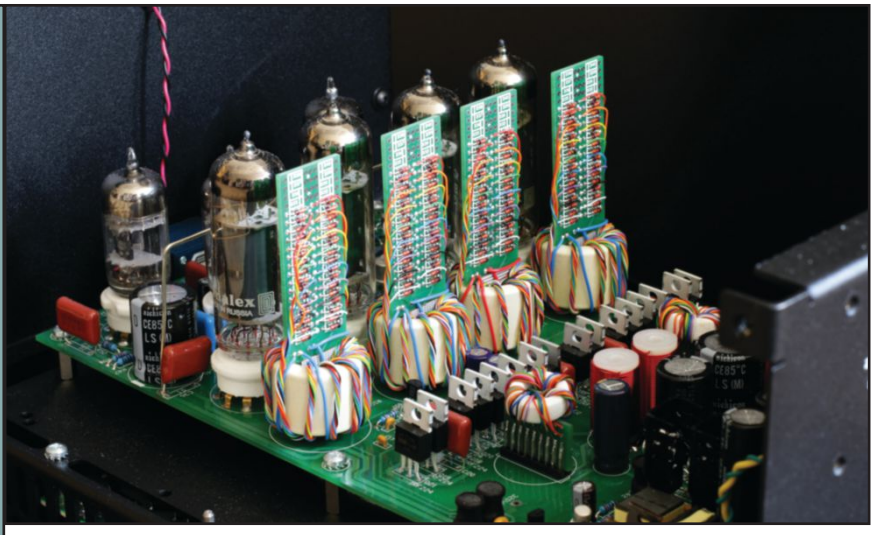
SHANNON: How did you first learn about it and what was it that attracted you to it?

MARK: My friend spent 20 years making his own electrostatic speakers and I used to help him with the construction. The speakers operated at 5,000 V and he had David make some custom ZOTL amps that had very high voltage output. They sounded like nothing I had ever heard.

SHANNON: When did you first start collaborating with David Berning and how did that come about?

MARK: My friend with the custom electrostatic





At left: The ZOTL10 and ZOTL40 Mk.II are shown. Both units utilize the same chassis, which was done to make them as cost-effective as possible. Linear Tube Audio does not recommend stacking your audio components. Above: An internal shot of the ZOTL10 clearly shows the impedance converters and other custom parts that make up David Berning's circuit.

speakers passed away and at his funeral reception, which was a large event, I was hanging out in the kitchen avoiding small talk. David was in the kitchen for the same reason, so we talked a lot. The conversation stayed with me. Later, I had a big medical procedure and was bedridden for months, so I learned PCB layout and designed the updated Eico HF-81. A few months later, when I was recovered and looking for a post-retirement endeavor, I thought it would be great to make high-value amps that sounded really good, so I took my updated HF-81 over to David's house to see what he thought. He showed me the ZOTL10 prototype and the letters asking him to build micro-ZOTLs again, and that's when we started to work together.

SHANNON: When and why did you start your company, Linear Tube Audio?

MARK: I spent a career as an engineer and always dreamed of making audio products. Since I had a strong manufacturing background, I wasn't afraid of making David's amplifiers (not knowing at that point about the two other companies that had given up).

I felt that David's technology had a real advantage over most other amplifier designs on the market, and David seemed to want to spend less time producing labor-intensive amplifiers and spend more time with other interests. I did not like the trend of the best audio equipment being crazily expensive, and David's amplifiers could compete with any amp at any price, but they were available to so few people. I was in a unique position of being able to bring a superior technology to the market with prices lower than Berning products had ever been offered. If we were successful, there was the possibility of offering some of the best-sounding audio at relatively affordable prices. So, in early 2015, I started the first batch of 50 microZOTL2s. In May 2015, I offered the first ones for sale to the people who had written letters to David asking him to make the micro-ZOTL again.



This is the audio board of the ZOTL40 Mk.II, with the KT77s standing taller than the impedance converters. The Berning-designed switching power supply, which is the same design as in his highest-end David Berning Co. amps, is located beneath the audio board.

SHANNON: Can you tell us a little bit about the team behind the products?

MARK: We have five full-time and three part-time employees. The core four-man manufacturing team have been with LTA since Nov. 2015. They're all musicians in their 20s and early 30s. I taught them to solder when they came in and they're now experienced, skilled technicians. After volunteering to sit at the booth during our first audio show, Capital Audio Fest in 2015, my son Will has gone from part-time to full-time and now helps me run the company.

SHANNON: Tell us about some of your products.

MARK: The MZ2 is the current iteration of the microZOTL2. Since the first batch, nearly every aspect has been improved in some way. It's really a remarkable little amp that David designed as his inexpensive showpiece for a new technology and it's been fun to see how we can build upon it to realize its full potential.

The ZOTL10 is our high-value entry point to ZOTL power amplification. EL84-based, it is a great partner to sensitive and single-driver speakers.

The ZOTL40 is our flagship and David's most recent step in refining the ZOTL circuit, this time around the EL34 (we use KT77s). This amp is designed to be the last amp one would desire to buy.

SHANNON: Do you have a favorite? If so, why is it your favorite?

MARK: The ZOTL40 is amazing to me. There is some unbelievable new speaker technology out there, but

I have always favored electrostatics. They shamelessly show the good and the bad. The ZOTL40 is fast, dynamic, super detailed like no other tube amplifier I've heard, and with my electrostatics fools me into thinking the performers are in my room.

SHANNON: Can you talk about the tubes a little bit? What is it about the tubes that you like?

MARK: The vacuum tube is still the most linear amplifier man has ever invented. When they are overdriven and clip, it is a much softer clip than the transistor. So many performers insist on tube electronics when playing and recording. I have had many solid state amps over the years, but once I was exposed to tube amplifiers I could never go back.

I love the classic audio tubes like the EL34 and EL84 that we built our first amps around, but after spending more time with David I've realized that taking advantage of the wider range of tubes that ZOTL allows (TV sweep tubes and others that don't get along with traditional circuits) one can get beautiful sounding new old-stock tubes that are inexpensive. You may not be able to tinker around with rolling name-brand, sought-after tubes, but you won't need to.

SHANNON: Linear Audio provides several pieces of the audio chain. What is the perfect combination of products for you musically?

MARK: The microZOTL2 as a preamp combined with the ZOTL40 is the most musical combination I have heard (except for David's much higher-end amplifiers). The holographic space of the performance, dynamics from soft to thundering, and overall realism create an experience that allow me for the first time to be completely immersed in the music, as if it were the original performance, rather than thinking that some aspect of my equipment could sound better.

SHANNON: What's next for Linear Tube Audio?

MARK: We have learned a lot in the last two years. Customer feedback has been extremely valuable, as has the advice we've received from other people in the industry. We've developed relationships with some great small companies that share our values in their own components and are looking forward to continuing to work with them.

We are always developing ways to improve our current products, we have a version of the microZOTL2 that will be a full-featured preamp coming near the end of the year, as well as our first integrated amplifier. There are also a few brand new amplifiers from David that we're hoping to see appear in 2018, as well as some interesting non-ZOTL.

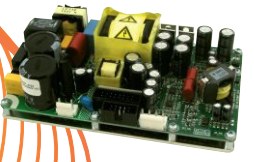
hypex NCxxxMP series

overview

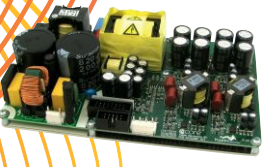
The NCxxxMP amplifier module incorporates a low power standby power supply (meets 2013 ERP Lot 6 0.5W requirements), a highly efficient switch mode power supply and a high- performance Class D amplifier in one compact and easily applicable power brick.



NC122MP



NC250MP



NC252MP



NC500MP

Highlights

- High efficiency
- Universal mains operation
- Flat, fully load-independent frequency response
- Low output impedance
- Very low, frequency independent THD
- Very low noise

Features

- One or two channel amplifier
- 5W standby SMPS
- Advanced over current protection
- External controlled operation
- Auto-switching (115/230V)
- Low weight
- Compact

Applications

- Monitor loudspeakers for recording and mastering studios
- Audiophile power amplifiers for professional and consumer use.
- Public address systems
- Active loudspeakers



NC502MP

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



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Build the Mojo Maestro



By
Ethan Winer
(United States)

For many years I've told people in audio and recording forums that distortion is a cheap commodity, and there's no reason to pay thousands of dollars for a "summing" mixer or other device with tubes and transformers just to get some subtle distortion. A controlled amount of soft-clipping is easy to create using two diodes and a few resistors. To prove the point, I finally decided to build such a device and the result is the Mojo Maestro.

The Mojo Maestro is a simple box that can add distortion ranging from very subtle—to add a little grit to a sterile sounding instrument track or to help "glue" a full mix together—through severe clipping to add "mojo" or "dirt" to any musical source that will benefit. The Mojo setting also switches in a capacitor to gently roll off the highest frequencies to counter the added buzziness. With gentler settings, the overall effect is similar to running music through a tube amplifier or an analog tape recorder but without the added tape hiss and flutter.

To devise this circuit, I used the free LTspice simulation software shown in **Figure 1**. Even though there are only half a dozen parts, this was an easier and faster way to choose appropriate resistor values than messing around with a physical breadboard and clipleads. LTspice lets you draw the schematic of an analog circuit and test it in software to see its frequency response and other attributes. Just as DAW software virtualizes an entire recording studio with effects plug-ins and even synthesizers, LTspice virtualizes a parts bin fully stocked with components, plus a workbench full of high-quality test gear.

In the past, I've used LTspice to verify the design of four-pole high-pass and low-pass filters, a multi-tone sawtooth wave generator-based on 555 timer chips, and a circuit that puts an audio isolation transformer in the feedback loop of an op-amp to minimize the transformer's distortion. Compared to those, this circuit was child's play! But the simulation was still very useful.

The oscilloscope view in LTspice lets me see how the waveform clips with different control settings and input voltages, and its Fast Fourier Transform (FFT) display shows exactly how much distortion is added at various frequencies. I settled on a 10 kΩ linear pot (replacing R2 shown in Figure 1) to vary the distortion from gentle to extreme, though I also added a switch with "Mojo/Glue" positions to control the maximum distortion amount.

Getting Your Mojo On

Figure 2 shows the schematic. **Photo 1** shows the labeled plastic project case and **Photo 2** shows the internal component assembly. Note that the 10 kΩ resistor in the LTspice schematic is present just to emulate the expected input impedance of the following device. The "real" schematic shown in Figure 2 details the parts I actually used. Since this is a passive device, its behavior depends on the input signal levels. Further, as you increase the distortion, the output level is reduced, so you may need to raise the volume later in the chain.

I initially built this device using 1N4148 silicon signal diodes that start to conduct around 0.6 V in their favored direction. That seemed like a good match for professional audio gear that operates at a nominal +4 dBu signal level (1.23 V). But I quickly realized that fairly large

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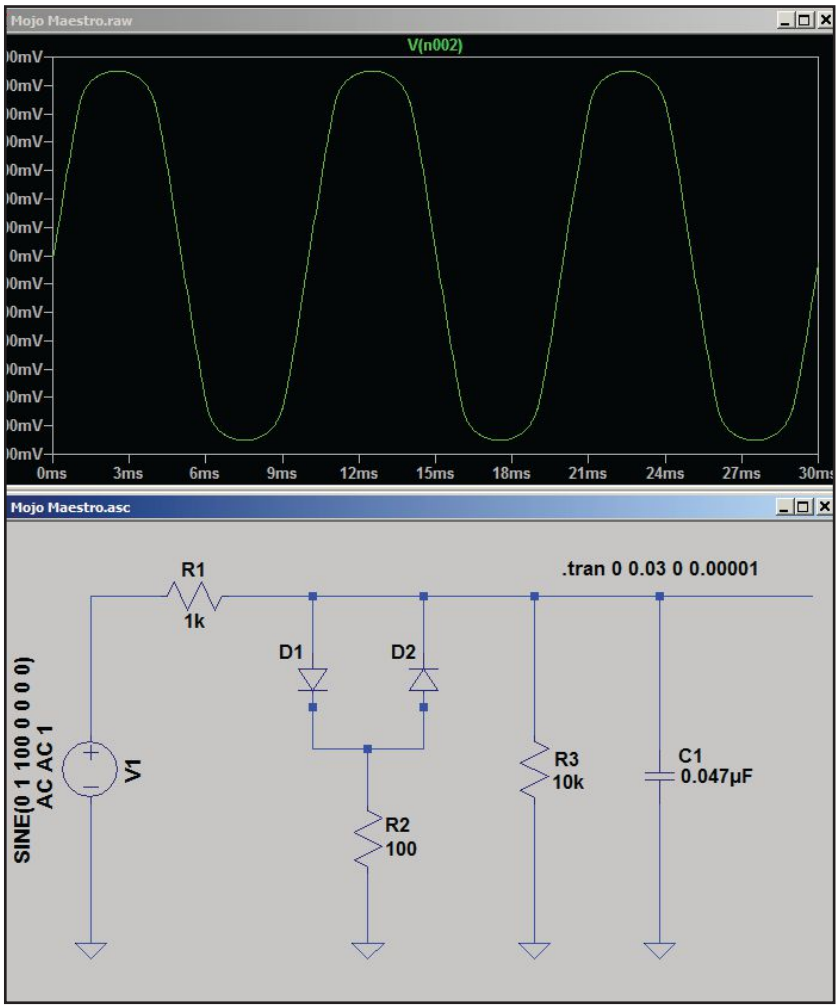


Figure 1: I used the free LTspice simulation software. LTspice combines a fully stocked parts bin with a signal generator, an oscilloscope, and a FFT analyzer.

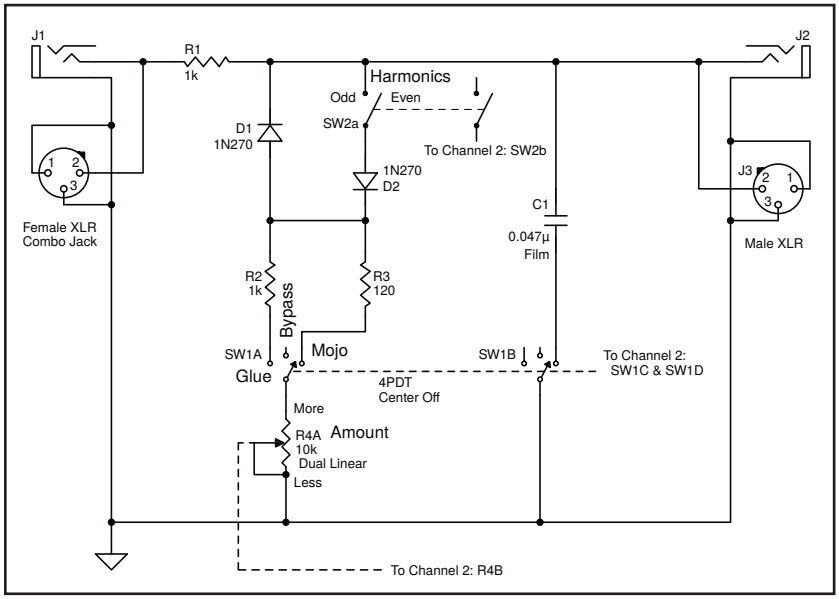


Figure 2: This is the schematic for one channel of the Mojo Maestro.

signals were required to get much of an audible effect, so I substituted 1N270 germanium diodes that conduct at around 0.2 V. This enables you to vary the effect amount from subtle to severe, using signal levels more typical of consumer devices while still accommodating professional gear operating at lower levels. Mostly as a joke, but partly to hide that this is just a few passive components, I labeled the front panel "State-of-the-art circuitry so efficient, it's self-powered by the input signal!" Which, of course, it is.

