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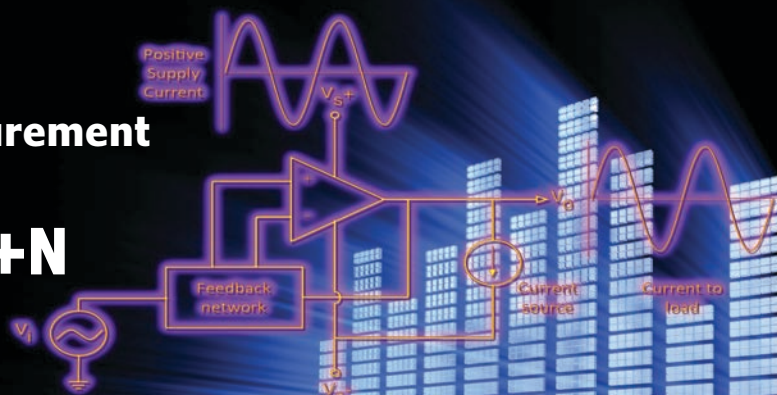
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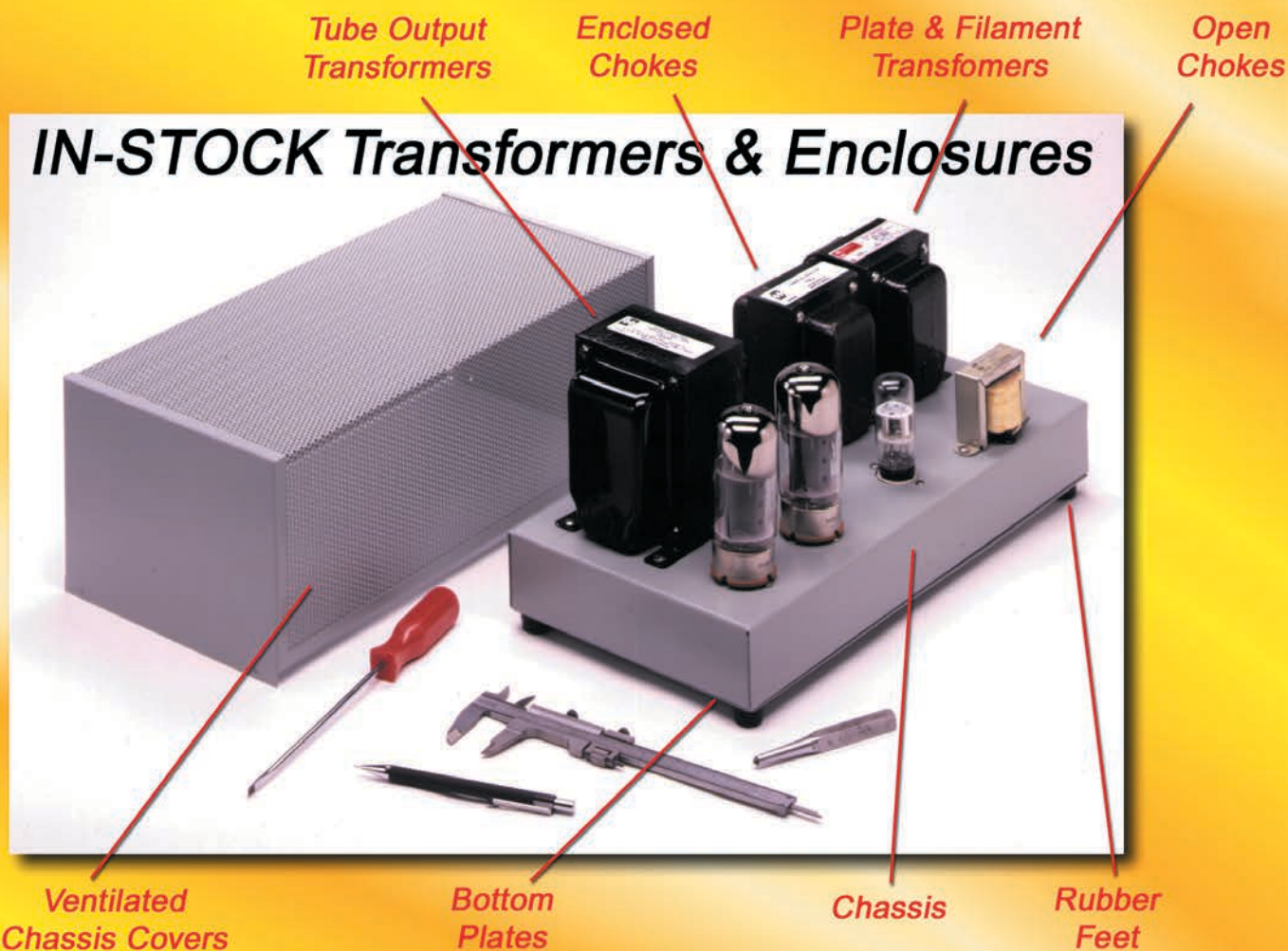
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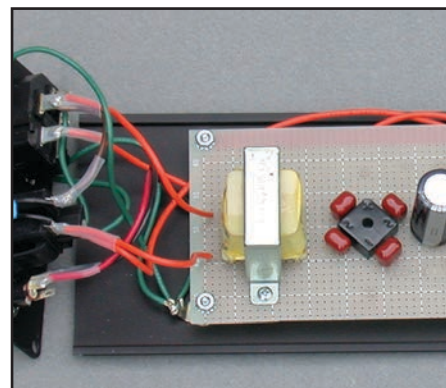
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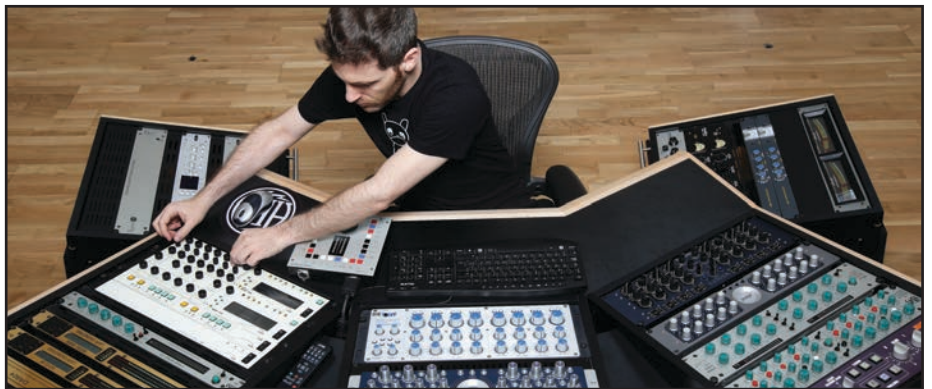
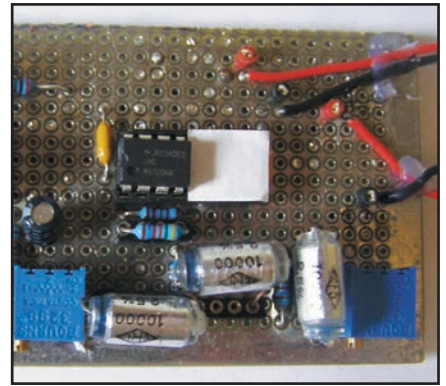
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Expanding horizons. Expanding a common passion. Welcome to a new *audioXpress*.

Having followed the audio market and visited the world's major trade shows for the last 20 years or more, I gained a broad perspective about how exciting and innovative the audio industry is. In particular, I recall the enlightening perspective you can receive from any Audio Engineering Society (AES) convention. The convention provides a place where industry veterans can share their experiences in engineering and communications. We chose to unveil the redesign of *audioXpress* at the 135th AES Convention.

In the early 1990s, I was fortunate enough to be responsible for a licensed electronics magazine. I quickly learned that the audience of such practical and project-oriented publications was a combination of students, enthusiasts, and industry professionals. They all share a passion for that field, are involved in many different areas, and use their spare time to pursue electronics-related hobbies—the most popular of which is audio electronics.

Since then, I have started several publications addressing the informational needs of professionals in the broadcasting, professional audio, and installation/systems integration markets. I also learned how the evolution of technology from analog to digital and the convergence with IT platforms and IP infrastructure was changing the market landscape at an exponential pace.

During this time, Edward T. Dell, Jr. (1923–2013) was devoting his life to people with a passion for audio electronics and creating magazines including *Audio Amateur* (rebranded as *Audio Electronics* in 1996), *Glass Audio*, *Speaker Builder*, and later, in 2000, *audioXpress*. In 2011, Ed Dell sold his company to Elektor International Media (EIM) and retired.

Much in the same spirit of the original *Audio Amateur*—and with the support of a worldwide organization deeply involved in the electronics industry—we believe that *audioXpress* will blossom into a fascinating publication that follows the latest audio innovation trends, independent of the application field, and shares a common audience of engineers, consultants, and enthusiasts in the electronics and audio fields, most of whom are involved in R&D.

Although it was deeply rooted in the US, *audioXpress*—together with its sister publications *Voice Coil* and the *Loudspeaker Industry Sourcebook*—reached professionals around the world (e.g., Europe, China, India, and Brazil). It has gained more of a global presence since its acquisition by EIM, which also publishes some of the best technical books in the electronics industry.

I am really excited to bring the “new” *audioXpress* to a wider global audience, knowing that we can build on the tradition of the original publication and its diversified audience. We are working to create a magazine you will enjoy and anticipate reading every month.



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.

Gel-Based Speaker Demonstrates Ionic Conductor Capabilities

Researchers Jeong-Yun Sun and Christoph Keplinger at the Harvard School of Engineering and Applied Sciences (SEAS) have demonstrated a new type of loudspeaker transducer, which works on a different principle than the conventional moving coil design.

It is made of a thin sheet of transparent rubber sandwiched between two layers of salt-water gel. A modulated high-voltage signal passed across the two outer gel layers exerts pressure on the rubber membrane, causing it to vibrate and produce sound, much like a planar (or flat) speaker. According to the Harvard SEAS report, the prototype speaker has a 20-Hz-to-20-kHz frequency response.

The important thing is that the vibration effect is produced by the electrical charges carried by ions and not the movement of electrons. So strictly speaking, this is not an electronic device. And the speaker is clear as a window.

The high-voltage signal that runs across the surfaces and through the layers forces the transparent rubber to rapidly contract and vibrate, producing sound.

Published in *Science's* August 30, 2013, issue, this new speaker represents the first demonstration that electrical charges carried by ions, rather than electrons, can be put to meaningful use in fast-moving, high-voltage devices.

"Ionic conductors could replace certain electronic systems; they even offer several advantages," says co-lead author Sun, a postdoctoral fellow at Harvard SEAS.

For example, ionic conductors can be stretched to many times their normal area without an increase in resistivity—a problem common in stretchable electronic devices. Second, they can be transparent, making them well suited for optical applications. Third, the gels used as electrolytes are biocompatible, so it would be relatively easy to incorporate ionic devices (e.g., artificial muscles or skin) into biological systems or wearable devices.

According to Keplinger, who worked on the project as a postdoctoral fellow at Harvard SEAS and in the Department of Chemistry and Chemical Biology,

such systems have great potential as sensors and for actuating automatically on movement, which opens a range of application possibilities.

The audio speaker represents a robust proof of concept for ionic conductors because producing sounds across the entire audible spectrum requires both high voltage (to squeeze hard on the rubber layer) and high-speed actuation (to vibrate quickly)—two criteria that are important for applications but would have ruled out the use of ionic conductors in the past.

The electrical connection to the power source is established outside the device's active region where it does not need to be transparent. The traditional constraints are well known. High voltages can set off electrochemical reactions in ionic materials, producing gases and burning the materials. Ions are also much larger and heavier than electrons, so physically moving them through a circuit is typically slow.

The system invented at Harvard overcomes these problems. "It must seem counterintuitive to many people that ionic conductors could be used in a system that requires very fast actuation, like our speaker," says Sun. "Yet by exploiting the rubber layer as an insulator, we're able to control the voltage at the interfaces where the gel connects to the electrodes, so we don't have to worry about unwanted chemical reactions. The input signal is an alternating current (AC), and we use the rubber sheet as a capacitor, which blocks the flow of charge carriers through the circuit. As a result, we don't have to continuously move the ions in one direction, which would be slow. We simply redistribute them, which we can do thousands of times per second."

According to Keplinger, "Our system doesn't need a lot of power and you can integrate it anywhere you would need a soft, transparent layer that deforms in response to electrical stimuli (e.g., on the screen of a TV, laptop, or smartphone to generate sound or provide localized haptic feedback). People are even thinking about smart windows. You could potentially place this speaker on a window and achieve active noise cancellation with complete silence inside."