I used a 4PDT toggle switch with a center-off position to allow a true bypass mode. Even with the Amount knob at minimum, there's still some effect unless the input signal level is very low. The Odd/Even switch allows clipping only negative peaks to add both odd- and even-order harmonics, or it enables a second diode to clip both peaks which adds mainly odd harmonics. When set for "Glue," the Amount control varies the soft clipping effect over a 10-to-1 range as limited by the 1 kΩ resistor R2.

The Mojo setting instead uses a smaller (120 Ω) limiting resistor to allow more aggressive clipping. This setting also engages C1 to gently roll off the highest frequencies, though you could leave out that feature and use a DPDT switch for SW1, which costs less and is more common. Or, you could experiment with other capacitor values to change the amount and the frequency range of the high-frequency reduction.

It's worth mentioning that for static waveforms, symmetrical wave shapes contain only odd-numbered harmonics, and asymmetrical shapes contain both odd and even harmonics. So square waves and triangle waves contain only odd harmonics, where sawtooth and pulse waves contain both types. Also note that regardless of the Odd/Even Harmonics switch setting, intermodulation distortion (IMD) is always added along with harmonic distortion unless the input source is a single frequency. This is true for all electronic devices, including gear with tubes and transformers, and analog tape recorders. Also, every device that adds even-numbered harmonics also adds odd harmonics. Further, when playing music, any type of harmonic distortion always adds a similar amount of IMD.

I didn't bother to test Mojo Maestro with multiple frequencies in order to measure the intermodulation components because measurements don't really matter when the goal is adding color to a recording. But I did test the basic operation using a 100 Hz sine wave and included some screenshots.

Figure 3 shows some of the waves I recorded through the unit using a few different settings. **Figure 4** shows the FFT display with the generated odd vs. even components, when applying a fairly extreme amount of distortion. You can see that the red spikes are most prominent at 100 Hz and it's odd-number multiples

300 Hz, 500 Hz, 700 Hz, and so forth. The blue spikes show the spectrum when only one diode was in the circuit, and that trace has spikes at every 100 Hz multiple of the 100 Hz fundamental.

Design and Assembly

To accommodate different devices, I used Neutrik's clever NCJ6FI-S combination XLR and 0.25" connector for the left and right inputs, though separate connectors wired in parallel were needed for the outputs. I could have also added parallel RCA jacks for more flexibility, but I have so many adapters laying around that it wasn't worth the effort. Of course, you can use whatever connector type(s) you prefer.

The schematic shown in Figure 1 only depicts the left channel, but the right channel is identical. Even though this device is useful in professional recording studios, it's so simple it seemed silly to over-design it. So the XLR input and output pins 1 (ground) and 3 (signal negative) are tied together, then wired in parallel with the 0.25" jacks. The XLR pin numbers are obvious, and for the 0.25" phone jacks the terminal labels are Ring (unused), Tip (hot), and Sleeve (ground).

This circuit is too simple to require a PCB, but I still needed a way to securely mount the parts. I had



Photo 1: I used Microsoft Visio to design the connector layout, front panel holes, and label the plastic project case.

Audio Files

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

Source

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





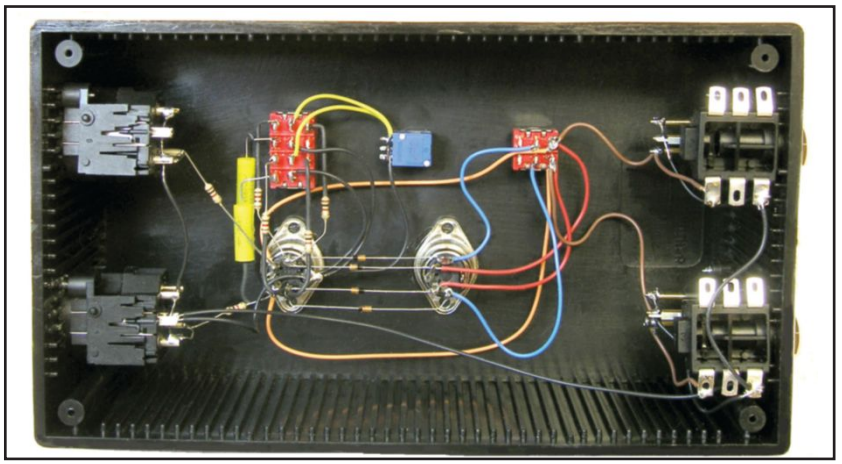


Photo 2: This circuit is too simple to require a PCB, so spare DIN connectors were used as stand-offs to mount the components.

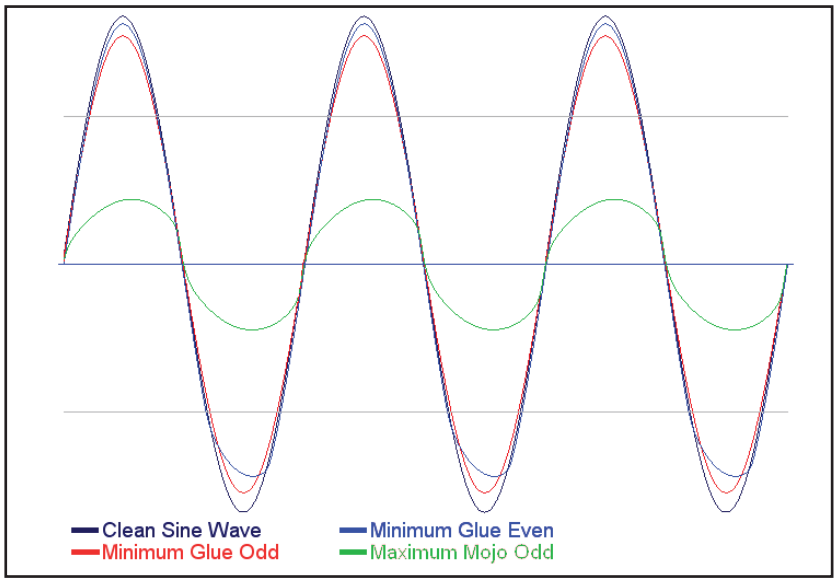


Figure 3: Mojo Maestro can clip signals gently or severely or any amount in between.

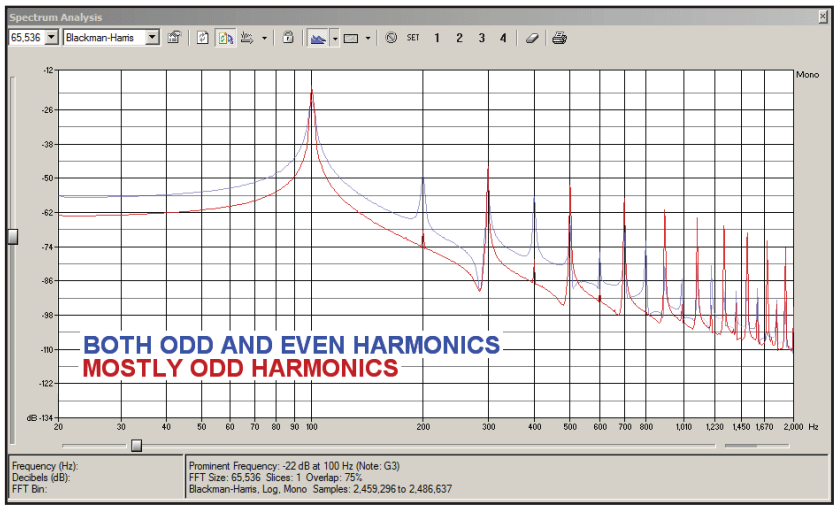


Figure 4: The FFT display shows the generated odd vs. even components, when applying a fairly extreme amount of distortion.

a bunch of extra MIDI-style DIN jacks from another project, so I used Epoxy to attach two of them inside the case to use as stand-offs to hold the parts, similar to terminal strips. Even though I built Mojo Maestro into an unshielded plastic box, it's meant to work with line level signals so I wasn't worried about picking up hum or noise. Just as important, the circuit impedance is relatively low, which also minimizes hum pickup.

Almost every modern audio device has a low output impedance and relatively high input impedance, so it wasn't necessary to use shielded wiring inside. To avoid wiring errors, I first made the layout drawing shown in **Figure 5**. Then, I soldered together all the parts and wires from that drawing rather than from the schematic.

I have a large collection of electronic parts that I've accumulated over time, and some are more than 40 years old. So I figured it would be prudent to at least check the diodes to be sure they're still okay. I'm glad I did because one of them was marked backward. I checked it three times comparing it to four other diodes because in all my years I've never seen that!

I bought new switches and connectors for this project, so I didn't have to worry about tarnish or corrosion on either the solder lugs or the internal contacts. Though I scraped off a bit of oxidization onto the diode leads to be sure they'd take solder well.

Speaking of switches, when wiring toggle switches you'll note that the terminals are opposite the handle position. Fortunately, if you wire it backward, you can simply rotate the switch 180° rather than have to unsolder then re-solder all the wires again. Likewise, always look carefully at a connector's pin numbers because, when soldering at the rear, they're reversed from the front view that's usually shown in a datasheet!

The last audio device I built was much more sophisticated than this one, and was a big investment in both time and parts cost, so I didn't mind spending $200 for a professionally punched and labeled front panel. I couldn't justify that sort of expense for this simple project. So, I decided to print my own control panel onto card stock and glue that to the plastic project box. I used Microsoft Visio to design the connector layout, front panel holes, and labeling. Then, I printed it onto a piece of legal size paper. Next, I wrapped the paper around the box, and spot-glued the corners with rubber cement to hold it in place while drilling the holes. A small "x" printed in the middle of each switch and connector hole made it easy to find the exact center when drilling.

After all the holes were drilled and de-burred, I printed the panel once more, this time onto heavy card stock. After using an X-ACTO knife to make all the holes in the card stock, I glued it to the case using a thin coat of contact cement. Finally, I mounted the

connectors and other parts. Card stock isn't as durable as real engraving or silk screening, but it looks a lot nicer than Dymo plastic labels! You could use colored card stock if you prefer, or spray it with a coat of lacquer after printing for protection against wear and smudges.

Listen for Yourself

To hear the results of the Mojo Maestro, go to the Supplementary Material section of the audioXpress website. In the zip file, you will find six audio examples, made using different settings of either Glue or Mojo, Odd or Even, and the Amount knob position.

Modifications

Besides trying different capacitor values to tame some of the brightness when adding more severe distortion as mentioned, you might want to try adding a pair of real audio transformers because that colors audio differently than clipping. The Triad TY-250P has reasonable fidelity at normal line levels, and at less than $6, it's very affordable. This transformer claims a response that's flat within 1 dB from 20 to 20 kHz, and it can handle up to 20 mW, which is adequate for line level devices even at the higher levels professional audio devices often have to handle.

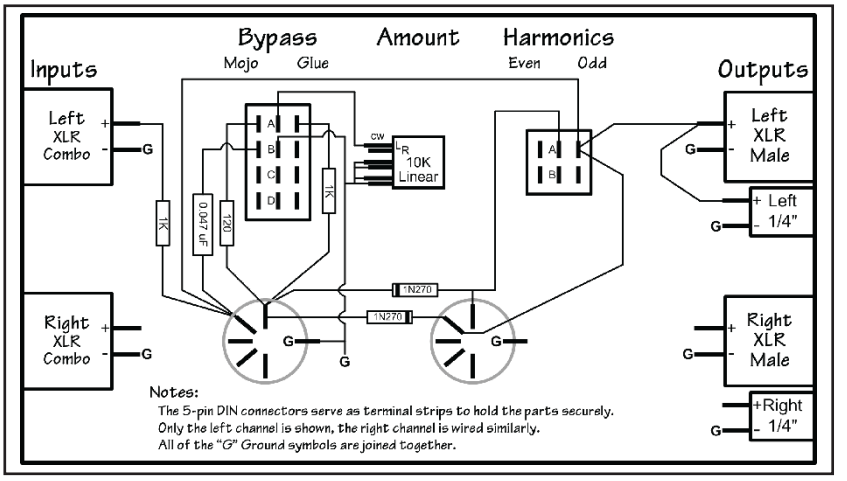


Figure 5: To avoid wiring errors, I first made this layout drawing.

If you try this, I suggest you connect the transformer at the circuit's input to control how much signal is applied to the transformer before clipping kicks in. Otherwise, the diodes would limit the level and restrict how hard you can drive the transformer into saturation.

About the Author

Ethan Winer has been an audio engineer and professional musician for more than 45 years. He's currently a principal at RealTraps where he designs acoustic treatment products for recording studios and listening rooms. Ethan's "Cello Rondo" music video has received nearly two million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press is available at amazon.com and his own website, ethanwiner.com.

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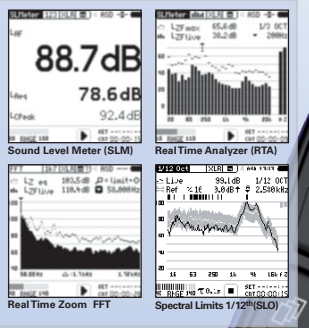
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Modern JFETs in Audio Circuits

The article describes the design and the performance of a high-quality DC-coupled all-JFET amplifier. The circuit was inspired by John Curl and Erno Borbely designs and was built with readily available JFETs from Linear Integrated Systems. The input stage is the complementary JFET differential pair, the second stage is symmetrical differential amplifier with two pairs of opposite polarity JFETs, and the output stage is a complementary source follower. The amplifier can drive loads up to 150 Ω. So, the all-JFET amplifier can be used for various audio applications.