The Harvard team chose to make its audio speaker out of simple materials. The electrolyte is a polyacrylamide gel swollen with salt water, but the team emphasized that an entire class of ionically conductive materials is available for experimentation. Future work will focus on identifying the best combinations of materials for compatibility, long life, and adhesion between the layers.

Harvard School of Engineering and Applied Sciences
www.seas.harvard.edu



Jeong-Yun Sun (left) and Christoph Keplinger (right) demonstrate their transparent ionic speaker, which uses a signal conducted by ions rather than electrons to vibrate a rubber membrane. (Photo courtesy of Eliza Grinnell, SEAS Communications)

LSR 3 Series: Affordable Studio Monitors From JBL Professional

JBL Professional introduced a new professional studio monitoring range at an affordable price. The 3 Series studio monitors leverage the new technology developed for JBL's flagship M2 Master Reference Monitor, which was launched earlier this year. This is also the first studio monitor line to incorporate JBL's patent-pending Image Control Waveguide, with a distinctive appearance and rugged build.

The JBL 3 Series features two models: the LSR305, 5" powered studio monitor and the LSR308, 8" powered studio monitor. The LSR305 responds from 43 Hz to 24 kHz and a 108-dB peak sound pressure level (SPL). The LSR308 features a 37-Hz-to-24-kHz response and a 112-dB peak SPL.

Surrounding the high-frequency tweeter and located directly above the woofer, the new Image Control Waveguide precisely controls the sound emanating from the monitor in the vertical and horizontal planes. With a complex contour employing tiny, meticulously engineered ridges, the waveguide optimizes the loudspeakers' phase relationship, and the blend of directed and reflecting sound arriving at the listening position. In addition, the waveguide enables 3 Series models to deliver neutral sound across a large working space.

The 3 Series's long-throw woofer and damped woven

composite tweeter are designed to reproduce the powerful transients and micro-dynamics of any mix. JBL's patented Slip Stream low-frequency port design works with the woofer to produce deep bass response at all playback levels. The port's double-flared shape is precisely engineered for greater low-frequency extension and reduced turbulence. Efficient Class D amplifiers provide enough power to the drivers to deliver the output and dynamic headroom needed for the most demanding production styles.

Each 3 Series model incorporates a range of features to integrate the loudspeakers into a variety of environments including multiple inputs, a detented volume control, and an input sensitivity switch for compatibility with professional and consumer equipment.

The loudspeakers cost \$199.99 and \$325 for the LSR305 and LSR308, respectively.



The 3 Series studio monitors leverage the new technology developed for JBL's flagship M2 Master Reference Monitor.

JBL Professional

www.jblpro.com/3series

Avid S3L: A New AVB-Based Live Sound Mixing System

Following the Venue systems, which enabled Avid to get a strong foothold on the live sound market, the company has introduced a new compact modular system for that market built on audio video bridging (AVB) standards. Avid has always been part of the AVB efforts from its early beginnings and it is currently a promoter in the AVnu Alliance, an industry forum dedicated to the advancement of professional-quality audio/video by promoting the adoption of the Institute of Electrical and Electronics Engineers (IEEE) 802.1 Audio Video Bridging (AVB) and the related IEEE 1722 and IEEE 1733 standards over various networking link layers.

The new modular Avid S3L System leverages the best technology from Avid's successful recording technologies (Pro Tools) and combines it with the expertise gained in the industry with its Live solutions (e.g., the VENUE software that drives the system). The S3L System implements a modular system architecture comprising a high-performance HDX-powered processing engine running AAX plug-ins from its studio technology with the scalable remote I/O (up to 64 microphone inputs) that can work on stage or in a rack, and a clever compact control surface built for the road, which could be equally at home in the studio.

The system uses Dual Gigabit Ethernet ports with latching etherCON connectors, supporting both the Ethernet AVB and EUCON standards. Due to the accessible entry-level price (starting at \$17,995), this is an interesting solution that helps bring more awareness to the full potential of networked audio solutions based on recent standards.

The S3L can be configured with up to four Stage 16 remote boxes, while the S3 compact 16-fader control surface can be easily transported as carry-on luggage. Direct recording/playback to Pro Tools (or other DAW) is possible through a laptop Ethernet port while direct stereo recording/playback can be achieved with any standard USB flash drive. Show files are compatible with all Avid Venue live systems.

Avid S3L is now shipping.

A networked, modular digital mixing system for professional live sound and studio applications is realized through the tight integration of technologies, including EUCON for networked control, AVB for networked audio, HDX floating-point processing, AAX plug-ins, and VENUE and Pro Tools software applications.



Avid Technology, Inc.
www.avid.com



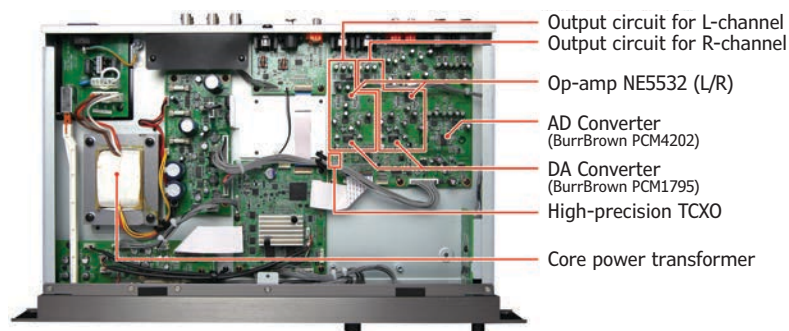
TASCAM's High-End Master Recorder and ADDA Converter

It is good to see new product launches from the TEAC group's pro audio brand after its recent acquisition by Gibson, especially this upgrade to the legendary DV-RA1000HD recorder. The new TASCAM DA-3000 offers the same famous Burr-Brown (now Texas Instruments) ADCs but it comes with a high-quality op-amp (NE5532), optimum condensers, and high-specification resistors for low-noise, high-accuracy, and high-heat capacity in a sleeker, more modern design. This new high-definition master recorder/ADDA converter is designed to fit in any professional or

home recording studio, for recording, mastering, broadcasting, replacing a DAT machine, or for audiophiles who want to upgrade their files. This recorder supports high sampling rates up to 192 kHz pulse code modulation (PCM) and 5.6 MHz direct-stream digital (DSD), with the option of recording to SDHC and compact flash with support for USB memory playback.

The high-precision TCXO fan-less design ensures pristine audio quality. The dual-monaural DACs help eliminate any interference. A balanced XLR I/O, unbalanced RCA I/O along with digital audio I/F AES-EBU, S/PDIF for PCM, and SDIF-3/DSD-raw for DSD is located in the rear of the unit. The DA-3000 warrants a clock frequency accuracy of 1 ppm by TCXO and uses a crystal direct system for low jitter.

The dual-monaural DAC is configured with Texas Instruments (TI) ICs (PCM1795) for each channel and uses TI's PCM4202 on the A/D conversion, adding an E-I core power transformer with separated coils for digital and analog circuits.



TEAC Corp.
www.tascam.com

Radial Engineering USB-Pro High-Resolution, 24-Bit, 96-kHz Stereo Direct Box

The USB-Pro (\$220 retail) is a high-resolution stereo direct box designed to convert sound files from a laptop computer and transfer them to a pair of balanced audio outputs to feed a PA, recording, or broadcast mixing console.

According to Radial President Peter Janis, "For years, Radial customers have been asking us to get into the digital world. We have hesitated due to lack of clear standards and challenges with respect to interfacing with computers. But with the recent advent of self-configuring USB ports, we feel the time is right to finally get involved and the USB-Pro is the first Radial product to sport digital connectivity."

Made to be Plug & Play, easy to use, the USB-Pro automatically configures itself for use with all popular operating systems including Mac OSX, Windows XP, Vista, and Windows 7, eliminating the need to load special drivers. And unlike devices that are limited with 16-bit, 44.1-kHz conversion rates, the USB-Pro elevates the performance with 24-bit, 96-kHz stereo converters to deliver more headroom and greater detail. This eliminates

the need for additional sound cards or separate converters when transferring files, further streamlining production in busy work environments.

Connection from the laptop is done via the pro-audio standard USB type-B port. D/A conversion is monitored with the built-in headphone amplifier to ensure the signal is properly downloaded and converted. A monosum switch may be engaged to check for phasing or facilitate signal distribution to two outputs should this be preferred. Just set the output volume control.

Should hum or buzz caused by ground loops be encountered, two set-and-forget side-access switches enable you to insert isolation transformers into the signal path to block stray DC voltage offsets. To further reduce susceptibility to noise, this is augmented with a ground lift switch that lifts pin 1 on the two XLRs.

The Radial USB-Pro is designed to handle the rigors of professional touring with protective zones around the switches, connectors, and controls to keep them out of harm's way. Inside construction ensures the internal PC board will not torque, which could cause premature part failure. Finally, a full-bottom no-slip pad provides mechanical isolation and electrical insulation.



The Radial USB-Pro, a high-performance stereo direct box, integrates a built-in headphone amplifier and balanced Lo-Z outputs with switchable isolation.

Radial Engineering, Ltd.
www.radialeng.com

THE POWER AMPLIFIER REDEFINED



CXD SERIES
FOR INSTALLED APPLICATIONS



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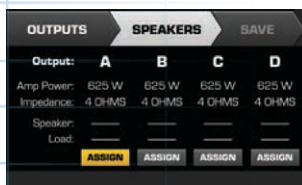
Introducing two new product lines that will forever change the way you think about power amplifiers. Powerful, flexible, highly efficient, and easy to operate, the PLD and CXD Series Processing Amplifiers offer an incredible combination of advanced technology, legendary QSC reliability, and of course, unparalleled sound quality. Visit us online at QSC.com for a complete product tour.

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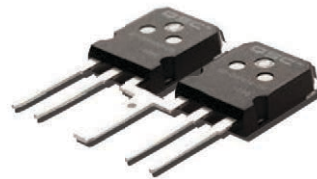
Flexible Amplifier Summing Technology™ (FAST) allows power to be distributed across almost any combination of 1, 2, 3 or 4 channels.



Intuitive and simple interface, including optional PC control and monitoring



Preset Wizard simplifies amplifier setup providing system design tools and loudspeaker selection from a list of loudspeakers.



Proprietary and revolutionary MOSFET output device delivers high voltage operation without needing a full bridge output.

QSC[™]
qsc.com

Audio Engineer Focuses on the “Masters”

By
Shannon Becker
(United States)

An Interview with Daniel Weiss

Daniel Weiss founded Weiss Engineering in 1985. The company designs and manufactures digital audio equipment for mastering studios.



SHANNON BECKER: Tell us a little about your background and where you live.

DANIEL WEISS: I live in Uster, a small city close to Zurich in Switzerland. In the 1970s and 1980s, I played music in a band, first as a violin player and later as the bass guitarist, which seemed preferable to the other band members. I also built various synthesizers, amplifiers, and speakers. I did a four-year apprenticeship as an electronics technician and during that time two friends and I formed a company called “White Amplifiers.” We built amplifiers and speakers

for musicians in our spare time. After the apprenticeship, I studied electronics engineering and eventually graduated with a BSEE.

SHANNON: In 1979, you joined Studer-Revox as an electronics engineer working in the digital audio lab. Can you share details regarding your work on the sampling frequency converter design?

DANIEL: In 1979, Willi Studer decided to enter the digital audio era and established the “PCM laboratory” with almost all the lab members being newly recruited engineers and technicians. We were kind of an isolated group as the other labs were slightly suspicious of digital audio technology. We also had a hard time (at least it seemed to me) defining digital audio products that would make sense in a mainly analog world.

There were several digital audio recorders around at the beginning of the 1980s (e.g., Sony, 3M, Soundstream, JVC, Mitsubishi, etc.). There wasn’t much standardization back then so the sampling rates and interface formats greatly varied. Thus, it made sense to create a universal sampling rate converter with custom wired interfaces. This became the SFC16, and I did most of the hardware design. It was a 6HU/19” unit with digital filters built in so-called distributed arithmetic. It is a very clever architecture that avoids the need for DSP or multiplier chips. For most of the units sold—I think 30 of them were manufactured—I also did custom interfaces.

SHANNON: What other types of audio products did you design? Can you share some of the challenges involved with the design(s)?

DANIEL: My colleagues at the PCM lab pursued various other projects, such as A/D and D/A design, analog reconstruction filter design (I also was involved), research in de-noising, and a preview unit for the delay required in vinyl cutting. This resulted in a

One of the largest setups of a 102 Series system was used at Sony Music in New York in the form of the IBIS digital mixing console.



A/D and D/A 6HU box, with enough memory to do the delay. It was not a simple task back then.

As Studer was mainly a tape recorder company, the design of a digital tape recorder was inevitable. The first model was an eight-channel unit using the newly established Digital Audio Stationary Head (DASH) format, which enabled you to interchange tapes with ones recorded on other DASH recorders. I did the audio processing unit for that eight-channel recorder, which was required for interpolation in case the data read from the tape could not be reconstructed via the error correction scheme employed.

Those were interesting times at Studer, as we were pioneers in the pulse code modulation (PCM) audio field. We did many side projects, such as a digital sine generator for measuring purposes (Audio Precision did not exist back then) or a study on TIM measurements with a new approach or a PWM-based analog track on the digital tape and so forth.

SHANNON: In 1985 you founded your own company, Weiss Engineering (www.weiss.ch). Initially, your company focused solely on designing and manufacturing digital audio equipment for mastering studios. How and why did you select that specific market niche?

DANIEL: One day in 1984, when I still was at Studer, a customer came to our lab and asked for an interface between a Sony F1 portable digital audio recorder and a Sony 1610 digital audio recorder. The F1 did not have any digital I/O, so it had to be a custom made interface box. Studer does not do such custom work, so I made that interface for the customer in my spare time. The customer was Ben Bernfeld, a recording and mastering engineer from Harmonia Mundi Acustica in Germany. He knew exactly what was required in terms of equipment for CD mastering (or pre-mastering to be exact). So we decided to build a modular digital audio system to interface and process digital audio. I did the design and manufacturing while he organized the sales. CD pre-mastering was popular in the US mainly, so we concentrated on that market.

SHANNON: Tell us about Weiss's first product. Is it still being sold today?

DANIEL: The first system became the Harmonia Mundi Acustica BW-102 unit, starting with modules for F1, 1610 interfacing, a digital high-pass filter for DC offset elimination, a digital de-emphasis and a digital level control module.

Over the years, dozens of modules were added.



The potential of a Weiss Engineering Mastering Studio "Mastering Mansion Madrid" uses Weiss Gambit Series equipment, which are the white faceplate units on the left.

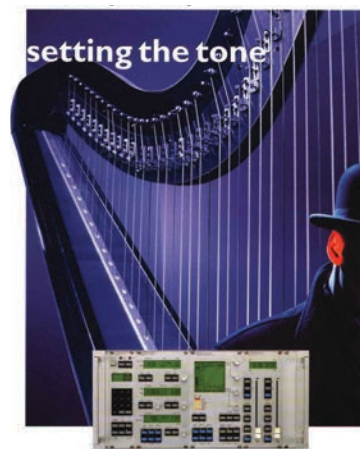
We even did digital mixing consoles based on the BW-102. The largest one was a 32-channel console with four auxiliary buses. Another one was a 24-channel configuration with GML fader automation used by Sony Classical in New York. Those consoles were a bit awkward in terms of hardware requirements, because the BW-102 initially was designed for two-channel applications. Later, we also upgraded most of the modules to handle 96 kHz. Quite a few customers still use the BW-102, we even occasionally sell modules. Technically it is still up-to-date with 96/24 capability and 32-bit floating point processing. After the BW-102, we started the Gambit Series with 19" units (e.g., analog to digital, digital to analog, parametric equalizer, dynamics processor, de-noiser/de-clicker, sampling rate converter, and more).

SHANNON: In 2000, Weiss entered the high-end consumer audio market with a new product line. What was the impetus behind that decision?

DANIEL: We thought that our DAC1 DAC could find a market within the high-end community. So we built the Medea DAC, based on the DAC1, to test the waters. The Medea became a huge success and it did not take long for customers to ask for more. So we built the Jason CD transport to complement the Medea. Other high-end products followed, up to the latest one, the MAN301 network player.

SHANNON: With the two separate aspects of your company—professional equipment for mastering studios and high-end consumer products—you are in the unique position of controlling, in part, the "production" of the masters and their reproductions. Do you think there is a direct correlation between the two "worlds?"

DANIEL: Correlation maybe in that both mastering engineers and audiophiles are interested in getting



The Weiss 102 Series consists of digital audio processing modules suited for CD mastering, mixing, and digital audio signal processing. You can configure a system according to your requirements.



The Weiss-designed MAN301 network player's front boasts a sleek design (a). It is a versatile unit that uses the Gracenote database (b).

topnotch sonic quality and ergonomics. We can use our design philosophy—with the utmost transparency—with both markets. But in the end, we simply supply tools. The mastering engineer needs to know how to use them properly.

SHANNON: To what do you attribute your company's continuing success?

DANIEL: At first, it was the fact that we built the right product at the right time (i.e., when the CD took off there was a huge demand for decent audio processing in the digital domain). In the consumer market, I think our customers like our “no bull” approach. I don't hold back with my opinions about \$1,000 mains cords, gold-plated fuses, small wood blocks for acoustics treatment, or CD demagnetizing, and so forth. I wrote some white papers firmly based on the laws of physics on various audio topics in an attempt to fight the snake oil with facts. This is something I like about the pro audio people, they are down-to-earth guys.

SHANNON: Tell us about your favorite high-end consumer product? What makes it different from other products on the market today?

DANIEL: One of my favorites is the MAN301 network player—from our product line, of course. It is an incredibly versatile unit for CD playback and ripping, and metadata tagging/artwork. It uses the Gracenote database and this is hardly seen on any other high-end network player. It also includes file playback (including DSD), DAC, preamplifier functions, and so forth. I use one at home and enjoy it every day. We continue to develop additional software for

the MAN301 (e.g., for room equalization, creative equalization, vinyl simulation, and so on).

I also like to listen to as many different speakers as possible to explore the various philosophies and designs. I think the speaker/room system has, by far, the greatest potential for improvement of the whole audio chain. Audiophiles should acknowledge that and stop messing around with mains cords. The industry still has a long way to go when it comes to speaker/room optimization.

SHANNON: Could you share your opinion on mastering for digital file distribution and, in particular, the mastering for iTunes initiative?

DANIEL: If it is mastering for an uncompressed format, then the procedure should not be different from a standard CD mastering—except maybe if the format is at a higher sampling rate and/or word length than for a CD.

Mastering for iTunes is different, as it means mastering for a lossy format (for the time being at least). But I think the best thing about that initiative is Apple imposes specific criteria on the technical quality of the supplied music, in particular that the music must not be clipped. There are also a number of recommendations available at http://images.apple.com/itunes/mastered-for-itunes/docs/mastered_for_itunes.pdf.

SHANNON: Where do you see the audio market headed in the next five years? Do you think we will eventually evolve to “high-end” streaming audio services rather than downloading files?

DANIEL: There always will be both variants. Many people like to “own” the music so they can play it anytime and anywhere. And, I think the emotional relationship to the music is different if you've got it “on file” and not just via a stream.

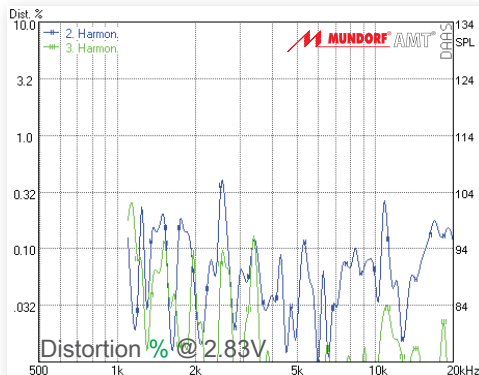
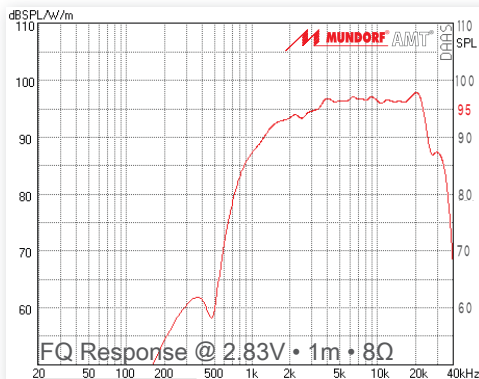
Streaming services are great to check out new music. They should have a “buy” button on their websites though. Streaming during travel can get expensive and/or can be annoying when the stream gets disrupted in the tunnel or because of too many people try to get streams on a train, for instance. Also it seems that for artists streaming services are far from lucrative. That could be changed maybe if they would simplify the buying process right from the streaming site. In any case, the majority of high-end playback systems will use computer-based playback devices because it is so much more convenient and easily enables people to discover new music from streaming services or even in their own libraries. 

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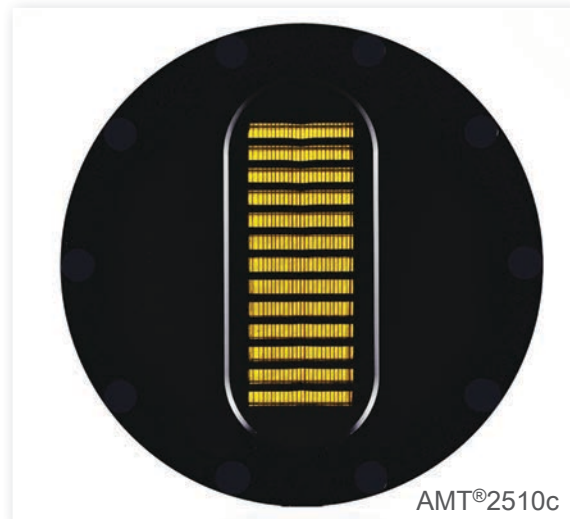
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HDBaseT Alliance Releases 2.0 Specification, Introduces HomePlay

By
João Martins

(Editor-in-Chief)

There are several competing technology proposals for using basic LAN/Ethernet infrastructure to distribute AV signals, some more proprietary than others. Some of the proposals address specific industry needs. Others are based on the Institute of Electrical and Electronics Engineers (IEEE) standards and represent an effort to expand the spectrum of applications. Some technologies are simple connectivity solutions, while others address signal/data distribution. Recently, we have seen a clear convergence targeting both areas, including power distribution on a simple CAT5e/6 cable and using standard network connectors.



The HDBaseT technology has captured the interest of audio and AV industry professionals and consumer electronics (CE) aficionados. HDBaseT technology was originally implemented on silicon by Valens Semiconductor. It has become a connectivity standard for whole-home and commercial distribution of uncompressed full HD digital video, audio, 100BaseT Ethernet, power over cable, and various control signals through a single LAN cable (e.g., the 5Play concept).

Recently, the technology gained traction with the worldwide adoption of HDBaseT in hundreds of products from more than 65 major companies and the publication of the HDBaseT 2.0 specification, which introduced the HomePlay concept for home entertainment.

What Is It?

The technology has been promoted since 2010, when LG Electronics, Samsung Electronics, Sony

Pictures Entertainment, and Valens founded the HDBaseT Alliance.

The HDBaseT 5Play feature set enables the delivery of full HD/3-D and 2K/4K uncompressed video to a network of devices or to a single device (point-to-point) over a single 100-m/328' CAT5e/6 cable. Since the video is passed through the HDMI chipset, HDBaseT supports all key HDMI 1.4 features, including electronic program guides (EPGs), consumer electronics control (CEC), extended display identification data (EDID), and high-bandwidth digital content protection (HDCP). The unique video-coding scheme ensures the highest video quality at zero latency.

HDBaseT audio is also passed through directly from the HDMI chipset. HDBaseT audio supports all standard formats, including Dolby Digital, DTS, Dolby TrueHD, and DTS HD-Master Audio.

HDBaseT also supports 100-Mb Ethernet, enabling communication between televisions, sound systems, computers, and other CE devices as well as access to any stored multimedia content (e.g., music or video streaming). HDBaseT also supports Ethernet Fallback Mode, which enables seamless integration between an HDBaseT device and an Ethernet-only infrastructure. In such a case, only the Ethernet capabilities of the connection are enabled.

More importantly, HDBaseT enables multiple control signals for multiple purposes: Consumer Electronic Controls (CEC), RS-232, USB, and infrared (IR). IP control is also enabled through the support of an Ethernet channel. HDBaseT's range of control

options provides manufacturers with virtually endless opportunities.

Up to 100 W of DC power can also be delivered over the same CAT5e/6 cable. This major aspect of 5Play provides the freedom to place equipment where you want it and not where an available power source happens to be installed.

Evolution of the HDBaseT Architecture

HDBaseT uses an asymmetric method, sending video, audio, Ethernet, and controls from source to sink, but only 100 Mb are transferred back (Ethernet and controls). Unlike conventional data communication, which is a symmetrical application by nature and requires a bit error rate (BER) of at least 10^{-12} , HDBaseT's asymmetric nature is based on an innovative DSP engine and application front end (AFE) architecture.