By
Dimitri Danyuk
(United States)

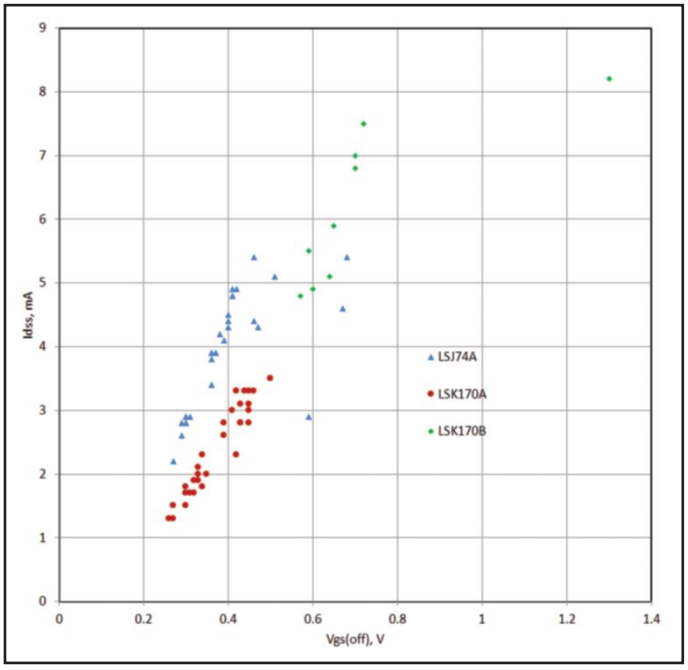


Figure 1: The drain saturation current I_{DSS} is shown as a function of pinch-off voltage V_{GS}(off) for single JFETs—LSK170 and LSJ74.

Toshiba Semiconductors manufactured many junction field-effect transistors (JFETs) for audio applications: the 2SJ74/2SK170, the 2SJ103/2SK246, and the 2SJ104/2SK364 to name a few. Once production of these through-hole JFETs stopped, they became expensive. These original devices are difficult to source. Another problem is a considerable amount of counterfeit devices on the market. Now, some compatible devices are sourced by Linear Integrated Systems (LIS) and it is time to resurrect those fine designs and build even better circuits.

The following parts suit best for audio applications: the 1 nV/√Hz low noise, high transconductance N-channel LSK170 and the complementary P-channel LSJ74; the dual monolithic 1 nV/√Hz low noise, high transconductance N-channel LSK389; the low noise, low capacitance N-channel LSK189; the dual monolithic low noise, low capacitance N-channel LSK489; the low noise, low capacitance P-channel LSJ289; and the dual complementary P-channel LSJ689. In the future, we may expect a dual 1 nV/√Hz low noise, high transconductance P-channel part from the manufacturer. Thanks to the availability of complementary JFETs from LIS, it is easy to design DC-coupled audio circuits. Parts, manufactured by LIS, can be purchased at Trendsetter Electronics (www.trendsetter.com).

Special attention is necessary when trying to accommodate newer devices in older circuits. Important parameters are maximum

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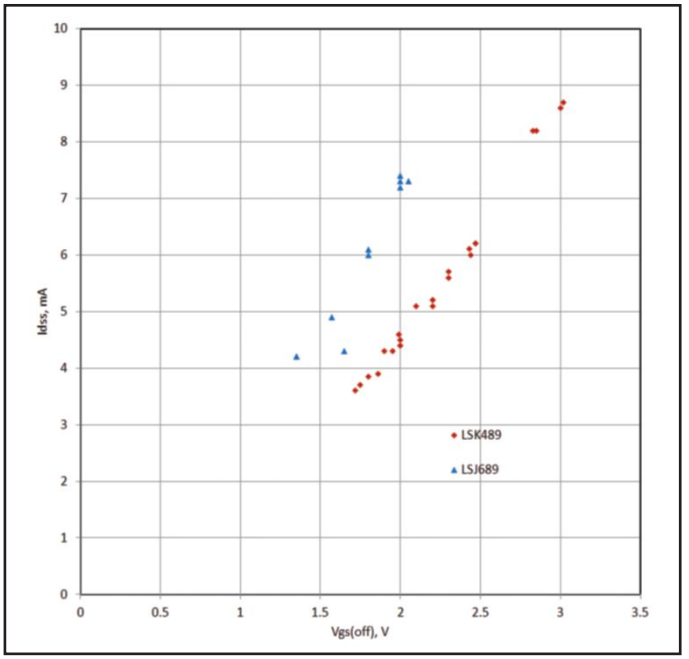


Figure 2: The drain saturation current I_{DSS} is shown as a function of pinch-off voltage V_{GS}(off) for dual JFETs—LSK489 and LSJ689.

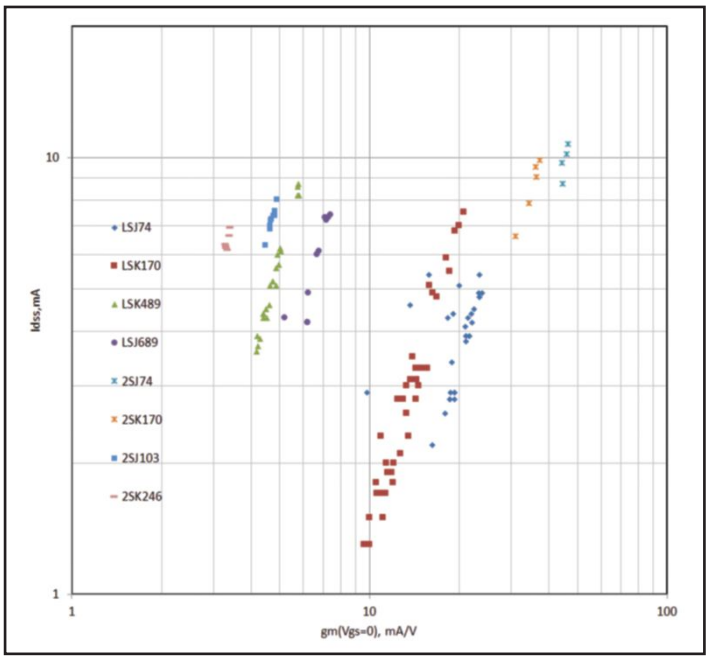


Figure 3: The drain saturation current I_{DSS} is shown as a function of transconductance at V_{GS}=0 for selected LIS and Toshiba JFETs.

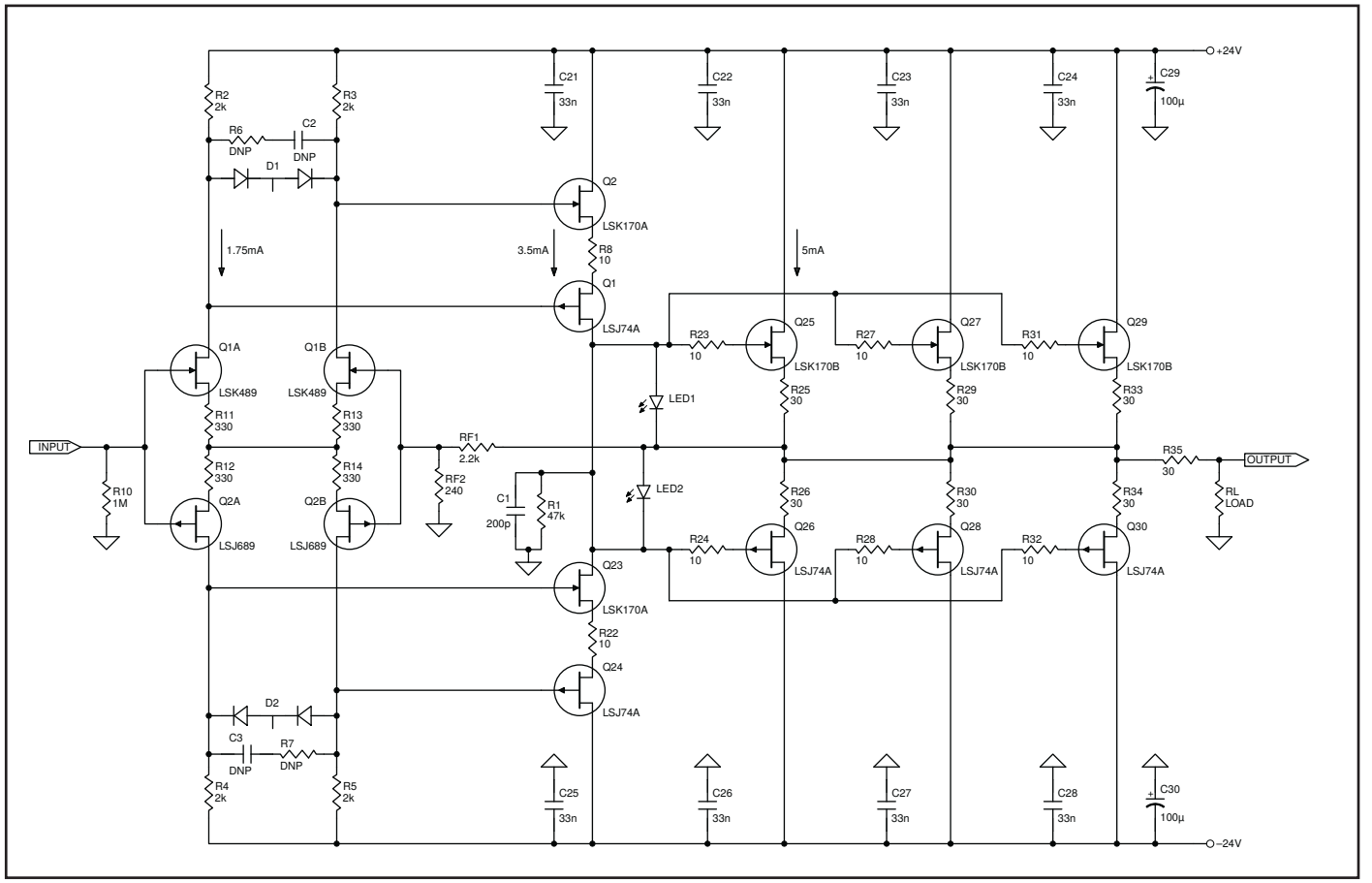


Figure 4: This is the schematic for the all-JFET amplifier.

power dissipation, gate-to-source breakdown voltage, gate-to-source leakage current, common source input capacitance, and common source reverse transfer capacitance.

DC Measurements for the LSK170, the LSJ74, the LSK489, and the LSJ689

I measured the distribution of the JFET drain saturation current I_{DSS} as a function of pinch-off voltage V_p for different samples of the LSK170A, the LSK170B, the LSJ74A, the LSK489, and the LSJ689. I measured about 50 samples of each single device and about 10 samples of each dual device. (Excel spreadsheets with additional data is available in the Supplementary Materials section of the audioXpress website.) Next, I performed the measurements with the MCubed Electronix semiconductor analyzer. Measurement results are shown in **Figure 1** for single devices and in **Figure 2** for dual devices. Each dot on these graphs represents a particular device. Dots do not fall on a single line because of the variation in the JFET manufacturing process. It looks as though the LSJ74A devices have wider spread of I_{DSS} than the LSK170A devices, but the LSJ74A with higher I_{DSS} can be paired with the LSK170B.

The data shown in Figure 1 and Figure 2 were used to calculate transconductance at V_{GS} = 0. Drain saturation current I_{DSS} as a function of transconductance at V_{GS} = 0 for selected LIS and Toshiba JFETs is shown in **Figure 3**. It is clear that for each complementary pair, the P-channel JFET has higher transconductance then N-channel device at the same I_D. I_{DSS} matching for complementary devices does not always lead to equal transconductance and compensation of even harmonics.

Circuit Description

Figure 4 shows the all-FET amplifier schematic. **Photo 1** shows the evaluation board. The circuit follows design principles outlined

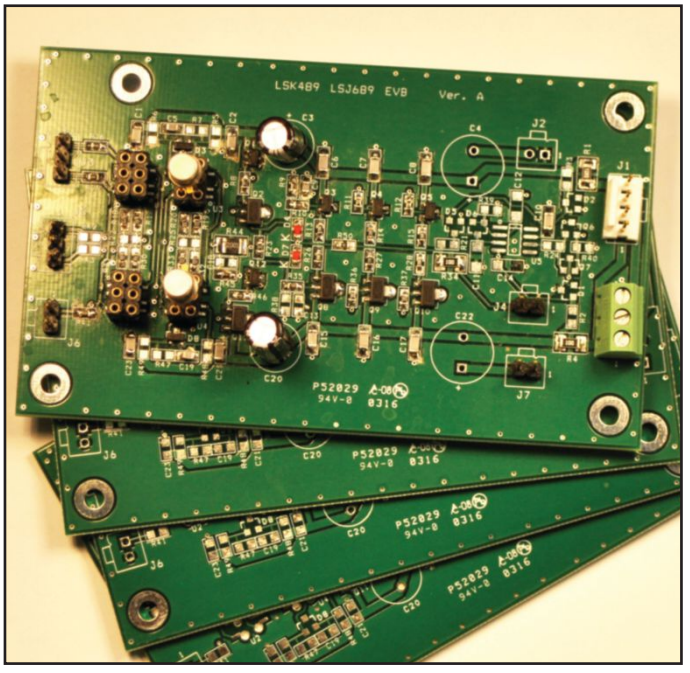
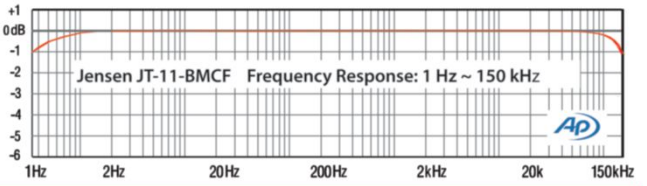


Photo 1: Pictured is the all-JFET amplifier evaluation board.



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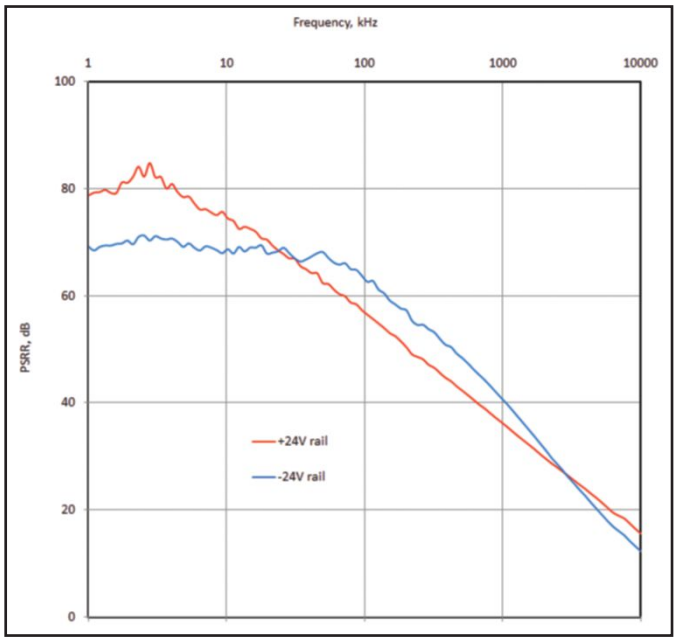


Figure 5: The power supply rejection ratio (PSRR) is referred to the input as a function of frequency.