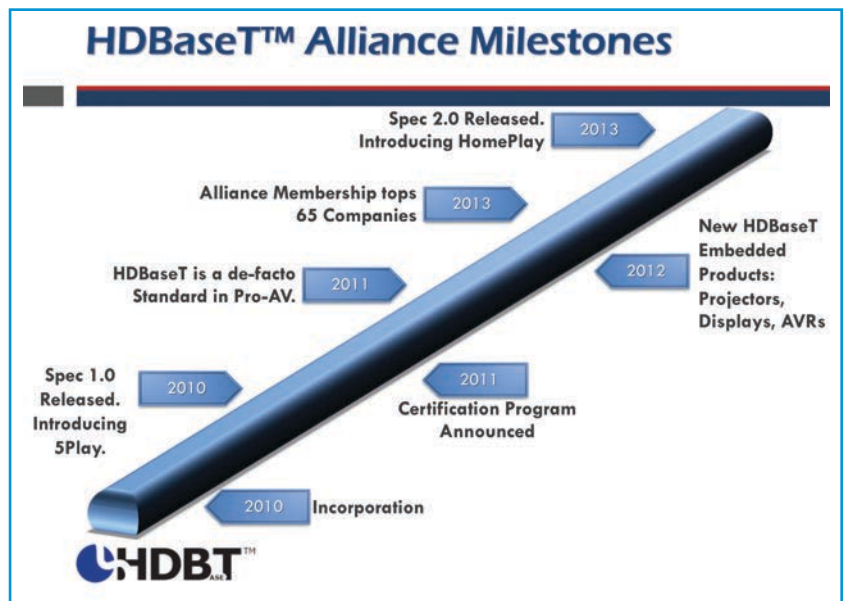
A special line-coding scheme was developed to provide a better transfer quality to some kinds of data (e.g., audio, controls, and Ethernet) without the need to "pay" the protecting overhead for the video content that consumes most of the bandwidth.

HDBaseT uses a proprietary version of pulse-amplitude modulation (PAM) technology, where digital data is represented as a coding scheme using different DC voltage levels at high rates. This enables the 5Play feature set to be maintained over a single 100-m CAT cable, without the wire's electrical characteristics affecting performance.

Although HDBaseT uses the same coding technology as Ethernet and even has an Ethernet channel, its packet-based technology is different from the traditional Ethernet packets. HDBaseT simply relies on the same physical cabling that provides the inherent benefits of the cost-effective CAT infrastructure.

In August 2013, the HDBaseT 2.0 specification became available to professional AV and consumer electronics products manufacturers. The change makes it simpler and more cost-effective for consumers to turn their entire homes into multimedia entertainment centers.

According to Micha Risling, HDBaseT Alliance's marketing chairman, "HDBaseT has become the digital connectivity standard for professional A/V installations. We have leveraged our experience working with the pros to ensure that HDBaseT 2.0 will enable consumers to enjoy the same benefits and user experience: highest video quality, reliability, and multi-room support all over a single inexpensive cable. "HomePlay, as defined in the new HDBaseT 2.0 specification, features an elegant, affordable plug-and-play solution that will allow the average consumer to enjoy the same multimedia benefits of HDBaseT that the professionals have become accustomed to."



HDBaseT 1.0 was defined to support the transmission of the 5Play feature set (e.g., video, audio, Ethernet, power, and controls). Specification 2.0 sets the specification that enables HDBaseT-HomePlay switches to control and distribute multimedia across consumers' homes.

Connected Home Entertainment

"The HDBaseT 2.0 specification defines the required adaptations across all seven layers of the open system interconnection (OSI) model to support time-sensitive, high-throughput video/audio and other time sensitive applications," says Eyran Lyda, HDBaseT Technical Committee chairperson. "The HomePlay switch is designed to provide all required services for the multimedia home, such as delivering time-sensitive high-throughput video, including 4K, audio, and USB to all the rooms in the house."

While HDBaseT 1.0 only addresses the physical and data link layers, HDBaseT 2.0 adds networking, switching, and control point capabilities. HDBaseT 2.0 also defines point-to-multipoint connectivity, providing multi-stream support. HDBaseT 1.0 defined a point-to-point connectivity standard.

Specification 2.0 also adds inherent support for USB 2.0, and other native interfaces, which enable manufacturers to remove multiple interfaces and conversion components required for HDBaseT 1.0 products. The result is a smaller and a simpler to implement switch, which means vendors can produce HDBaseT switches with a consumer-friendly price tag.

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A HomePlay Switch that complies with HDBaseT 2.0 specifications will provide all required services for a multimedia home. (Photo courtesy of HDBaseT Alliance)



more cost effective for consumers and professionals is making the HDBaseT-enabled components more affordable, and the new features HDBaseT 2.0 offers will help manufacturers do just that," adds Risling.

In June 2013, the HDBaseT Alliance announced it had doubled in size in less than a year, with more than 60 member companies. During the Infocomm 2013 show, the HDBaseT Alliance exhibited dozens of new HDBaseT-enabled components across new categories, including new projectors from Hitachi, Epson, and NEC; displays from Panasonic; and AV receivers from Pioneer and Onkyo. For commercial installations, Atlona, Crestron, Emcore, Geffen, Kramer, and many others presented new interfaces and distribution solutions.

For more information about HDBaseT 2.0, visit www.hdbaset.org.

Integra's A/V Processor and Receiver with HDBaseT

The new HDBaseT standard, promoted by the HDBaseT Alliance, is gaining ground. In fact, Integra, a leading brand of A/V components specifically engineered for custom installation, has introduced a new mid-range THX Certified A/V pre-amp processor and A/V receiver with HDBaseT support. The HDBaseT port can be assigned for use in either the main zone or a second zone, while the HDMI output is used exclusively for the main zone.

The new DHC-60.5, 7.2-channel network A/V preamp processor is THX Ultra2 Plus-Certified. It also includes built-in Wi-Fi and Bluetooth and full support for the new 4K Ultra HD video displays, using either the 4K pass-through feature or up-scaling for conventional SD and HD video sources via a Marvel QDEO processor. There are connections for nine HDMI inputs and one output, plus HDBaseT, along with 7.2 multichannel balanced XLR pre-outs (with front-channel bi-amp capability) and two-channel balanced XLR audio inputs.

Meanwhile, the DTR-60.5 is a 9.2-channel, THX Select2 Plus network A/V receiver that boasts 135 W per channel of power, with 11.2 multichannel pre-outs to support full DTS Neo:X up-mixing for surround-sound playback. If preferred, the unit can be configured to assign four of the nine available channels to simultaneously drive audio in Zone 2 and Zone 3.

There is a full package of surround processing including Dolby Pro-Logic IIz and Audyssey DSX, plus Audyssey MultEQ XT32 room set-up processing.

Both models also support the playback of high-definition music files in FLAC, DSD, ALAC, HD 24/96, and HD 24/192 formats from HD music download services.

In addition to having the power to steer high-resolution movie soundtracks at theater-reference volumes, the receiver is loaded with high-end features to optimize two-channel performance. Pre-processing incorporates three 192-kHz/24-bit Texas Instruments (TI) DACs (with one stereo DAC per front channel in Differential DAC mode). Audiophiles who want to bi-amp their front speakers can find performance increases from the proprietary Digital Processing Crossover Network, which splits the audio signal precisely into separate high- and low-frequency signals, feeding them to the appropriate speaker drivers.

Both the Integra DHC-60.5 A/V preamp processor and the DTR-60.5 A/V receiver are available with suggested retail prices of \$2,000 and \$2,300, respectively.

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Designing for Ultra-Low THD+N (Part 1)

By
Bruce Hofer
(United States)

Understanding Measurable Noise and Distortion

The design of ultra-low-noise, low-distortion circuits is more than simply a matter of a well-considered schematic. Because of their impact on measurable noise and distortion, layout and component quality are often more important.

Let me clarify up front what may be a misconception. This article is not about what makes the “best” audio amplifier nor do I make any claims about audible quality. I shall leave those subjective matters to others, and instead focus on what we at Audio Precision have been doing for many years: understanding what is in fact measurable and what mechanisms contribute to the imperfections we find.

Few audio products qualify as “Ultra-Low THD+N” and for most purposes this is perfectly acceptable. But for those who work in this rarefied environment, only a select set of tools can reveal behaviors that separate the merely good from the truly remarkable and a deep understanding of circuit behaviors is required to address and improve the designs produced. This means moving beyond the approximate representation of Maxwell’s equations encapsulated in a schematic and focusing on the actual circuit’s physical implementation.

Is Circuit A Better Than Circuit B?

After many years in the audio business, you observe trends and ideas that become popular among designers for certain time periods. These trends often take the form of circuit types, some of which claim to solve specific problems or to somehow “sound better” than others. While some designs are genuinely clever and may address certain issues (e.g., circuit protection) most do not become canonically accepted as “the best” for audio because actual implementations reveal a range of behaviors from product to product. The magic bullet is only magic in a suitable context. It is otherwise unremarkable. Circuit A is only better than circuit B when circuit A is realized with the appropriate attention to detail, and roles may easily be reversed with decisions around bias, compensation, component quality, layout, and so forth.

Lenz’s Law’s Unintended Consequences

We all recall learning the basic physics of electricity in high school and college. We know from the earliest lessons that a conductor’s current flow generates a magnetic field, and conversely that magnetic fields may induce potentials in nearby conductive loops. While we use this explicitly in the design of transformers, motors, and loudspeakers, it is a fact often obscured by the approximation of electromagnetic systems represented in electronic schematics. When exploring extremely low levels of THD+N, these approximations become very important to understand.

Consider an op-amp. This common amplifier employs a Class-AB output, and when configured with feedback can deliver a very linear current to a

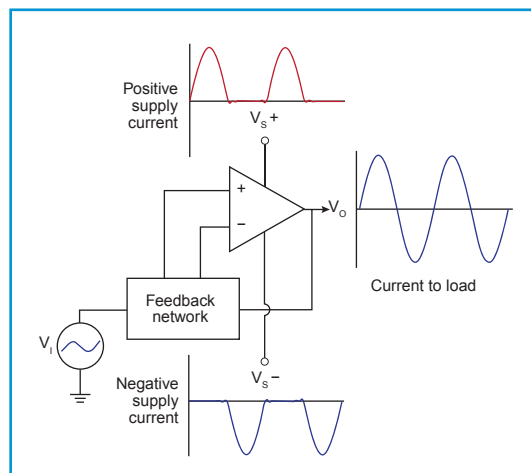


Figure 1: The op-amp supply currents look fine in the schematic; however, Lenz’s law indicates there may be a problem.

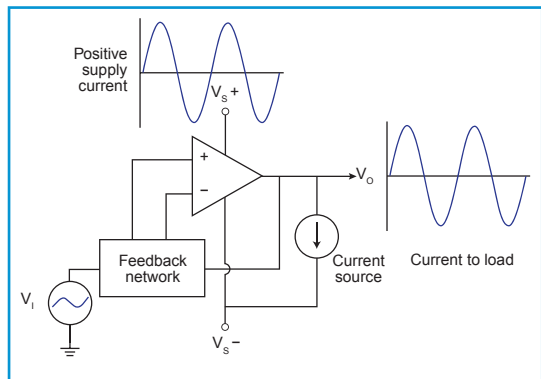


Figure 2: The op-amp is forced to Class A and only one rail is used.

may even be the dominant distortion product in otherwise very high-quality amplifiers.

Does this mean that this problem is immediately audible? No, but in the world of extremely low THD+N, we must consider sources of nonlinearity that others ignore. We must look very carefully at the schematics and the physical layout to understand this.

Planes and Traces

When is more ground not such a good thing? The answer is: when you don't know where the currents are.

Some designers favor the use of a solid "power supply plane" as a PCB layer. As such, a plane certainly appears to be an effective way to carry a large amount of current and hence will reduce problems of interference from the perspective of relative voltages. There is very little measurable resistance and thus few voltage variations at different points on the plane.

But the currents, visible or not, must follow a specific set of paths from point-to-point along this plane. The flow will not be evenly distributed but will follow, literally, the path of least resistance, however small the difference. Lenz's law still applies, which means that noisy currents flowing across the plane may still induce signals into other areas of the circuit, only now we have no means to control these effects by moving traces. For this reason, well-designed traces are preferable to supply planes in ultra-low THD+N designs, as they give the engineer control over precisely where current flows and hence where noise and distortion can be injected.

Isn't This an RF Problem?

While the previously described phenomena are more pronounced as $f \rightarrow \infty$, the higher currents involved in power amplifier design mean that these effects can be easily

load. Since all current is being provided by the supply rails, you may assume that the supply current is likewise very linear with respect to the input signal. This would be incorrect and has measurable consequences.

Since the op-amp's output is Class-AB, the output transistors are transitioning from OFF to ON states in a somewhat complementary fashion, responding to changes in the input signal. This means that each supply rail ($-V_s$ and V_s) delivers a current waveform that contains a rectified version of the total output current superimposed on the DC value (see Figure 1).

Why Is This a Problem?

Here is where Lenz comes back to us with mutual inductance. In the real world, Lenz's law tells us that currents generate magnetic fields around conductors that in turn generate potentials in adjacent conductive loops. If these currents contain distorted representations of the signal, then this distortion may well "bleed" into other stages (e.g., the feedback path) and become measurable as a function of frequency. This has nothing to do with the amplifier's schematic, per se.

At higher currents, this can seriously affect measurements. As a result, power amplifiers often demonstrate relatively high levels of this form of distortion at higher frequencies. This occurs regardless of quality of components or circuit design on paper. It is a result of high currents injecting nonlinear signals via mutual inductance to other loops within the amplifier and is more a function of physical layout than schematic. This behavior

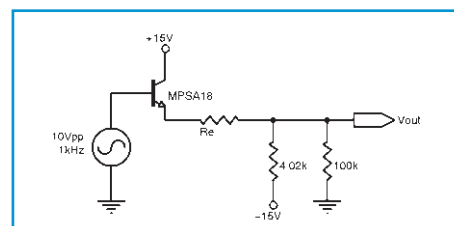


Figure 3: A simple emitter follower was constructed using a MPSA18 transistor.

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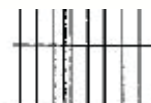
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About the Author

Bruce Hofer is a co-founder of Audio Precision and a widely respected analog design expert with a career spanning more than 40 years. His designs rank among the quietest and lowest distortion audio measurement circuits in the world.

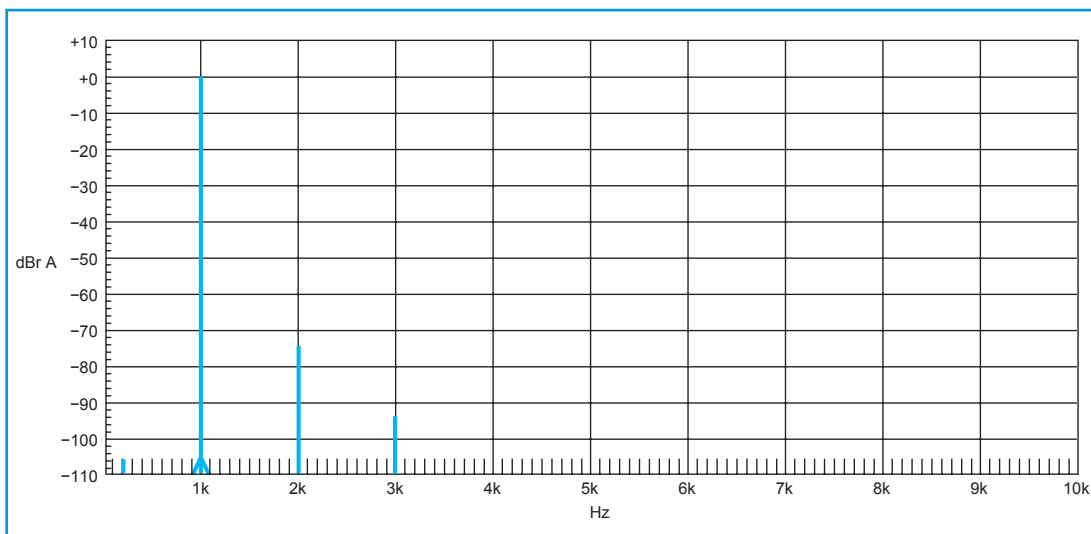


Figure 4: Emitter follower distortion measurement shows $R_E = 4.02\text{ k}$, $V_S = \pm 15\text{ V}$, and $V_{\text{signal}} = \pm 5\text{ V}_{\text{PEAK}}$ at 1 kHz.

measured at frequencies as low as 5 kHz. It is precisely that attention to detail that separates the good from the great at the extremes of performance.

Some Partial Solutions with Op-Amps

Op-amps are at the heart of much analog audio. Some useful techniques can help reduce the problems mentioned here.

Decoupling

Ensure that power supplies are differentially decoupled as physically close to the op-amp as possible to minimize loop area. The nonlinear AC supply currents will be shunted to each other, thus reducing unwanted induced potentials.

Forcing Class-A Operation

Another partial solution to rectified supply currents is to force op-amp output stages to run (nearly) Class A by loading the output to a rail, forcing only one rail to be used all the time (see **Figure 2**). Currents will be more linear but, of course, the op-amp now is expending more energy and has reduced output capacity.

Estimating Semiconductor Distortion Contributions

While the use of op-amps with abundant feedback negates much

of the need to worry about the nonlinearity of discrete transistors, there are still many cases in which discrete designs are desirable. These nonlinearities are significant and can be well estimated before prototyping.

A Taylor series approximation works well to estimate distortion from certain forms of nonlinearity (e.g., the behavior of semiconductor junctions. However, it does not generalize to all forms of nonlinearity (some nonlinearity forms will be discussed in our next article).

Use a Taylor Series to Model Nonlinear Behavior in Semiconductor Junctions

The circuit is modeled as having a voltage dependent gain:

$$\frac{V_{\text{OUT}}}{V_S} = f(V_S) = A_0 (1 + V_S k_2 + V_S^2 k_3 + \dots)$$

The second (2HD) and third (3HD) harmonic ratios can be estimated with surprising accuracy using only three values for dynamic gain at the positive peak (A_P), negative peak (A_N) and, zero (A_0) points of an assumed sine-wave signal:

$$2\text{HD} \approx \left(\frac{k_2}{2} \right) = \frac{|A_P - A_N|}{8A_0}$$

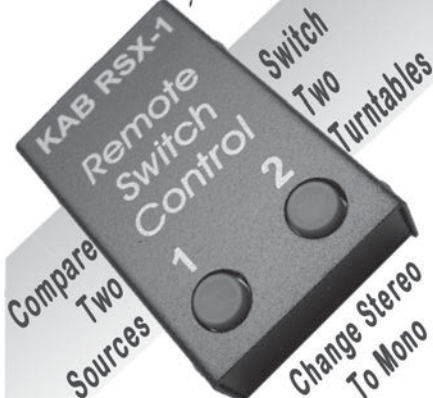
$$3\text{HD} \approx \left(\frac{k_3}{4} \right) = \frac{|A_P - A_N - 2A_0|}{24A_0}$$

Note that 2HD (as a ratio) is proportional to V_S , while

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3HD is proportional to V_S^2 . Since these first harmonics will dominate overall distortion, they are sufficient for estimation purposes.

Emitter Follower Distortion

We constructed a simple emitter follower using a MPSA18 transistor with $R_L = 4.02\text{ k}$ wired to -15 V and the collector to 15 V (see **Figure 3**). Performance with a $\pm 5\text{ V}_{PK}$ (10 V_{PP}) signal is to be determined at 21.8°C ($\approx 295\text{ K}$).

- The $100\text{-k}\Omega$ load resistor represents the input impedance of the audio analyzer (DC coupled)
- The element " R_E " models the dynamic impedance of the emitter-base junction. " R_E " interacts with the total load impedance to provide a voltage gain that is slightly less than unity and that varies as a function to the instantaneous signal voltage.

Dynamic emitter impedance, $R_E \approx kT/qI_E$

" k " = Boltzmann's constant = 1.38065×10^{-23}

" T " = 295 K (ignoring self heating within the transistor)

" q " = electron charge = 1.6022×10^{-19}

Gain is calculated at $V_B = -5, 0$, and 5 V :

$$V_B = -5\text{ V}: I_E = 2.282\text{ mA}, R_E = 11.138\ \Omega \therefore A_N = 0.099713$$

$$V_B = 0\text{ V}: I_E = 3.576\text{ mA}, R_E = 7.1085\ \Omega \therefore A_O = 0.099816$$

$$V_B = 5\text{ V}: I_E = 4.870\text{ mA}, R_E = 5.22\ \Omega \therefore A_P = 0.099865$$

Hence:

$$2\text{HD} \approx \left(\frac{k_2}{2}\right) = \frac{|A_P - A_N|}{8A_O} = 0.019\% \text{ } (-74.4\text{ dB})$$

$$3\text{HD} \approx \left(\frac{k_3}{4}\right) = \frac{|A_P - A_N - 2A_O|}{24A_O} = 0.0023\% \text{ } (-92.8\text{ dB})$$

The Envelope, Please

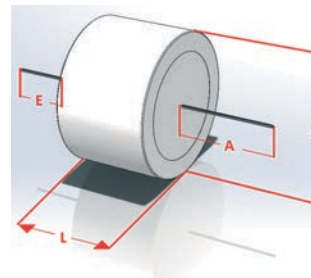
How well does this technique work? The measured levels of 2HD and 3HD with a 10-V_{PP} sine wave at 1 kHz are -75.1 dB and -93.4 dB compared to the estimates of -74.4 dB and -92.8 dB , for a difference of 0.7 dB and 0.6 dB , respectively (see **Figure 4**).

This estimation method can be used to analyze distortion contributions for any semiconductor junction in a design and can be used to help identify separate contributions from other sources, such as those discussed in the earlier sections.

Next Up

In my next article, I will discuss the importance of resistors and capacitors in ultra-low THD+N designs. The conclusions may surprise you. 

EVOLUTION through INNOVATION



The main feature of the EVO winding technology comes with its unusually narrow but high capacitor reel. This geometry results in two, acoustically clearly perceptible benefits: A much shorter

and utmost little loss signal path between huge contact areas is granted, thus an extremely low equivalent series resistance/ESR. Secondly, in order to meet capacitance the number of paralleled windings is larger than with regular caps, thus an effectively minimized equivalent series inductivity/ESL.

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QSC's FAST Platform

By
Stuart Yaniger

(United States)

QSC Audio Products in Costa Mesa, CA, is a global professional sound company with a reputation for making solid, reliable gear. QSC rarely leads the market, but it always seems to fill customer needs in a cost-effective and efficient way. QSC is successful because it does things just a bit better than the first players out there.

This has resulted in consistent growth and market acceptance over the years, with QSC remaining a solid competitor even against companies with significantly more resources. A survey of review sites shows extremely high customer satisfaction, with almost no negative comments and many glowing endorsements.



Photo 1: Housed in a 2U rack-mount configuration, CXD and PLD Series amplifiers feature new QSC proprietary Class-D power devices, which enable multichannel, high-performance power. Flexible Amplifier Summing Technology (FAST) provides better power allocation by actively distributing the total amplifier power across one, two, three, or all four outputs enabling amplifier channels to be combined for maximum current or voltage output.

QSC's amplification product platform is based on an approach invented by founder Pat Quilter, who has been associated with reliable and well-accepted analog amplifiers for many decades. When designing his Class-D amplifier line, Quilter used conventional circuitry and added a few twists to it.

In Quilter's circuits, an amplified version of the input signal is compared to a constant frequency triangle wave. The triangle wave is generated by a simple RC integration of the clock, which has a typical frequency of about 400 kHz. The comparator's output then drives the gates, which control the output MOSFETs's switching. This is all conventional but done with a particularly simple triangle wave generator.

Normally, the amplifier's distortion performance depends on the accuracy of the sawtooth—its frequency and linearity. Quilter's amps can use the simple triangle circuit and still realize low distortion because of two feedback loops, a traditional global loop from the output to the first stage's inverting input, and a second loop that applies a current proportional to the switched (i.e., pre-output-filter) output voltage to the summing point of the clock voltage and the amplified analog input voltage at the comparator input. By choosing the right ratio of the three source resistances (and hence currents, with clock signal current being larger than current

from the local feedback resistor), the PWM output is linearized, locking the switching frequency in synchrony with the clock. Quilter claims that this circuit trick reduces distortion of the unfiltered PWM signal by an order of magnitude, linearizing everything up to the output filter. His patent shows half- and full-bridge implementations.

A Class-D amplifier's output is also modulated by the power supply rail for the output stage. Quilter's patent addresses that issue as well, using a novel combination of an output-controlled regulator and rectification via an envelope detector. The combination of a tightly regulated low-noise supply and the multiple feedback loops is supposed to reduce overall amplifier distortion levels to figures comparable to analog amplifiers, but with greater simplicity and efficiency than other Class-D approaches. None of this is revolutionary, it's evolutionary.