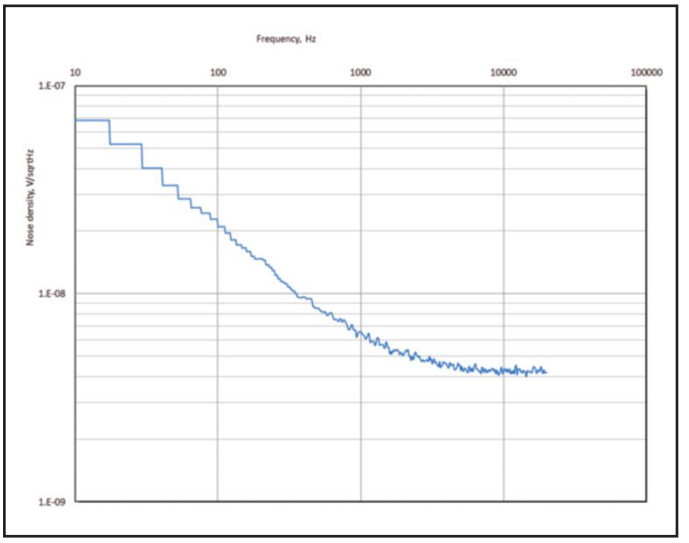


Figure 6: The input voltage noise density is shown as a function of frequency. Input voltage noise density shown at 1 kHz is about 6 nV/√Hz.

by John Curl and Erno Borbely for JFET amplifiers (see Resources).

The input stage is a complementary differential circuit. The differential gain stage is realized with dual transistors Q1 and Q2. Each of these JFETs is working with 1.75 mA drain current, which is set by source resistors R11–R14. Q1 and Q2 are dual monolithic JFETs, the LSK489 and the LSJ689 respectively. JFETs inside these pairs are matched closely thus input DC offset voltage of the all-JFET amplifier is low. Measured DC offset voltage is below 1 mV. Such low offset allows omitting the DC servo circuit for certain applications.

The second stage is the dual differential amplifier circuit (see T. Davis, W. Kulas, R. Lohman, and G. Sziklai patents in Resources). Each differential amplifier has two JFETs of opposite polarity. The sources of the JFETs are connected with a resistor (R21 or R22). Drain of the first JFET (Q22 or Q23) is connected to the load R1 and drain of the second JFET (Q21 or Q24) is tied to the supply rail. The second stage couples to the input stage through drain resistors R2–R5. The second stage operating current value is slightly lower than I_{DSS} and is about 3.5 mA.

The open-loop gain is set by transconductance of the input and second stages and resistive loads for the input (R2-R5) and second stages (R1). With R1 = 47 kΩ, the open-loop gain is 63 dB. Open-loop frequency is set by capacitive load (C2 and input capacitance of the output stage) of the second stage. With 200 pF the bandwidth of a closed-loop is around 15 kHz. The amplifier is stable with the closed-loop gain above five. For the lower closed-loop gain series, RC network (C2, R6 and C3, R7) should be inserted between input stage drain resistors (R2, R3 and R4, R5).

The second stage output is buffered from the amplifier load with a complementary source follower stage constructed with transistors Q25–Q30. Each output device operates slightly below I_{DSS} with the drain current close to 5 mA. The total bias current is 15 mA and the output stage works in Class A with light loads

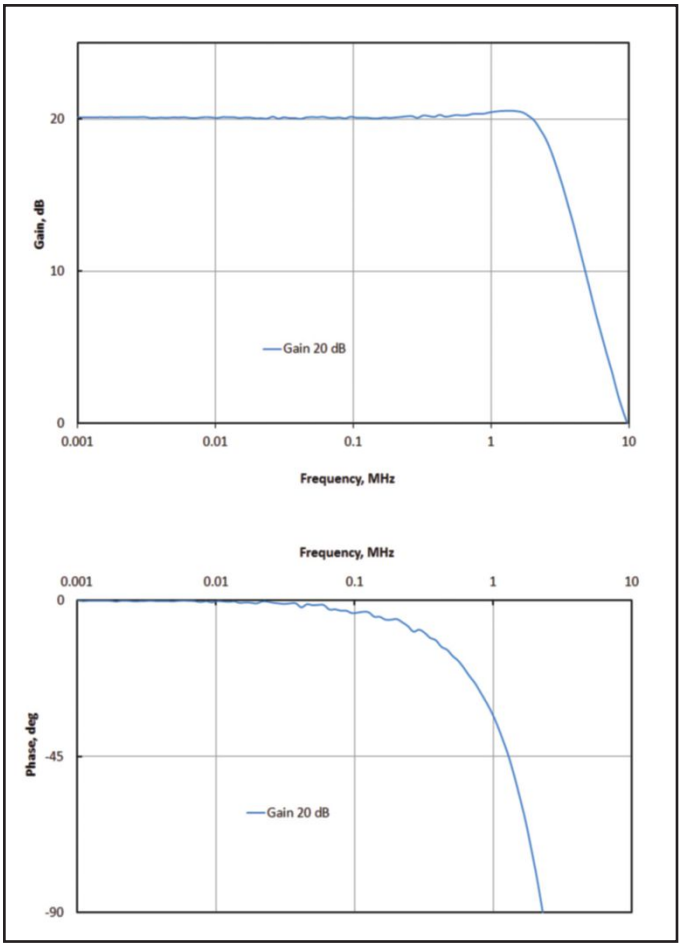


Figure 7: This is the closed-loop gain and phase for the all-FET amplifier with gain x10 (20 dB) as a function of frequency. R2, R3, C6, and C7 are not stuffed.

(above 600 Ω). The amplifier can drive loads down to 150 Ω, with no increase in total harmonic distortion (THD). Source resistors in the output stage are carefully selected to prevent distortion increase while switching from Class-A to Class-B operation with heavier loads. Each output device has a large copper pad for better heat dissipation. Light emitting diodes (LED1 and LED2) limit the voltage on the gates of output devices, thus limit the output current, and provide output short circuit protection. The current limit is about 85 mA, and the amplifier can drive 100 Ω load with up to 6 V_{RMS}. Small resistors are included in series with gate of each output JFET to suppress any potential parasitic oscillation in the output stage.

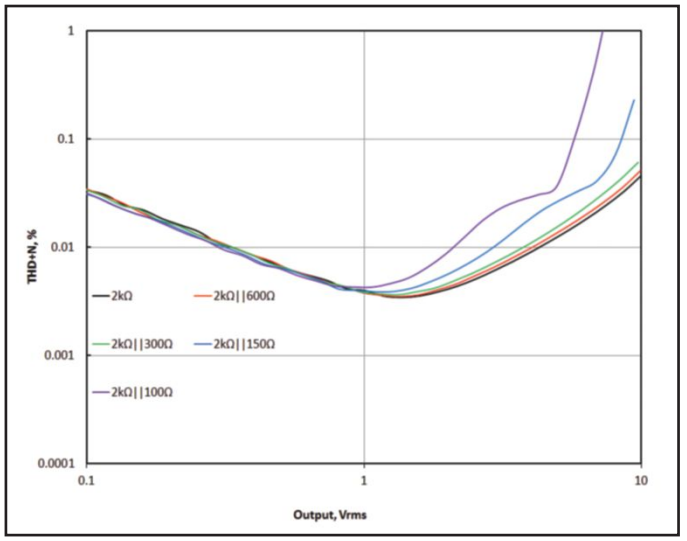


Figure 8: The THD+N is shown at 10 kHz for gain x10 (20 dB) and different loads as a function of output amplitude.

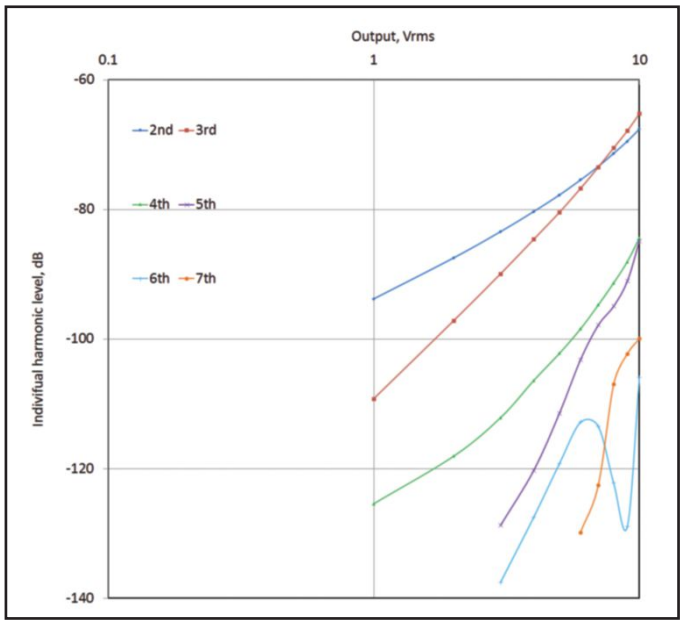
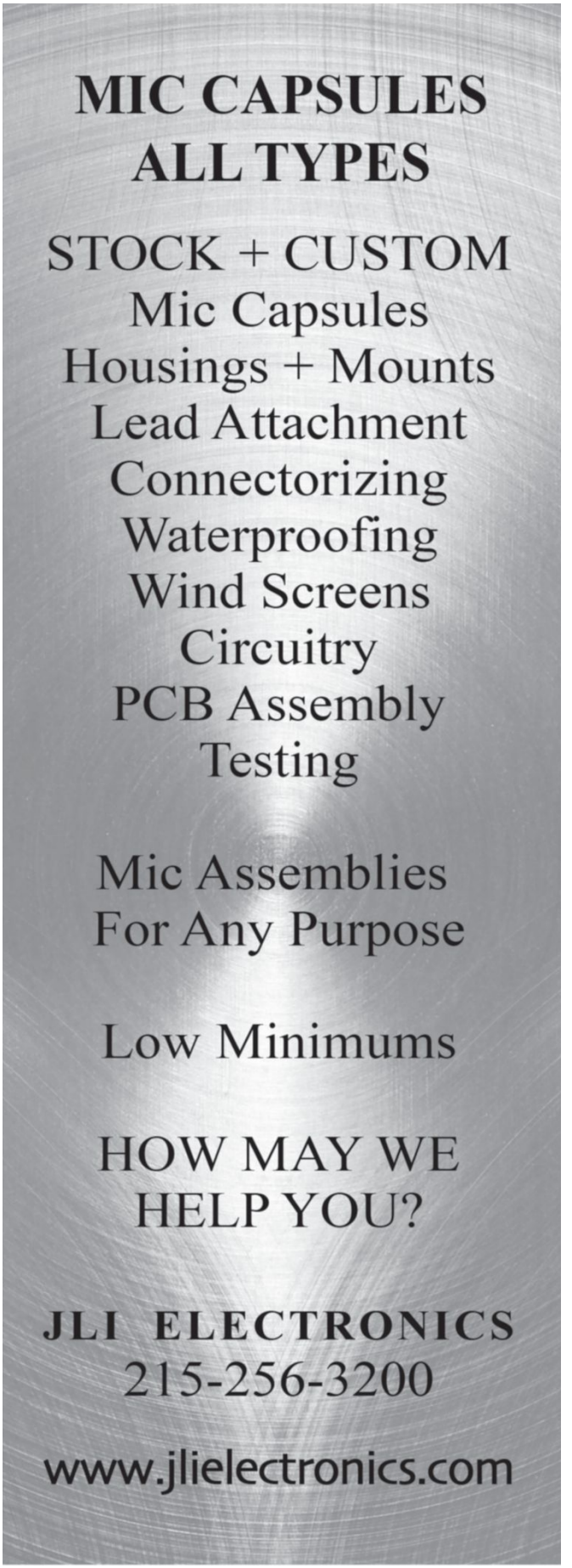


Figure 9: The individual harmonic level for gain x10 (20 dB) is shown as a function of output amplitude. The fundamental frequency is 10 kHz, load is 300 Ω.



The closed-loop amplifier bandwidth (closed-loop gain is equal to 20 dB) is wider than 2 MHz. Frequency response and distortion for different gain setting will be discussed later in detail.

The circuit shows good power supply rejection. Power supply rejection ratio (PSRR) is about 60 dB at 100 kHz. **Figure 5** shows the plot PSRR vs. frequency for the all-JFET amplifier.

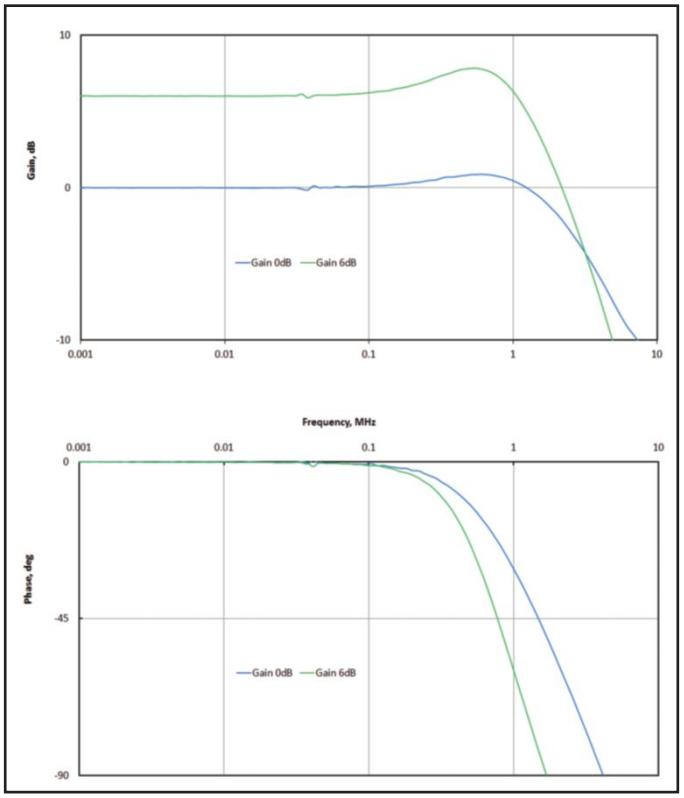


Figure 10: The closed-loop gain and phase is shown as a function of frequency for gain x2 (6 dB) and x1 (0 dB) with series RC (C2, R6 and C3, R7) networks (C2 = C3 2200 pF, R6 = R7 100 Ω).

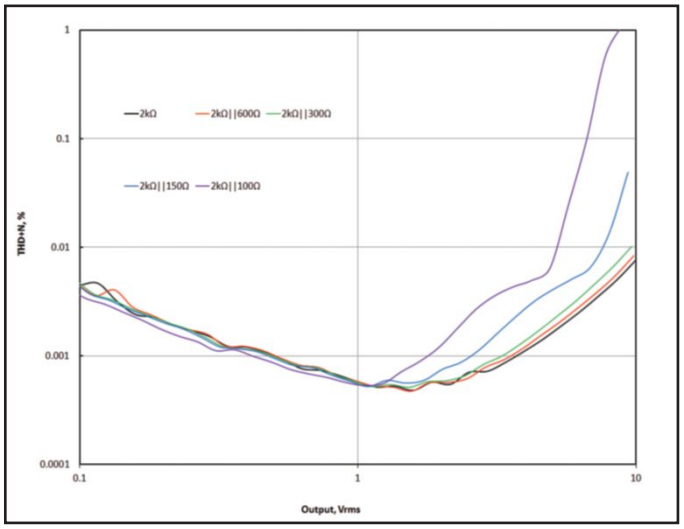


Figure 11: THD+N is shown at 10 kHz for unity gain and different loads as a function of output amplitude.