Power Trios

A perfect illustration of QSC's "don't worry about being first, just worry about delivering a solid take on a well-accepted technology" approach is a new pair of four-channel amplifier product lines, each line comprising three models with different power ratings (400, 625, and 1,150 W/channel into 8 Ω). The CXD line is intended for permanent installations (e.g., churches, conference centers, auditoriums, and public addresses). The PLD series amplifiers are essentially the same as their CXD counterparts, minus the 70- and 100-V output lines, and with standard XLR input and NL4 output connectors. The target markets for the PLD series are touring shows, DJs, and other portable applications. Both product lines

have the same 2U form factor and use half-bridge outputs (see **Photo 1**).

The PLD and CXD amplifiers are QSC's first products to incorporate its proprietary Flexible Amplifier Summing Technology (FAST), a method that uses a multichannel amplifier as user-selectable sets of bridged, paralleled, or solo channels (see **Photo 2**). For example, a four-channel amplifier could be configured as four independent channels or as a two-channel amplifier with one side comprising two of the four channels in parallel (doubling the current capability) and the other side comprising the remaining two channels in bridge (doubling the voltage capability). It could also be used as a single amplifier with all channels in parallel, and or other permutations.

The concept of bridging and paralleling isn't new, and there have certainly been analog and other Class-D power amplifiers with the same capability. QSC claims that its new FAST amplifiers take a different approach and can achieve the same results with simpler circuitry and greater efficiency (e.g., in parallel mode). They don't need ballasting resistors. Although QSC is as yet unwilling to discuss how this is done, you can garner a few clues from its public statements.

First, the output stages are configured as a half-bridge. Second, parallel channels use the same comparator signal to control the output stage MOSFET gate drive circuitry. This can serve to balance dead time with shoot through, both of which become bigger issues with multiple-output devices. QSC doesn't specify whether paralleling occurs before or after the output filter. Third, instead of having separate feedback loops for each paralleled channel, there's a single feedback loop for all of the paralleled channels (see sidebar). In addition, you can infer the high-power supply rails are the same for each of the paralleled channels and that the regulator feedback is also fed from just a single channel.

One of the FAST amplifier's construction features is an output device that appears to be matched power MOSFETs on a common thermal carrier (see **Photo 3**). There's additional unspecified circuitry on the same carrier. The QSC engineers were rather coy about revealing what's inside the package, but it's entirely possible that it comprises the matched MOSFET pair and output control circuitry, much in the manner of Crown's DriveCore devices.

A look inside the box shows QSC followed the pattern of its earlier amplifiers and mounted the circuit board component side down (see **Photo 4**). This is said to reduce the dust accumulation on thermally sensitive parts, but does require some care in



Photo 2: The CXD lineup includes three models. The rear panel features four balanced Euroblock input connectors and four Euroblock touch-proof loudspeaker connectors enabling integrators to terminate wiring before connecting to the amplifier (a). The PLD series, designed specifically for portable sound applications, also consists of three models from 400 to 1,150 W/channel at 8 Ω (continuous) with four balanced XLR input connectors with loop-thru and six Speakon NL4 loudspeaker connectors for easy setup (b).

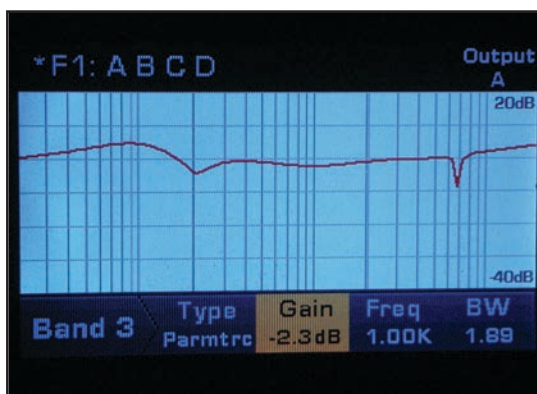


Photo 3: QSC's proprietary output MOSFET packaging contains two MOSFETs (presumably matched at the die phase) and steering circuitry.



Photo 4: This isn't a top view of the interior; it's a bottom view! Note the inverted construction, which QSC uses to avoid dust collecting on the circuitry and heatsinks.

Photo 5: The user interface for PLD/CXD DSP functions has several options, parametric equalization is shown here.



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.

thermal management, since the hot air flow through the heatsinks is restricted by the PCB (see **Photo 4**).

Divided We Stand

Professional sound equipment operators usually have a rack full of components to drive the loudspeakers. In addition to power amplifiers for each driver or stack, there are electronic crossovers, equalization, and limiters. Since these requirements are near-universal, the current generation of top-end professional amplifiers usually builds in these other features. The amplifier then functions as a flexible control center with full DSP capability and can be customized for the specific loudspeakers or drivers to which it's connected.

There are several ways in which the DSP control may be implemented:

Front panel control and display—The user or

installer can set up the amplifier configuration through a user interface integrated into the amplifier.

Network—The amplifier is set up as a device on a network and controlled via a computer on the network, and eventually through smartphone apps for true remote flexibility.

Integrated system—The amplifier system can include a network-linked computer internally to enable overall system control.

The QSC PLD and CXD amplifiers use the first type of implementation. The advantage of the current QSC approach is compactness and simplicity—the amplifier system's operation does not require an external computer network and can be highly automated via amplifier firmware. The user has a shallow learning curve and doesn't need to have expertise in networking. For example, instead of needing to decide "single-channel," "bridge-mono," or "parallel" configurations, PLD simply asks for loudspeaker impedance and power handling, then decides the right configuration internally to optimize power delivery.

The disadvantage of this approach is the lack of flexibility and limitations on system control, since the operator/installer must use the front panel interface or be tethered by USB. Future versions are likely to incorporate a more sophisticated network capability using a standard Ethernet connection, which is certainly the trend in modern professional amplifiers.

Included in the on-board DSP for the FAST amplifiers is source selection and mixing,

Bridging and Paralleling

The Ideal World

When it comes to theoretical amplifiers and theoretical loads, an amplifier is generally idealized as a voltage source. "Voltage source" means that the output voltage is linearly related to the input voltage, and the source impedance of the amplifier is zero. The latter implies that 100% of the voltage generated by the amplifier is delivered to the load. There are a few oddball amplifiers that act as transconductance amplifiers, that is, amplifiers that output a current that is proportional to the input voltage (and hence ideally have an infinite source impedance). However, most loudspeakers are not designed to provide a flat frequency response when driven that way.

The power delivered to the load, which is usually a loudspeaker, is a function of available voltage and the loudspeaker's impedance. In reality, the loudspeaker's impedance may be regarded as a fixed resistance R_L , and the average power dissipated in the load is provided by the canonical $P = V^2/R$,

where R is the loudspeaker resistance and V is the root mean square (RMS) voltage produced by the amplifier.

The Messy Real World, Phase I: Voltage Limits

A voltage amplifier has limits, notably the power supply voltage. An amplifier is simply a modulator of the power supply, so in general, its peak voltage output cannot exceed the power supply voltage. (There are some exceptions, but even in those cases, there is a hard limit to how much voltage the amplifier can source.) Let's say an amplifier has ± 50 -V voltage rails and the loudspeaker has an $8\text{-}\Omega$ impedance. The maximum sine wave power that can be delivered to the loudspeaker is then $V^2/R = (50)^2/(2)(8) = 156\text{ W}$. The "2" in the denominator comes from taking the RMS value of the voltage of a sine wave (50-V peak divided by $\sqrt{2}$). For a $4\text{-}\Omega$ loudspeaker load, the same nearly ideal amplifier could deliver twice the power. For $2\text{ }\Omega$,

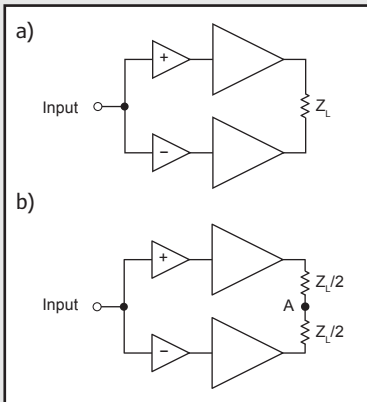


Figure 1: Bridge two amplifiers to make a single amplifier with higher voltage swing capability. The basic bridging topology is shown (a). The load impedance on the output of each amplifier in the bridge is half that of the load driven by a single amplifier (b).

have to dissipate. Both these consequences can add up to higher cost and poorer reliability.

A second and often better solution is “bridging” two amplifiers. In this process, a pair of amplifiers is dragooned into service (see **Figure 1**). The signal is fed with no change in polarity to one amplifier (see **Figure 1a**) and with reversed polarity to the other amplifier (see **Figure 1b**). Circuits to provide equal signals of opposite polarity are extremely simple to implement, often needing only a single transistor or op-amp. The load is connected between the “hot” terminals of the two amplifiers. This ideally allows double the voltage swing of a single amplifier.

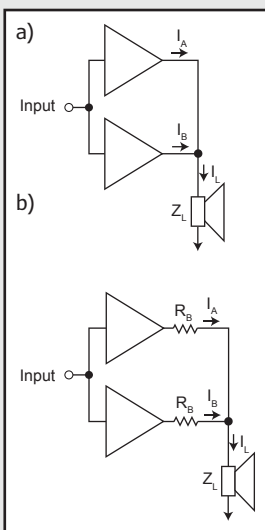


Figure 2: Parallel two amplifiers to make a single amplifier with higher output current capability. The basic paralleling topography—ideally $I_A = I_B$ —enables all current to be equally shared and delivered to the load (a). Use ballast resistors to equalize current draw from each amplifier in the parallel pair (b).

The Messy Real World, Phase 2: Current Limits

I previously noted that, for a

the nearly ideal amplifier could deliver four times the power, and so on.

Suppose you want to increase the power of a nearly ideal amp into that 8-Ω loudspeaker. One way to do it is to raise the rail voltages. The increase in power to the 8-Ω loudspeaker will scale as the square of the voltage increases—a doubling of the rails to ± 100 V will mean four times the power (625 W) can be delivered to the speakers. Unfortunately, this will double the voltage requirement for the transistors and double the amount of heat they

given output voltage, halving the load impedance doubles an ideal amplifier’s delivered power. This is interesting because in the limit of a short circuit for a load, the power delivered would be infinite. Clearly there are limits—if you try to draw too much current out of a transistor, it will overheat and burn out, which is generally not desirable. Hence most professional amplifiers use some form of current limiting to protect the output devices. The other limits are the output MOSFET channels’ resistance, the output filter’s resistance, and the power supply’s impedance, with the MOSFET resistance generally being the dominant factor.

One way to increase the maximum output current is to parallel two output stages (see **Figure 2a**). Ideally, each amplifier would source half the current, but reality is not always so kind. To provide some sort of feedback to equalize currents, ballast resistors are added to the amplifier outputs (see **Figure 2b**). The R_B ballast resistors work as advertised, but the downside is the power dissipated in them, reducing the maximum current that can be delivered to the load. Additionally, the resistors’ effectiveness increases with increasing resistance—better balance, more wasted power. This is why the absence of ballast resistors in the FAST amplifiers is significant.

The Messy Real World, Phase 3: Channel Matching

Even the real-world examples given make an unwarranted assumption: that the two amplifier channels have identical gain. If they don’t, then not only is there an imbalance in the delivered currents, but not all the current will be delivered to the load. **Figure 3a** shows a typical pair of paralleled amplifiers. The gains of each are determined by the ratio of R_F/R_S for each amplifier. If R_{F1}/R_{S1} is not equal to R_{F2}/R_{S2} (and given real-world component tolerances, they are unlikely to be exactly equal), then the amplifiers will attempt to have different output voltages. Current will be driven from one amp to the other until the voltages are equal—or they catch fire.

Figure 3b shows QSC’s unitary feedback approach. A single feedback loop corrects both paralleled channels, ensuring near perfect matching and a minimum of wasted current. The switching between a pair of feedback loops and a single one is automatically handled in the amplifiers’ circuitry.

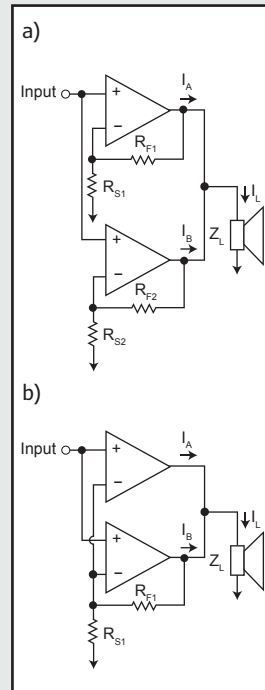


Figure 3: Feedback systems can be used for paralleled amplifiers. Conventional approach—tolerances in the feedback components mean that one amp drives current into the other (a). QSC’s single feedback system is automatically implemented when channels are paralleled (b).