The circuit also shows good noise performance. Equivalent input noise in 10 Hz to 22 kHz frequency range is 0.7 μV or -123 dBV. The noise is 123 dB below a reference point of 1 V (0 dBV). **Figure 6** shows the amplifier's input voltage noise density.

Voltage noise density for source resistors R11–R14 is 2.3 nV/√Hz. The total noise density for a single LSK489 with 330 Ω source resistor is:

$$\sqrt{\left(\frac{1.8\text{nv}}{\sqrt{\text{Hz}}}\right)^2+\left(\frac{2.3\text{nV}}{\sqrt{\text{Hz}}}\right)^2}=\frac{2.9\text{nV}}{\sqrt{\text{Hz}}}$$

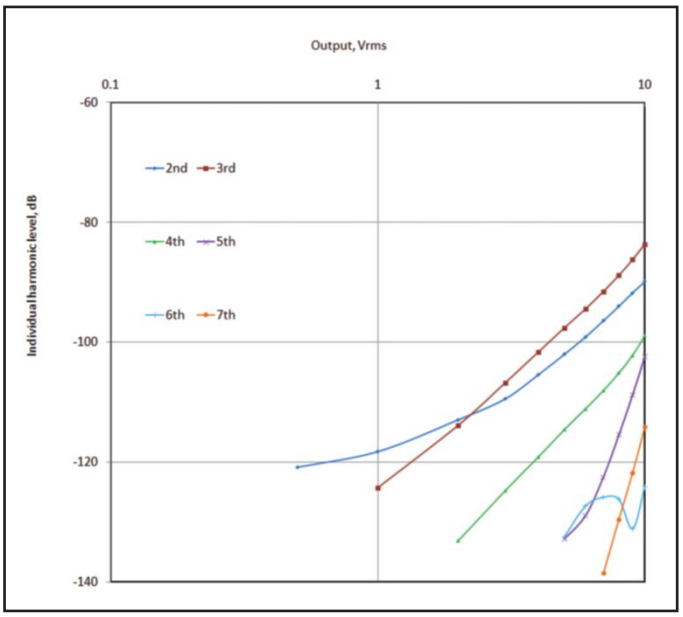


Figure 12: The individual harmonic level is shown as a function of output amplitude for unity gain. The fundamental frequency is 10 kHz, the load is 300 Ω.

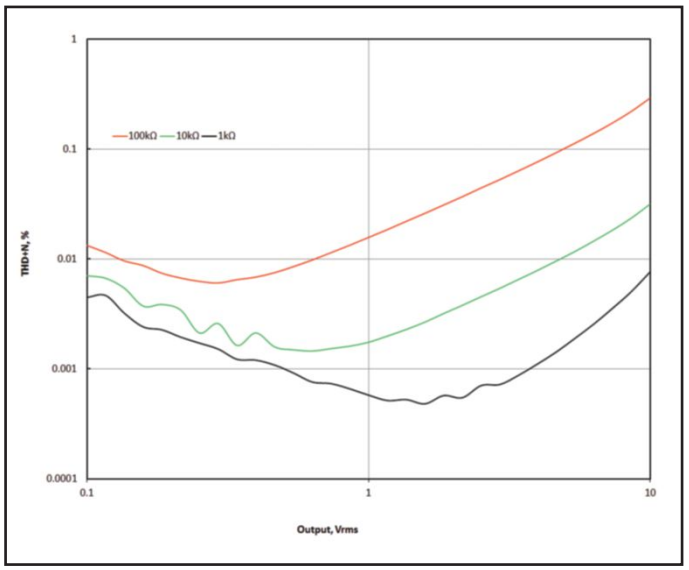
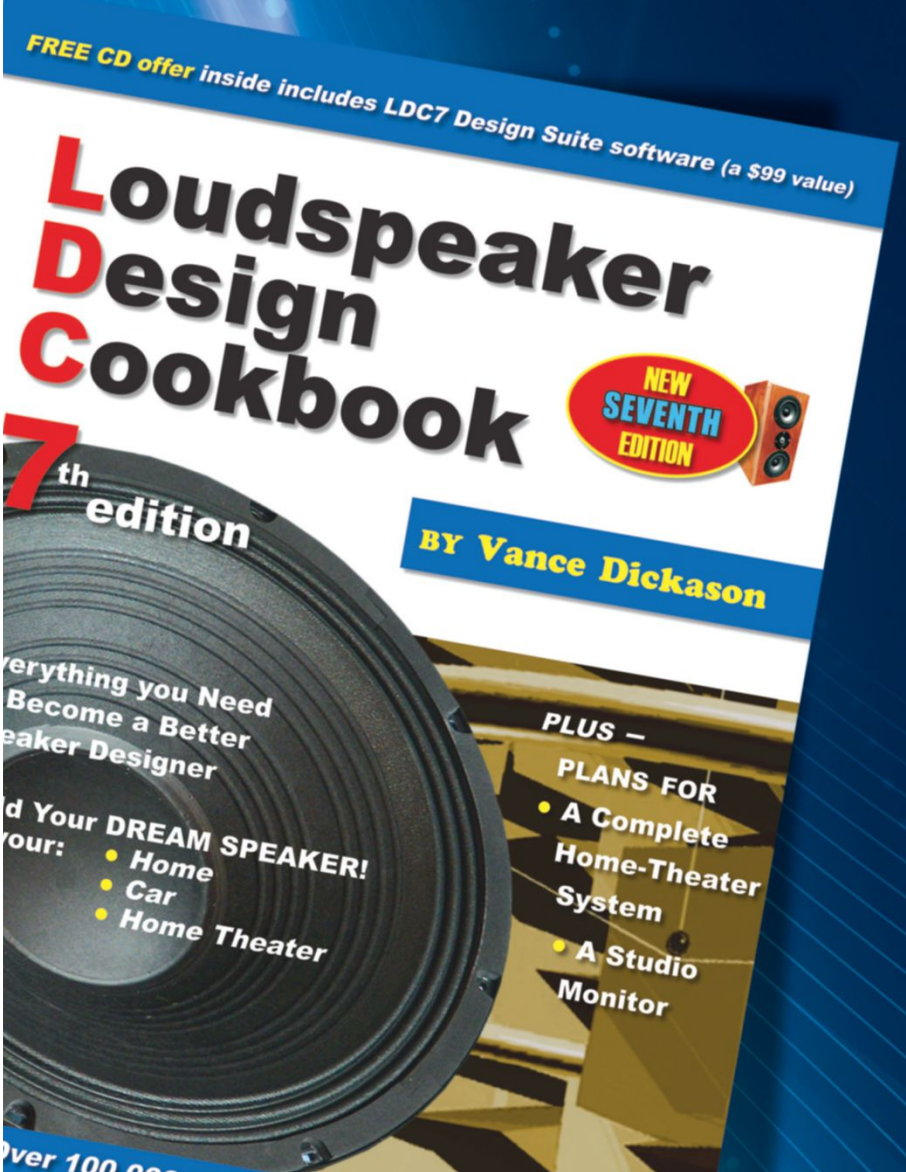


Figure 13: THD+N at 10 kHz is shown as a function of output amplitude for unity gain and different signal source resistance (1 kΩ, 10 kΩ, and 100 kΩ), the load is 2 kΩ.

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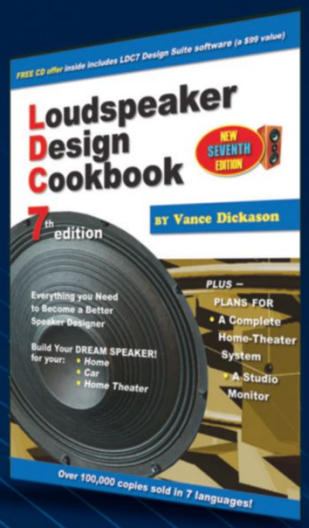
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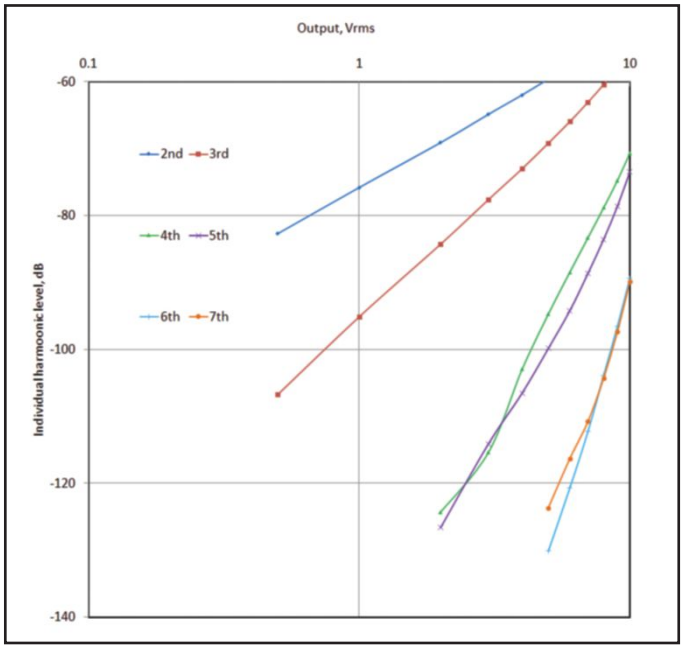


Figure 14: The individual harmonic level is shown as a function of output amplitude for unity gain and 100 kΩ source resistance, the load is 2 kΩ.

Measured input voltage noise density at high frequency is 4nV/√Hz. Measured noise is higher than calculated. The input stage has low gain (about 6 dB) and thermal noise of drain resistors (R2–R5) makes significant contribution to the amplifier's total noise. The noise can be reduced by implementing active loads, reducing value of source resistors (R11–R14), by paralleling several JFETs and by using LSK389 with a future dual monolithic 1nV/√Hz low noise, high-transconductance P-channel device.

Amplifier Performance for Gain of 10

Figure 7 shows the closed-loop frequency and phase response for all-JFET amplifier. **Figure 8** shows THD at 10 kHz plus noise for different load as function of frequency. The distortion profile is consistent for different loads. Distortion for 100 Ω load increases for the output above 6 V_{RMS}. This sharp rise is caused by current limiting action of LED1 and LED2.

Figure 9 shows the harmonic distortion spectrum. It looks as if, with audio reproduction, the quality of the harmonic distortion is important with respect to its quantity and different subjective sonic results correlate with different individual harmonic distortion profiles (see Jean Hiraga's article in Resources). The optimum harmonic distribution is characterized by a progressive decreasing harmonic distribution pattern with the absence of higher-order products. In this amplifier level of higher-order harmonics (fourth,

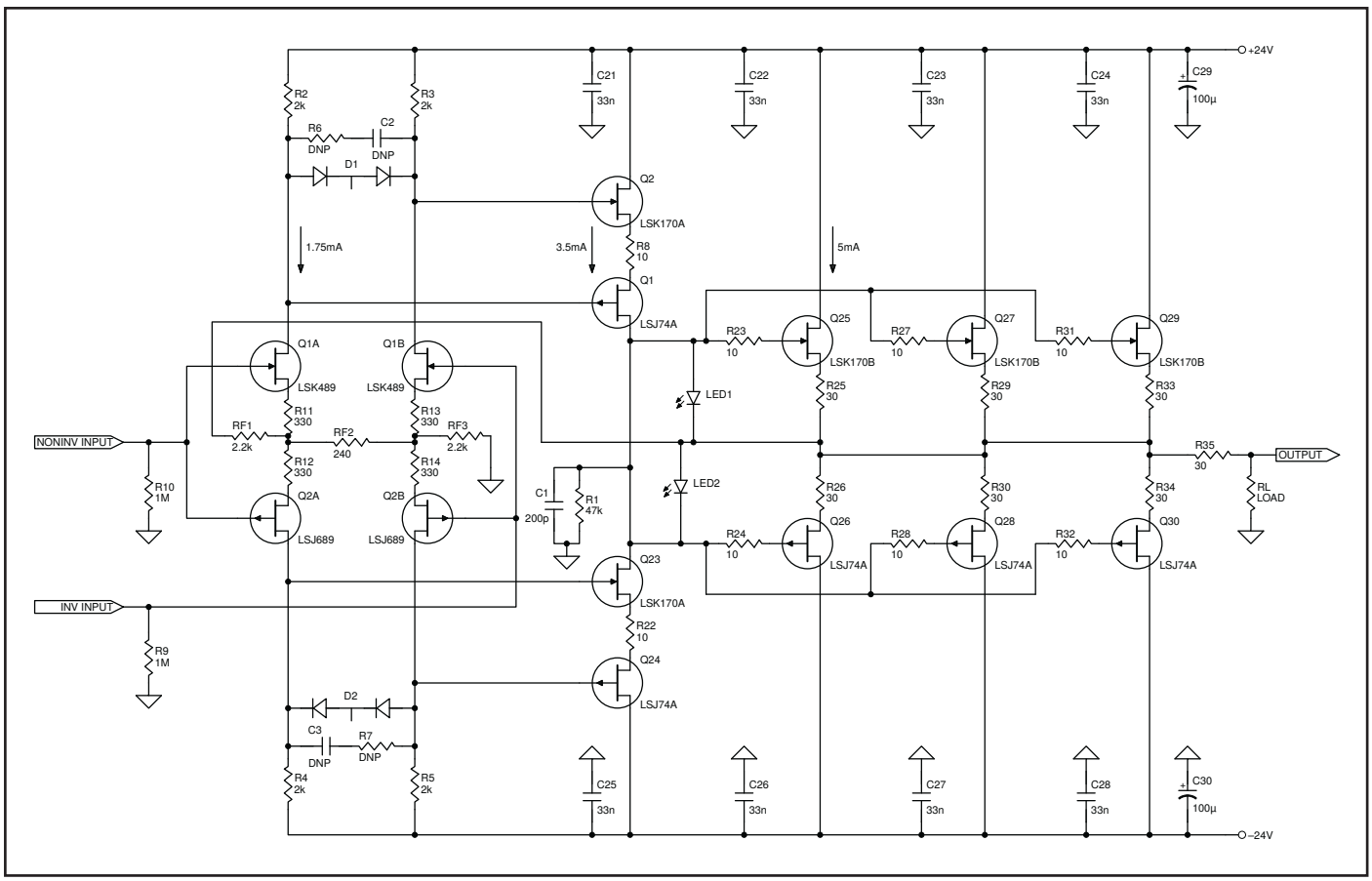


Figure 15: The schematic shows the balanced all-FET amplifier with current feedback.

fifth, sixth, and seventh) is at least 20 dB below than the level of the second and third. The second harmonic is prominent at low and moderate output level (below 7 V_{RMS}). The third harmonic prevails slightly at higher output level.