Photo 6: QSC's new amplifier platform was designed as a custom Class-D multichannel high-performance device with agile power distribution across one, two, three, or four channels via Flexible Amplifier Summing Technology (FAST). It also has a PowerLight universal switchmode power supply with power factor correction (PFC) for high efficiency and low weight.

electronic crossover capability (6-48 dB/octave, Bessel, Butterworth, and Linkwitz-Reilly), parametric equalization (up to five bands of parametric or shelf EQ), delay, and limiting (see **Photo 5**). What this means is that a portable equipment user has fewer items to lug around and that there's more flexibility in setting up the amplifier for varying speaker systems.

Likewise, installed system users can configure the same amplifier for different requirements, which reduces the number of different types of amplifiers needed, simplifying installation design and implementation.

A nice feature is a group of presets for different brands of popular professional speakers (e.g., Peavey, Yamaha, JBL, EV, B52, and Cerwin-Vega) with optimized crossover frequencies and amplifier configuration to match nominal impedance, power, crossover, and delay requirements. Again, this simplifies setup for the non-expert user. The DSP capabilities for QSC's own speakers are more extensive, including FIR and dynamics (see **Photo 6**).

As discussed in the sidebar, you can optimize the amplifier configuration for a given loudspeaker impedance. The amplifier does this automatically if you give it the impedance and power. But unfortunately, the loudspeaker's impedance is not a fixed value and can, in fact, vary almost 10:1. So a sophisticated user will consider not only the nominal speaker impedance, but what the impedance is at frequencies that are likely to have the highest power consumption given the source signal's spectral balance. If a soprano is miked, the impedance at 600 Hz will be far more relevant than the impedance at 60 Hz.

Born in the USA

Although QSC is not unique in having design and engineering based in the US, it is one of the few companies that are actually manufacturing product domestically. Domestic manufacturing is unusual in electronics these days, especially consumer electronics. Regulations and labor costs have made offshore (usually China) manufacturing economically irresistible. The quality of facilities and work overseas has increased, so it's possible now (though not trivial!) to do world-class manufacturing to US-quality standards in Chinese technical hubs such as Shenzhen. China is no longer just a source of lower labor costs, restricted to making labor intensive goods; in fact, wage cost increases have caused many low-tech commodity companies (e.g., shoes and clothing) to relocate to other areas of Asia.

The trend toward parity in manufacturing costs highlights the insuperable difficulties associated with off-shoring, mainly logistics. "QSC is committed to developing and manufacturing professional audio products (e.g., signal processing, amplifiers, and loudspeakers) in the US," said Dale Sandberg, Product Manager for the FAST platform. "We have developed significant skill sets in this area and have invested heavily in the tools necessary to produce efficiently in the US. That is not to say that we do not believe in overseas production, but we choose to do so when it meets our needs."

According to Sandberg, QSC currently manufactures all models of both the PLD and CXD line in the US (see **Photo 7** and **Photo 8**). However, it is in the process of transitioning the PLD to an overseas manufacturer.

"It takes time to bring up manufacturing with the correct techniques and sufficient test resources. It was easier and much quicker for us to develop and prototype the amps in our plant here in the US than work with an overseas manufacturer," said Sandberg.

Besides speed of development, several other



Photo 7: The QSC assembly line in Costa Mesa, CA, is set up in a sequential workflow arrangement to produce a product from start to finish by physically grouping tasks and machines to minimize unnecessary time and motion expenditures.

factors go into a decision to build a product in QSC's Costa Mesa factory versus overseas. Cost of goods is a consideration, as is their desire to continue manufacture in the US.

"We also have to think about demand planning. Some products tend to have a more consistent demand while others have greater variability," said Gary Tschetter, QSC's Vice President of Marketing. "For example, amplifiers intended for installation are subject to swings in demand that are driven by large projects such as stadiums, theme parks or big cineplexes. It's far easier for us to manage this demand when we are building these amplifiers here in Costa Mesa. Building amplifiers overseas means that forecasting lead-time is much longer. To maintain desired service levels, we would have to carry large quantities of safety stock. For some types of product, the cost of carrying safety stock can outweigh savings from overseas manufacture."

Sandberg agrees and looks to the future. "As is typical, we experience a new product ramp where the demand increases over a period of years then plateaus. The period is longer for some products (installation products in particular) where the product can increase in sales volume over five or more years before the plateau occurs. Production tries to predict and keep up with demand. I see these products succeeding in the marketplace and the demand increasing. I also see expansion of the platform with the development and production of new models. The cost of the product is not the only



Photo 8: Part of QSC's FAST amplifiers' assembly process includes automated parts placement.

thing we consider. We produce more now in the US than we ever have and can see it increasing in the coming years."

Overall Impressions

Clearly, QSC has aimed to make a solid, reliable unit with great ease of use and flexibility and a reputable company standing behind it. While the new FAST amps aren't challenging the boundaries of networking and control, they provide a simple and understandable setup and appear to be a great choice to match up with QSC's speakers. 

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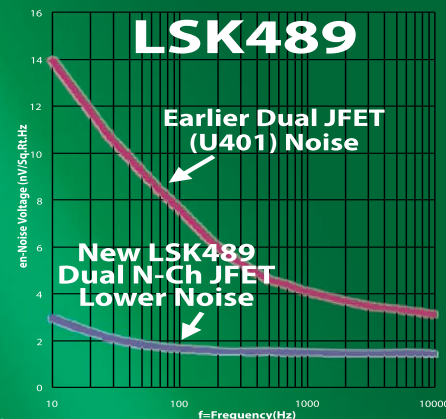
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The DE980TN-8 by B&C Speakers

By
Vance Dickason
(United States)

A High-Performance Compression Driver

Vance Dickason recently examined the DE980TN-8 from B&C Speakers, an Italian pro sound OEM. The DE980TN-8 is a neodymium-motor compression driver coupled with the company's ME90 horn. The DE980TN-8 joins B&C Speakers's 1.4" compression driver series that includes the DE620TN, the DE820TN, the new DE880TN, and the DE920TN.



Photo 1: B&C Speakers's DE980TN-8 is a compression driver coupled with the ME90 horn.

This series has several unique features, including titanium diaphragms, neodymium ring magnet motor assemblies with cast-aluminum heatsink caps, and copper shorting rings. The B&C Speakers DE980TN-8's throat diameter is 36 mm (1.4") coupled to a 75-mm (3") diameter voice coil wound with copper-clad aluminum wire (CCAW). It also includes nominal 110-W-rated power handling (220-W continuous) and color-coded chrome-plated push terminals. The ME90 horn supplied with the DE980TN-8 has a 1.4" throat diameter, 80° horizontal × 60° vertical, constant-directivity cast-aluminum horn with a solid 900-Hz cut-off frequency.

I used the LinearX LMS analyzer to produce the 200-point sine wave impedance plot shown in **Figure 1**. The solid black curve in **Figure 1** shows the

DE980TN-8 mounted on the ME90 horn. The dashed blue curve represents the compression driver without the horn. With a 6-Ω DCR, the DE980TN-8/ME90's minimum impedance was 7.8 Ω at 5.9 kHz.

For the next frequency-response tests, I recess mounted the B&C Speakers DE980TN-8/ME90 in an enclosure with a 10" × 15" baffle and used a 100-point gated sine wave sweep to measure both the horizontal and vertical on- and off-axis at 2.83 V/1 m.

Figure 2 displays the compression driver/horn combination's on-axis, which is smooth with no major anomalies out to 18 kHz. There is a small mode at 15.6 kHz and a declining response as the frequency increases, which is typical of constant directivity horns. B&C Speakers's recommended crossover frequency for the DE980TN-8 is 1.2 kHz with a second-order or higher slope.

Figure 3 shows the on- and off-axis response in the horizontal plane. **Figure 4** gives the DE980TN-8's normalized horizontal plane response. **Figure 5** provides the on- and off-axis response in the vertical plane. **Figure 6** shows its normalized vertical plane response. **Figure 7** displays the two DE980TN-8 compression driver samples to be closely matched in a two-sample comparison.

I used the Listen AmpConnect ISC analyzer and 0.25" SCM microphone to measure distortion and generate time-frequency plots. For the distortion measurement, I mounted the B&C Speakers DE980TN-8/ME90 combination with the same baffle used for the frequency-response measurements. I used pink noise stimulus to set the SPL to 104 dB at 1 m (1.88 V) and placed the Listen microphone 10 cm from the horn's mouth. This produced the

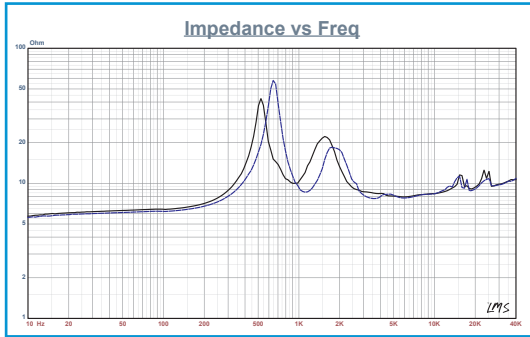


Figure 1: B&C Speakers DE980TN-8 free-air impedance plot

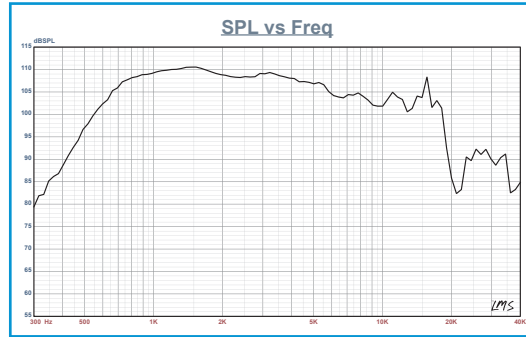


Figure 2: B&C Speakers DE980TN-8 on-axis frequency response

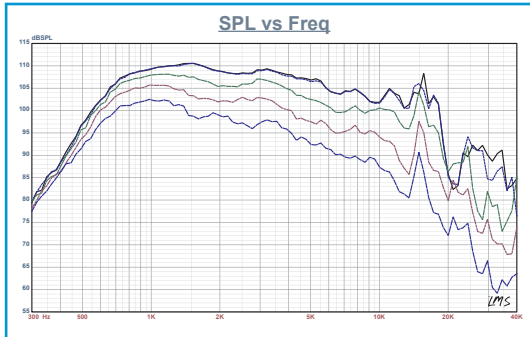


Figure 3: B&C Speakers DE980TN-8 horizontal on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

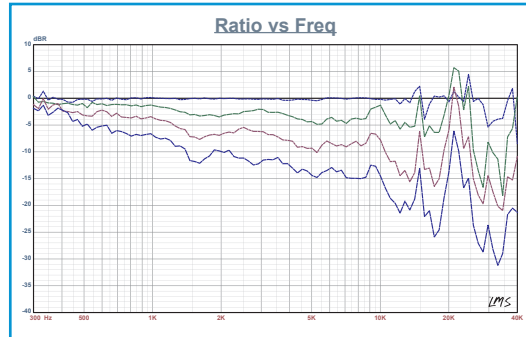


Figure 4: B&C Speakers DE980TN-8 normalized horizontal on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)





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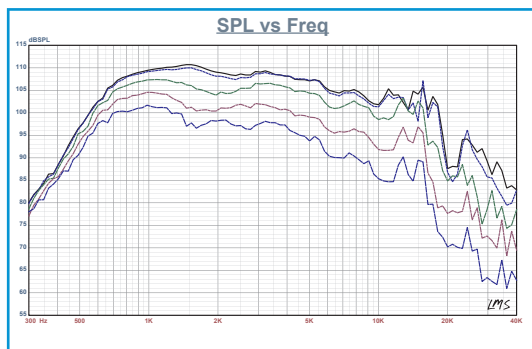


Figure 5: B&C Speakers DE980TN-8 vertical on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

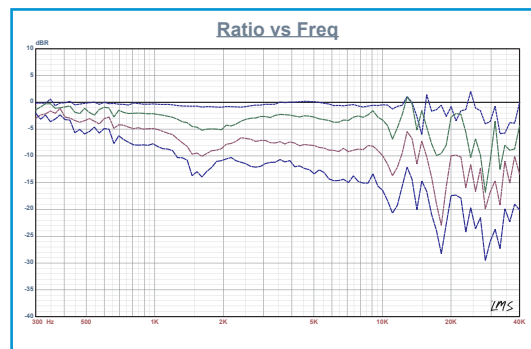


Figure 6: B&C Speakers DE980TN-8 normalized vertical on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot; 60° = dash)

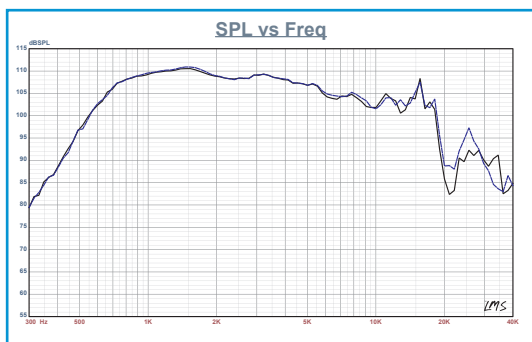


Figure 7: B&C Speakers DE980TN-8 SPL comparison

distortion curves shown in **Figure 8**. I then used SoundCheck to get a 2.83 V/1 m impulse response and imported the data into Listen's SoundMap time-frequency software. **Figure 9** provides the resulting CSD waterfall plot. **Figure 10** shows the short-time Fourier transform (STFT) plot.