Amplifier Performance for Gain of 2 (6 dB) and Unity Gain (0 dB)

Figure 10 shows closed loop frequency responses for the all-JFET amplifier. **Figure 11** shows THD at 10 kHz plus noise for different load as function of frequency. THD is under 0.01%. **Figure 12** shows the harmonic distortion spectrum. Extra 20 dB of loop gain pushes all the harmonics down. There is still considerable gap (about 12 dB) between the second, the third, and higher order harmonics. Lower order harmonics (second and third) behave in the same way as for amplifier with 20 dB gain.

Importance of the Signal Source Impedance

The LSK489 and the LSJ689 dual JFETs have small gate source (~4 pF) and gate drain capacitance (~1.5 pF). The frequency response of an all-JFET amplifier with unity gain and a signal source resistance of 100 kΩ has high-frequency rolloff with a -3 dB frequency about of 200 kHz. The input capacitance of the all-JFET amplifier with unity gain is about 8 pF.

Figure 13 shows the total harmonic distortion plus noise (THD+N) as a function of output amplitude for gain x1 (0 dB) and different source resistance (1 kΩ, 10 kΩ, and 100 kΩ. One can observe the distortion rise even when the source impedance is equal to 10 kΩ. In "LSK489 Application Note," Bob Cordell suggested that a lower-capacitance device can help to create a circuit that is much less susceptible to intermodulation distortion with higher source resistance. However, the input capacitance of the JFETs is voltage dependent, and even being low, when driven from high impedance it can cause distortion, as shown in Figure 1. If extremely low distortion is required with higher source resistance and large excursion of the input signal, it may be obtained by cascoding of the input JFET with another JFET or with bipolar transistor (refer to Erno Borbley and John Curl articles in Resources).

Figure 14 shows the measurements of the level of each harmonic. It is generally accepted that reverse transfer capacitance of JFET is strongly nonlinear and cause high order distortion products. While one can observe increase in higher order distortion, second and third harmonics are prevailed with a considerable margin. This is due the fact that the

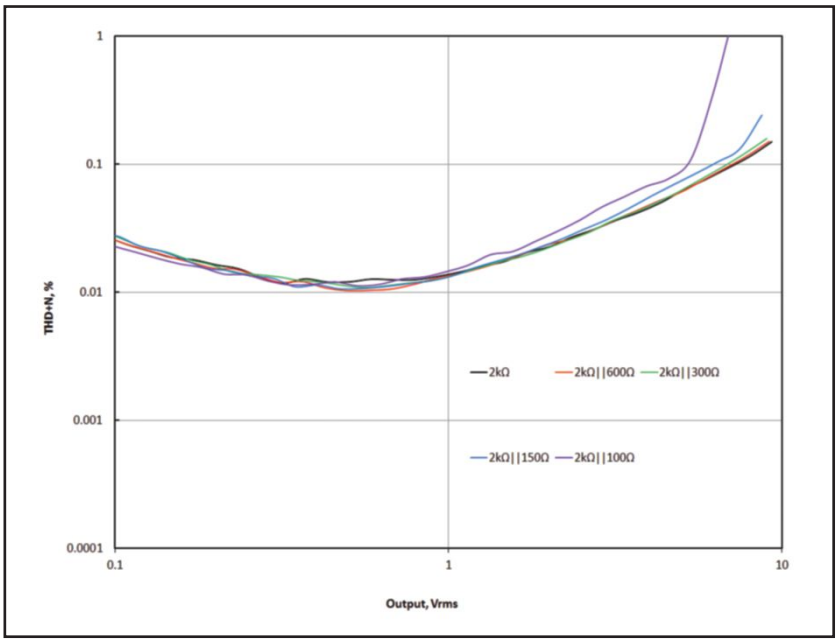


Figure 16: THD+N at 10 kHz is shown as a function of output amplitude for the instrumentation amplifier circuit, presented in Figure 3, gain 20 dB.



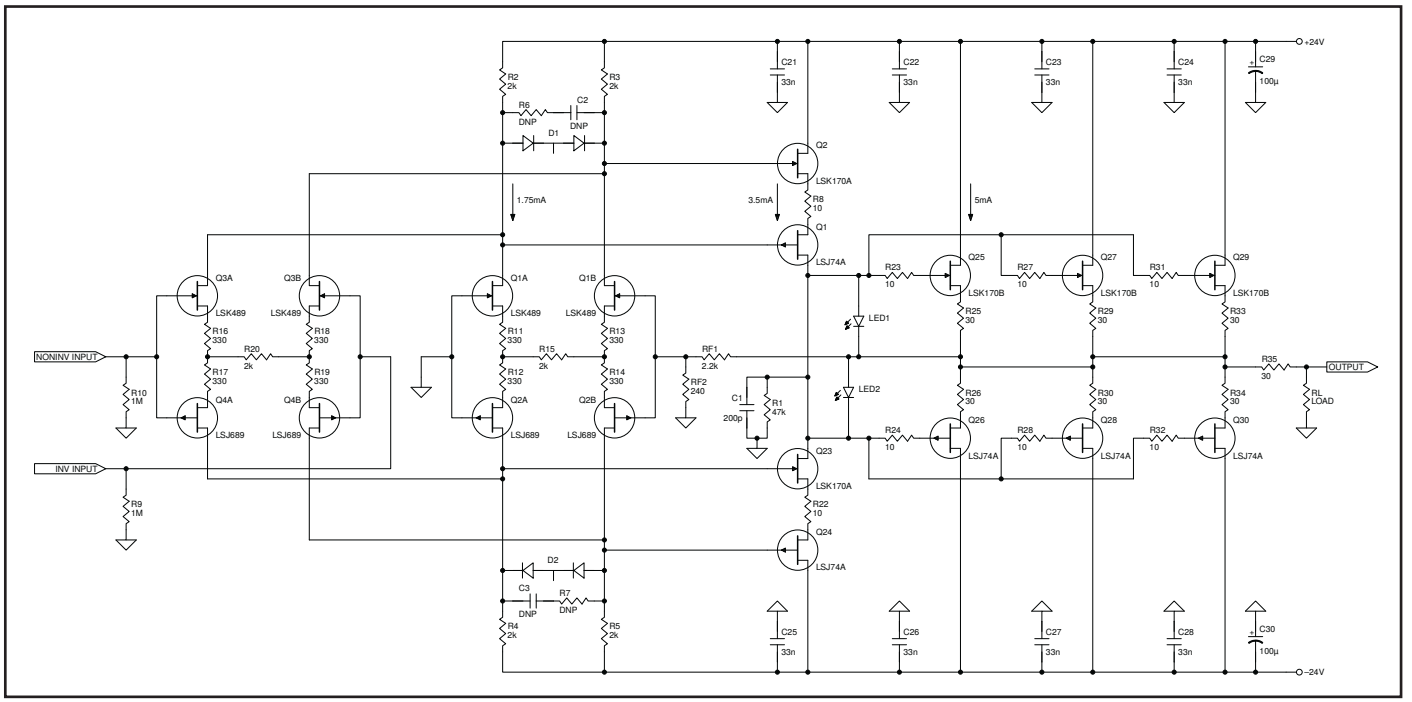


Figure 17: This schematic shows a balanced all-FET amplifier with current summing.

junction capacitance varies inversely as a square or cube root of applied reverse voltage, depending on doping profile.

Balanced All-JFET Amp with Current Mode Feedback

An amplifier with balanced input responds only to the potential difference between its two terminals—inverting and noninverting inputs. Balanced input may be preferable because it rejects ground loop-induced noise and suppresses the interference from the environment. There are many realizations of balanced inputs—with transformer, differential amplifiers, etc. **Figure 15** shows a different solution (refer to D. F. Bowers and R. Demrow articles in Resources). All-JFET amplifier employs current mode feedback (Resistors Rf1–Rf3) to the input JFET sources.

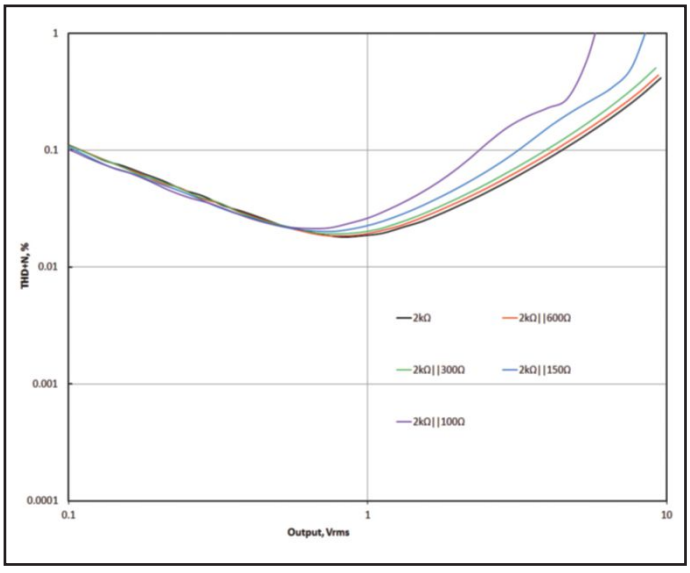


Figure 18: THD+N is shown at 10 kHz as a function of output amplitude for the circuit presented in Figure 5, gain 20 dB.

Figure 16 shows the measurements results for this circuit for 20 dB of gain. Distortion is 6 to 10 dB higher than distortion in noninverting amplifier with single-ended input. Resistor Rf2 between the sources reduces the input stage transconductance and the open loop gain. Additional distortion is caused by asymmetry of the feedback network Rf1–Rf3.

Balanced All-JFET Amp with Current Summing

Figure 17 shows a balanced All-JFET amplifier with current summing (refer to the A. P. Brokaw and P. M. Timko article in Resources). Q3 and Q4, and R16–R20 form the first transconductance stage, and the second transconductance stage consists of Q1, Q2, and R11–R15. Drains of the transconductance stages are cross coupled so that the currents are subtracted and the result is fed back in the correct polarity. Input transconductance stage works without global feedback, and to obtain low closed-loop distortion both transconductance stages should have identical transfer characteristics. Resistors R15 and R20 provide local feedback in transconductance stages making their transfer less dependent on JFET pairs matching.

Figure 18 shows the measurements results for this circuit for 20 dB of gain. One can observe elevated noise level, caused by local feedback resistors (R15, R20) in the transconductance stages. The

overall gain is set by Rf1, Rf2 and the transconductance ratio of both transconductance stages. To improve overall distortion one can use dual parts in such way that first transistor in the pair belongs to first transconductance stage and the second transistor belongs to the second transconductance stage (opposite to the figure shown in Figure 17). This may cause DC shift, which can be removed by a servo circuit.

It might be worthwhile to combine a JFET input pair, which can be optimized for a specific source (e.g., a magnetic cartridge) and a BJT feedback pair. Another interesting application is differential current to voltage converter. In this case, current (e.g., from DAC) can be injected in the source network of the input pair.

Summary

This is an All-JFET amplifier. Measurements have shown that amplifier harmonic pattern remains unchanged with the output load. It appears that the distribution of harmonics with output level match to the pattern that was produced by the ear. This may provide a certain auditory benefit.

Project Files

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

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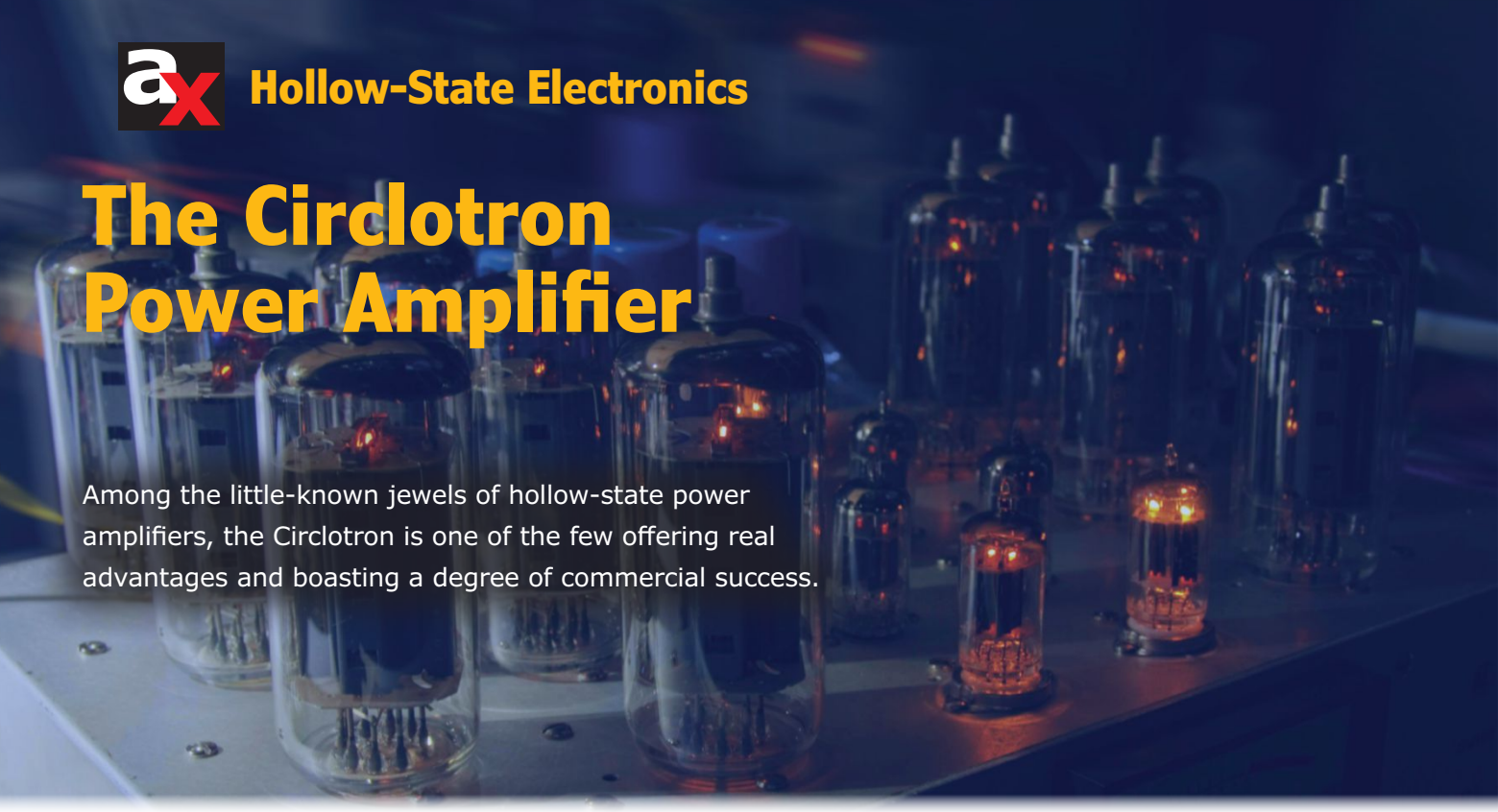
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Trendsetter Electronics (Linear Integrated Systems distributor) www.trendsetter.com/Linear-Integrated-Systems-s/36101.htm.



The Circlotron Power Amplifier



Among the little-known jewels of hollow-state power amplifiers, the Circlotron is one of the few offering real advantages and boasting a degree of commercial success.

By
Richard Honeycutt
(United States)

Most hollow-state power amplifiers have two characteristics in common: They accept inputs with maximum amplitudes of about a volt or so, and they are designed to feed loudspeaker loads having nominal impedances of typically 4 Ω to 16 Ω.

More specialized units have been built, including those that are driven to full power by smaller or larger inputs, as well as those designed for special loads (e.g., the capacitive load represented by an electrostatic loudspeaker). But let's concentrate on the more common designs. Hollow-state power amps can be built using any of several topologies. The oldest and least expensive is the one using a Single-Ended Triode (SET), shown in **Figure 1**.

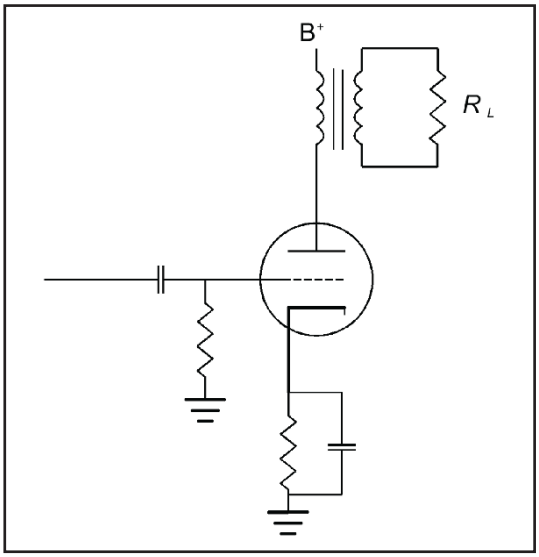


Figure 1: The single-ended triode (SET) output stage is the oldest and simplest topology for a hollow-state power amplifier.

SET Topology

While some audio enthusiasts prefer the sound of the SET amplifier, presumably because of its particular distortion signature, it does have some limitations. First, a SET by definition operates as a Class A amplifier, and the highest possible theoretical efficiency (η) for a Class A amplifier is 50%. Under actual operating conditions, driving a loudspeaker, maximum achievable efficiency can be as low as 25%. This means that the maximum output power is no more than the maximum plate dissipation (η = 50%), or perhaps half the maximum plate dissipation (η = 25%).

For a 300B Gold triode, the maximum plate dissipation is 40 W (36 W for the original Western Electric 300B); and 46 W for a KT90 in triode connection. So the maximum output power for a SET is typically less than 20 W. This is only the thermal limit—the KT90 spec sheet gives 12 W as the maximum power output, with 14% total harmonic distortion (THD).

Fans of the SET may be quick to point out that these amplifiers sound much louder than the measured power output would suggest, but this is a psychoacoustic phenomenon: Added distortion makes program material sound louder, because at high SPLs, both HD and IMD are generated in the ear,

so the ear-brain mechanism associates distortion with loudness.

Aside from limited power output, the single-ended output stage requires the output tube's Q-point to be near the center of the tube's most linear range, which is about 62 mA for the 300B and 140 mA for the KT90. This constant direct current causes some heating of the power transformer, depending upon the primary resistance, and also produces a steady magnetic field that can cause transformer saturation at high signal levels. In order to prevent transformer saturation, output transformers for single-ended stages are usually produced with a gap in the magnetic circuit. This gap increases the leakage inductance and reduces the primary's actual inductance, thus requiring more turns to achieve good low-frequency response. The maximum power output of a single-ended stage can be increased by paralleling two or more output tubes (see **Figure 2**). The efficiency, distortion, and transformer-size limitations remain, however.

Push-Pull Output Stages

For applications other than the listening rooms of audio enthusiasts who fancy a lot of second-harmonic distortion, push-pull output stages (see **Figure 3**) are more suitable. Because the output tubes are operating in a balanced configuration—input and output signals for the upper and lower tubes are out of phase, but the two output tubes are biased for identical operating points—even-order harmonic distortion is cancelled.

If the output tubes are biased at cutoff, requiring a positive instantaneous grid-to-cathode signal to turn the tube on—the operation is referred to as Class B. With a push-pull amplifier in Class B operation, the maximum efficiency is 78.5%, making the thermally limited maximum output power 3.14 times that of a single-ended stage using one output tube. (The value of 3.14 is the product of 1.57 times the efficiency and twice the number of output tubes.) Also, the THD is much lower than that of a single-ended stage.

The problem with Class B amplifiers for audio is that when the signal crosses zero volts, both tubes are turned off, and the operation very near the cutoff point is rather nonlinear. This leads to a problem called "crossover distortion," which is particularly irritating to the listener. The output wave of a Class B amplifier driven to clipping is shown in **Figure 4**. Crossover notches are visible near the zero-volt axis.

One solution to crossover distortion is to bias the output tubes for Class A operation, but this forfeits

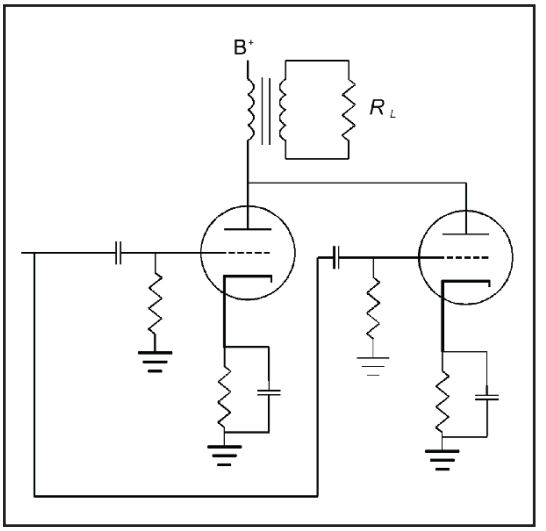


Figure 2: Paralleling output tubes provides more output power.

the efficiency advantage of the push-pull topology, even though the even-order harmonic cancellation is retained. Another approach is to bias the output tubes just barely into conduction, so that at zero-crossing, both conduct. If the bias is optimally set, crossover distortion is greatly reduced. This is called Class AB operation, and it is sometimes subdivided into Class AB_1 and Class AB_2, with the former having a slightly higher quiescent current and lower crossover distortion. Class AB operation entails a small (<7%) efficiency penalty, compared to Class B.

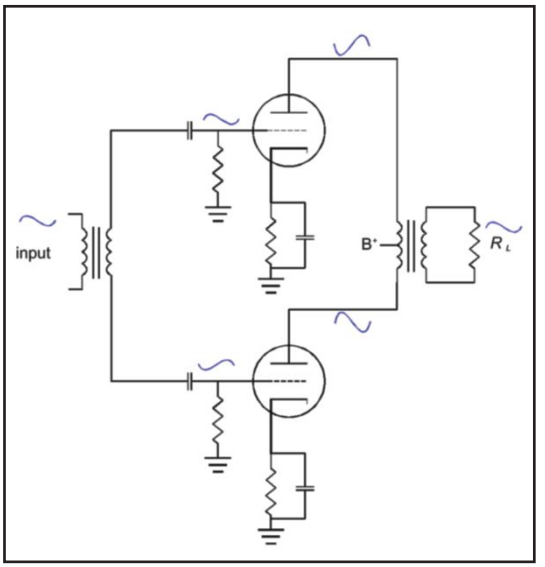


Figure 3: The push-pull topology offers several advantages over the single-ended one.

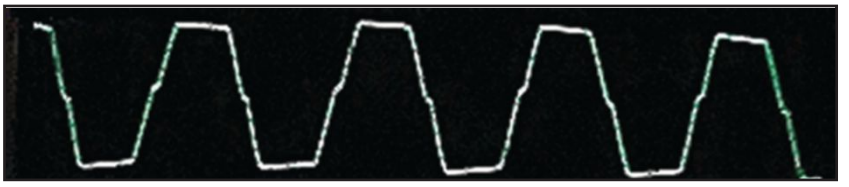


Figure 4: Crossover distortion in a Class B amplifier shows up as a discontinuity when the signal crosses zero volts.

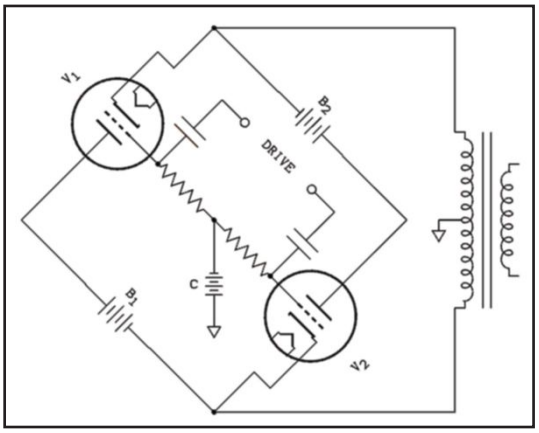


Figure 5: This variant of a push-pull stage was introduced in 1951. (Schematic courtesy of Max Robinson)

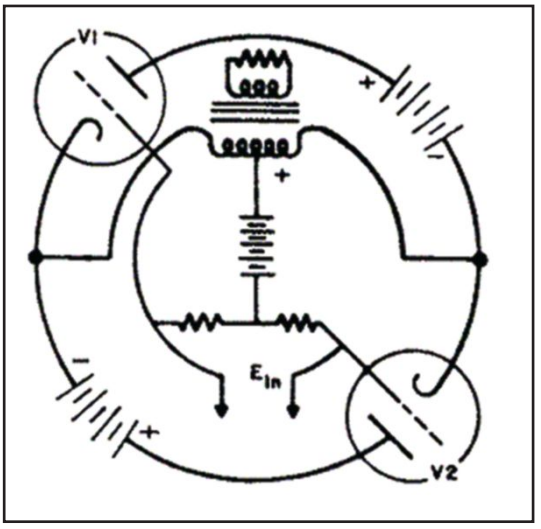


Figure 6: Drawing the schematic in this way clarifies its operation and shows the appropriateness of the name "Circlotron." (Schematic courtesy of Bob Turnbull)

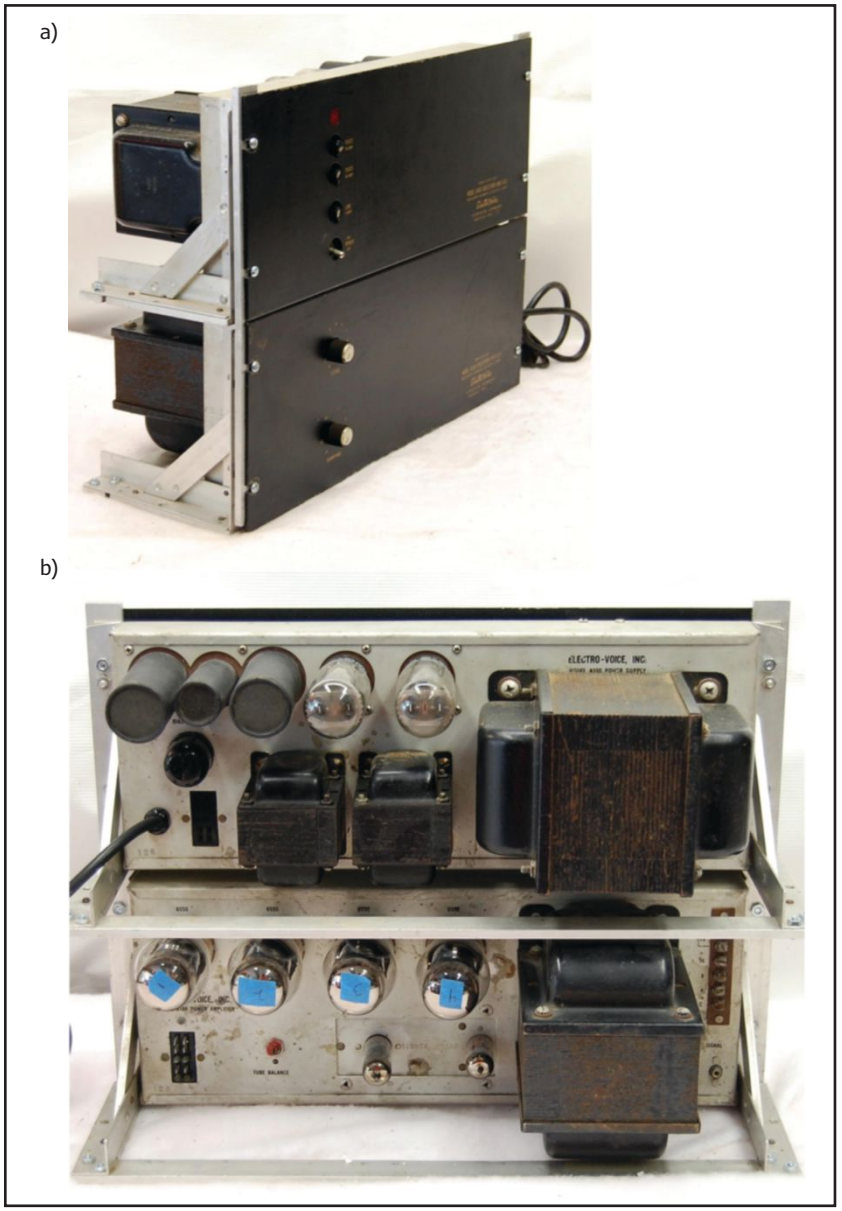


Photo 1: The Electro-Voice Model A100 Circlotron amplifier (a) consisted of two separate chassis—the amplifier and the power supply (b).

In any class of operation, the DC in the output transformer primary is split, with equal amounts flowing in opposite directions, thus preventing the creation of a steady magnetic field that could cause transformer saturation. Thus, output transformers for push-pull stages need no gap in the magnetic circuit, allowing fewer turns to be used to achieve the necessary primary inductance. But there is still a small heating effect due to the DC in the primary.

The Evolution of the Circlotron

In 1951, Cecil T. Hall filed a patent (US Patent 2,705,265) for a variant of the push-pull output stage. The circuit is shown in **Figure 5**. Tapio Köykka filed a patent (Finnish Patent 27332) for a similar circuit in 1952. He later filed a second patent (Finnish Patent 29643) based on improvements he made to it in 1955. Satoshi Shimada of Japan invented a similar circuit in 1952, which he called the Cross Shunt Push Pull (CSPP) topology. In 1954, Alpha M. Wiggins of Electro-Voice filed a patent for this circuit, but later Electro-Voice secured a license to manufacture under the Hall patent. Thus, it was Electro-Voice that first commercialized the circuit under the trademarked name "Circlotron." **Figure 6** shows the schematic redrawn in the form that makes it clear why the name Circlotron was adopted. (The input signal in Figure 6 is assumed to come from a floating AC source, so the coupling capacitors shown in Figure 5 do not appear in Figure 6.)

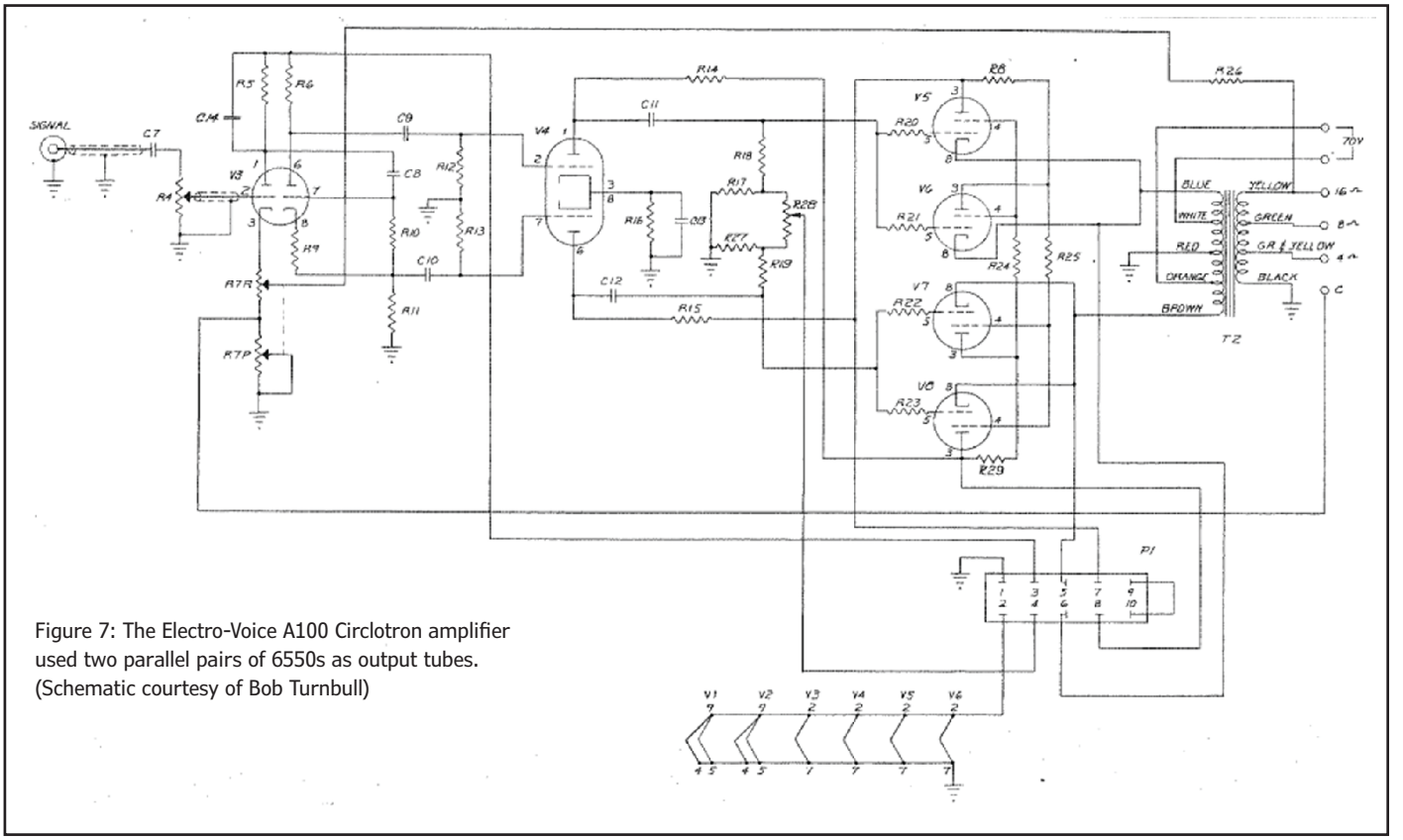


Figure 7: The Electro-Voice A100 Circlotron amplifier used two parallel pairs of 6550s as output tubes. (Schematic courtesy of Bob Turnbull)

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As you can see from the schematic, the Circlotron uses fixed bias for the output tubes, plus a separate floating B+ supply for each tube. The input signal is applied in balanced fashion from grid to grid, following a phase splitter as with an ordinary push-pull stage. The output is taken from cathode to cathode. Thus, the output stage provides current gain and power gain, but a voltage gain slightly less than unity. The lack of voltage gain in the output stage has prompted some designers to insert a balanced driver stage between the phase splitter and the output stage.

There is some disagreement about whether the Circlotron's output tubes can operate as cathode followers. A true cathode follower (i.e., a "common-plate" or "grounded plate" stage) has the plate maintained at AC ground via filter/decoupling capacitors. The only actual ground point in a Circlotron is the positive terminal of the grid-bias supply. But it is true, that for both the cathode follower and the Circlotron, the voltage gain is just under unity.

Circlotrons have been successfully built using pentodes rather than triodes. In fact, the Electro-Voice Circlotron amplifiers, which used pentodes, were offered in models ranging from 15 W to 100 W (see **Photo 1**). The flagship A100 boasted 100 W rated output power (200 W on peaks), a frequency response flat ±0.5 dB from 20 to 50,000 Hz, THD was less than 0.5%, and IMD was less than 1.2% at rated output. It could drive 4 Ω, 8 Ω, 16 Ω, or 70 V speakers, and included the option of a 600 Ω output. The damping factor was adjustable between 0.1 and 10 via a front-panel control. The schematic is shown in **Figure 7**.

Built primarily for studio use, the Circlotrons were moderately successful, but the relatively high cost and the thermal runaway problems of some early-production units hampered full market acceptance. Electro-Voice modified the design to eliminate the thermal runaway problems for later-production units, and provided field-modification instructions for past purchasers.

In spite of the disadvantage of requiring two B+ supplies, Circlotrons offer several advantages:

- There is no output tube DC in the transformer, a direct result of the balanced bridge circuit.
- All switching transients are eliminated through unity coupling of output tubes.
- Class AB_1 operation is effected, with the resultant improvements in power output, efficiency, and tube life.
- The primary impedance of the output transformer is one-quarter of that found in conventional amplifier output circuits, affording a vast increase in power output at extremes of the frequency spectrum.

In addition to Electro-Voice, other manufacturers of Circlotron amplifiers have included VOIMA, Philips, Carad, and MEL, in addition to at least one Russian company. Experimenters and designers of hollow-state audio equipment have perpetuated an interest in Circlotrons, and several websites focus specifically on this topology.

Bob Turnbull's website (http://circlotron.tripod.com) is very informative, especially concerning the history of the Circlotron. Turnbull worked for Electro-Voice in Buchanan, MI, in the 1980s and mid-1990s, and is now employed in automotive electronics. Much of the information on Turnbull's website came from the late John Overley, who was the Circlotron manufacturing engineer at the RME division of Electro-Voice in Peoria, IL.

Max Robinson's website (www.angelfire.com/electronic/funwithtubes/Amp-Circlotron.html) contains information about the Circlotron's operation, as well as the design details for a complete 50 W Circlotron amplifier and its power supply. Robinson describes himself as someone who "never lost the wonder."

Author's Note: I would like to thank Bob Turnbull and Max Robinson for their generous permission to use information and illustrations from their websites. The recent design/construction article by Gerhard Haas in the May 2017 issue of audioXpress *contains excellent details concerning the operation of the amplifier and its power supplies.*

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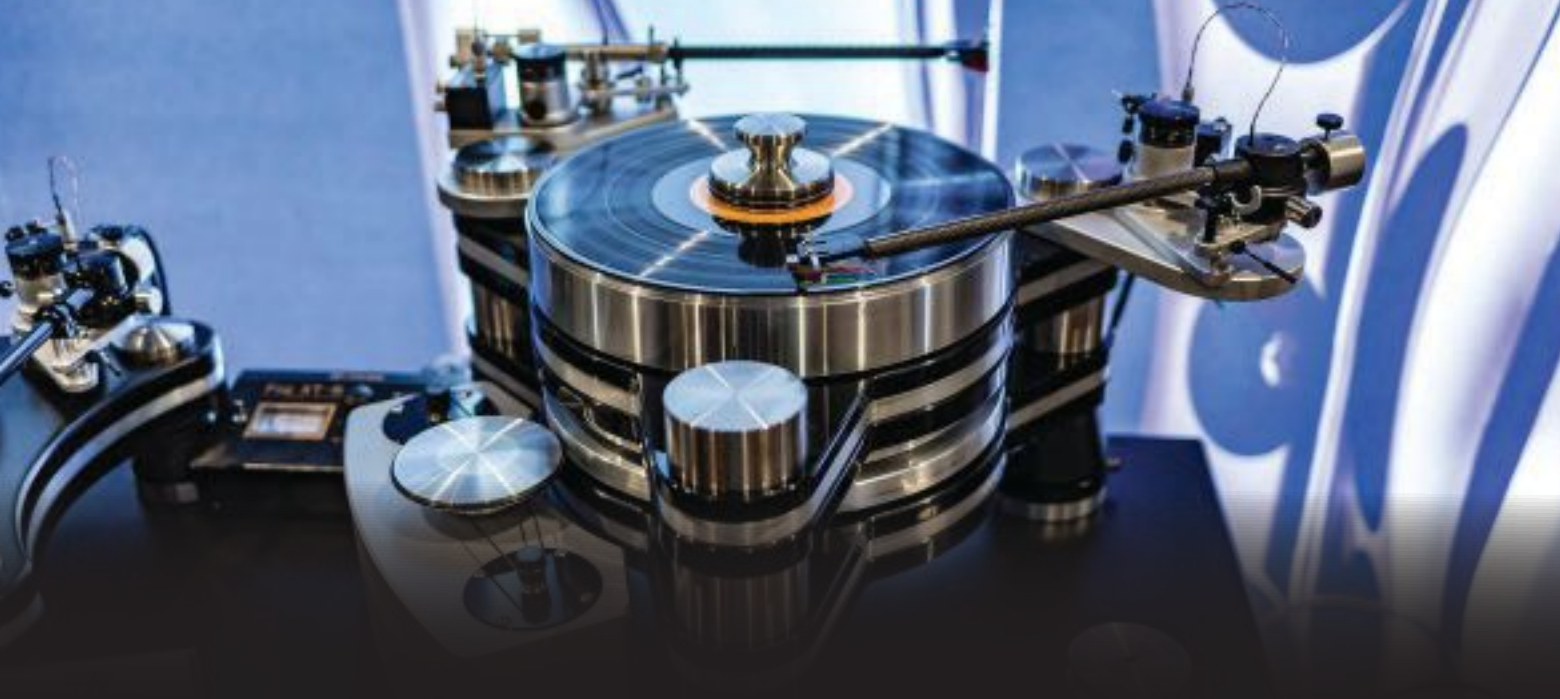
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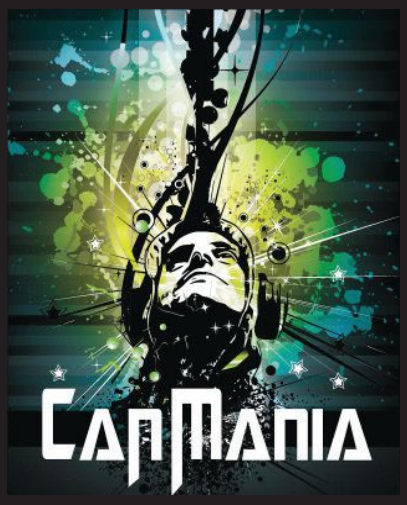
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Rocky Mountain Audio Fest
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The Annual Rocky Mountain Audio Fest (RMAF) is the largest consumer audio and home entertainment show in the US. Denver's own audio wonderland now features over 160 exhibit spaces representing more than 400 companies. CanJam at RMAF is a show-within-a-show. Organized by Head-Fi.org, it features the latest in headphone technology, which is the fastest growing segment of the audio industry.

October 11–14, 2017
Prolight+Sound Shanghai
Shanghai New International Expo Centre, Shanghai, China
www.prolightsound-shanghai.com

Prolight+Sound Shanghai returns in 2017 with an enlarged area of 45,000 m² in four halls (N1–N4). More than 600 companies and brands are expected to demonstrate a range of entertainment products and technologies at the fair. Two thematic zones, including the "Recording and Production Zone" and "AV Entertainment Solutions Zone," will facilitate the industry to tap into new market segments.

October 13–16, 2017
KTDC Hong Kong Electronics Fair (Autumn Edition)
Global Sources Electronics Show
electronicAsia
Hong Kong Convention and Exhibition Center, Hong Kong
www.hktdc.com/hkelectronicsfairae

Organized by the Hong Kong Trade Development Council, the show presents electronics products and services (e.g., audio-visual products, branded electronics, eco-friendly products, iWorld, packaging and design, navigation systems, new inventions, telecommunications products and testing, inspection and certification services). The show runs concurrently with electronicAsia, a professional trade fair for components, assemblies, and electronic production and display technologies jointly organized by Hong Kong Trade Development Council and MMI Asia Private, Ltd.

October 16–18, 2017
The Loudspeaker Sourcing Show
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The Loudspeaker Sourcing Show was created to connect suppliers directly with the companies that need their services. The exhibitors at this show provide OEM/ODM services in the manufacturing of loudspeakers and related products and components. This year, the show starts immediately after the international Hong Kong Electronics Fair (Autumn Edition) trade show.

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Following the acquisition of the former PALME Middle East show by Messe Frankfurt, the inaugural edition of Prolight+Sound Middle East 2016, featured 37 exhibitors from 11 countries representing more than 120 brands, and attracted more than 1,577 trade visitors from 50 countries. Prolight+Sound Middle East is now the fifth event in the Prolight+Sound international brand. Product launches, educational seminars, training programs, and a live outdoor demo area all shared center-stage throughout the three-day run of the annual showcase in this second edition.

October 18–21, 2017
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www.aes.org/events/143

The 143rd International Audio Engineering Society (AES) Convention in New York will be, for the first time, co-located with the NAB Show New York 2017. The AES also confirmed that the association's annual US international convention will stay in New York again in 2018 This year's 143rd International AES Convention offers four days of intense audio research presentations, workshop and tutorials, and three days of the largest dedicated pro audio gear exhibition of the year. The series of in-depth seminars will address methods of bringing products to market faster, new headphone design techniques, development methods and better audio coding methods, and more.



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