The DE980TN-8's build quality and performance is excellent. For more information, visit www.bcspeakers.com, or in the US, contact Bennett Prescott at bprescott@bcspeakers.com.

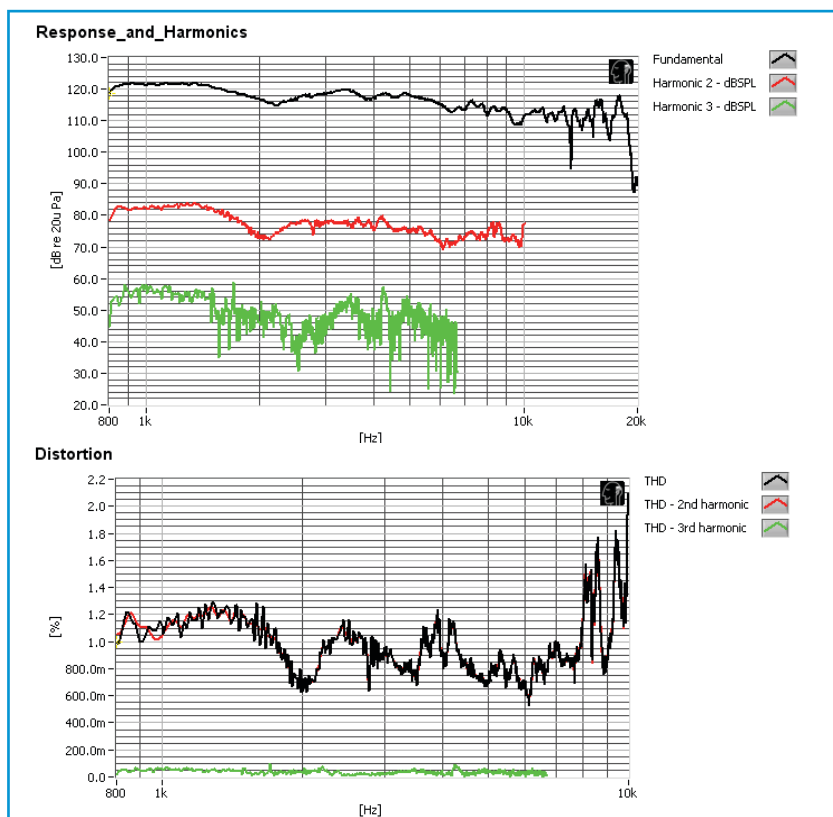


Figure 8: B&C Speakers DE980TN-8 distortion plots

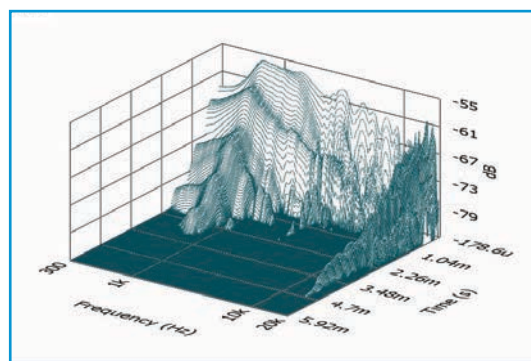


Figure 9: B&C Speakers DE980TN-8 CSD waterfall plot

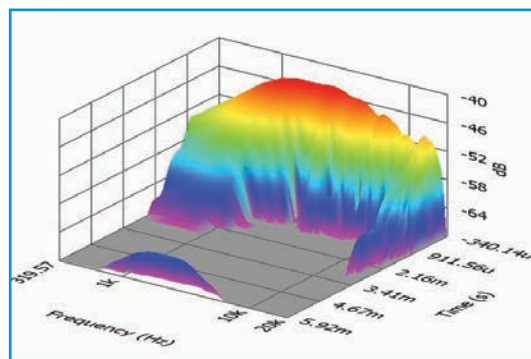


Figure 10: B&C Speakers DE980TN-8 STFT plot

The Lowdown on Woofers, Subwoofers, and Bass Shakers (Part 1)

An Introduction to Woofers

By
**Mike Klasco and
Steve Tatarunis**
(United States)

This column has covered several speaker parts (e.g., cones and voice coils) and investigated many tweeters including domes, ribbons, air-motion transformers, and so forth. The speaker exploration process now shifts to the low-frequency world of woofers and subwoofers. In this article series, you'll review the history, technological milestones, and developments that have resulted in the modern-day woofer and subwoofer. In upcoming articles, you'll explore the internal workings of woofers, subwoofers, super-subwoofers, and bass shakers.

Woofers and subwoofers are application-specific drivers for bass reproduction. To properly reproduce low frequencies, a lot of air must be moved so the speaker will need to have a large excursion or surface area. In fact, a combination of these properties is desirable as is the use of multiple speakers, especially if you want to keep distortion low or make a lot of noise.

Woofers vs. Subwoofers

Let's begin with some descriptions and term definitions. Tutorials often define the differences between woofers and subwoofers by their frequency ranges. With consumer electronics (e.g., docking stations, Bluetooth speakers, soundbars, and 2.1 and 5.1 systems) dominating the speaker world and "big box" speakers (i.e., bookshelf or floor standing designs) a fading memory, we need to adjust our terms to accommodate the present-day audio landscape. So, let's just say the woofer reproduces the sound below where the midrange or tweeter is rolled off by the crossover network.

Subwoofers are intended to augment a loud-speaker system's lower frequency ranges. For audiophile systems, the subwoofer may relieve the full-range speaker of some of its excursion requirements. But more commonly in today's systems, the subwoofer handles the majority of the bass reproduction and the satellite speakers kick in at 80 or 120 Hz, one of the two Dolby standard

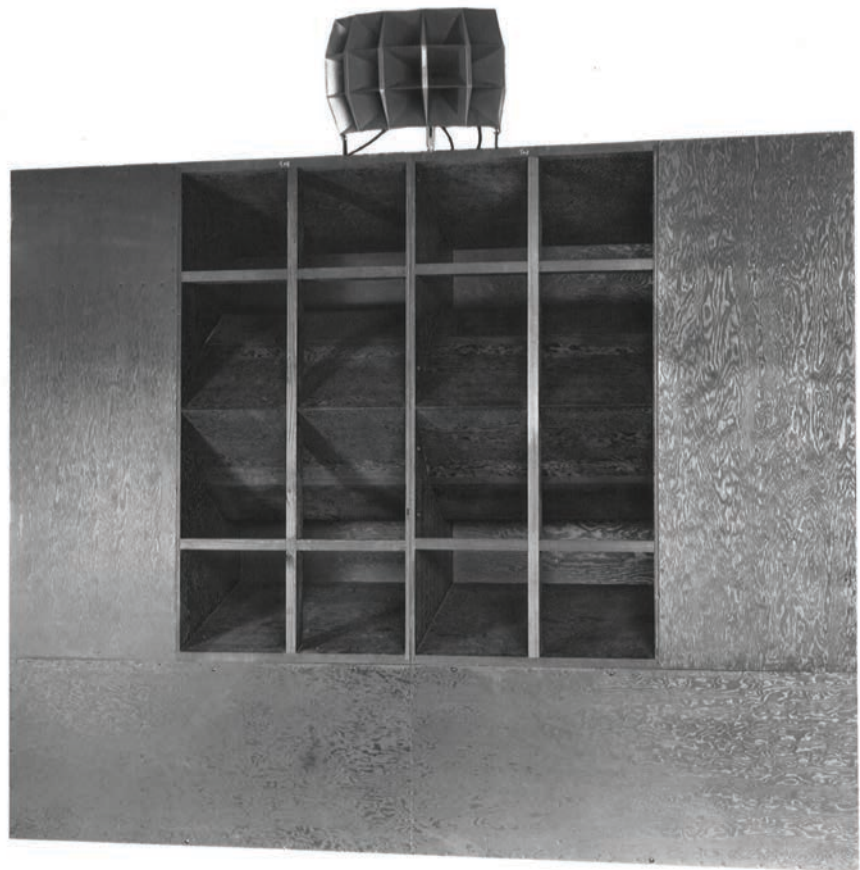


Photo 1: The Lansing Shearer Horn is an example of an early woofer system. Its 2" diameter voice coils were enormous by 1930s standards. (Photo courtesy of Harman International)

15-INCH SPEAKER MECHANISM

TYPE LC-1A

SPEAKERS



FEATURES

- Excellent frequency response, uniform 50-15,000 cycles
- Wide angle sound radiation of all frequencies
- Low non-linear distortion
- Ideal for monitoring AM-FM television programs
- Alnico V magnets

USES

The LC-1A is a "Broadcast Quality" loudspeaker with a low distortion, wide angle distribution, of extended frequency range, and specifically designed for use in recording studios, executive offices, reception rooms, sponsors' booths or any location that warrants a pleasant setting and tasteful styling.

For applications where it is desired to mount the mechanism on a wall baffle, ceiling, etc., the speaker mechanism may be used with assurance that the entire frequency range will be realized. The speaker's outstanding performance makes it ideal for wide frequency range wide angle radiation.

DESCRIPTION

The LC-1A is a dual-cone speaker mechanism of the direct radiated type, consisting of high and low frequency units mounted co-axially together. The 2" high frequency cone and the aluminum wound voice coil has a low mass utilizing the wide angle of the shallow, low frequency cone to effect its remarkable directional pattern (see curve). An equilibrium has been reached between the electrical and mechanical design to impart a high frequency radiation of 120° arc with a loss of approximately 6 db at 15,000 cps. This eliminates the conventional "beam effect" usually experienced at this frequency.

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Photo 2: An early advertisement for the RCA LC-1A describes the speaker as "broadcast quality."

transition points. When used with a soundbar, the subwoofer may be remotely located and the entry-level packaged soundbar systems may have a much higher transition frequency—perhaps as high as 250 Hz.

Full-range speakers tend to be two-way designs with a woofer and tweeter. Sometimes a midrange driver is added to make it a three-way design. In these systems, the woofer reproduces the lows to a point just below midrange frequencies. In some systems, the reproduction may go down to 50 or even 40 Hz, depending on the design. A few larger speakers can go even lower, but not many.

A subwoofer takes over where a woofer's response naturally rolls off, maybe a bit higher in some systems. As a rule of thumb, the lower the transition frequency is from woofer to subwoofer, the better the system's blend. However, dragging

the satellite speaker woofers down too low may result in distortion when pushed too hard.

We'll go into some detail on subwoofers later, but in this introduction to woofers, let's just say these have a common bass speaker/enclosure, a pair of "satellites" for stereo, and possibly additional satellites for surround sound.

One or Two Subwoofers?

The listener's ability to localize a sound source gets progressively weaker below 250 Hz. By 120 Hz, most listeners are unable to perceive much spatiality. On the other hand, some studies indicate there is a more natural spatial ambience in large spaces when multiple subwoofers are used. However, comb filtering becomes an issue with spaced subwoofers. Another aspect to consider is subwoofer placement; however, room acoustics and acoustic perception will need to be addressed in a separate article.

Woofer History—The Early Years

During World War I, Western Electric developed motion pictures with sound for military training. By the late 1920s, ongoing sound system development focused on bringing sound playback to movie theaters. The audio engineers who developed these products had telephony backgrounds, so it is no surprise that these early theater speaker designs were huge, scaled-up telephone receivers loaded by large horns. After further evolution and development, this driver became the precursor of the modern day compression driver. For speaker history, it was truly a giant step, but in reality, this was actually a midrange design attempting to reproduce bass frequencies—no real woofers here!

Having learned from Western Electric's first efforts at low-frequency reproduction, some serious products quickly followed. By 1931, Western Electric introduced the Jensen low-frequency 18" drivers for their flagship theater systems to fill the bottom-end response down to 50 Hz.

By 1935, Lansing also began offering theater sound systems. Lansing developed and manufactured the Shearer horn system woofers (with input from the system's namesake, MGM's Sound Department head Douglas Shearer). The woofers had massive (for the time period) 2" diameter voice coils (see **Photo 1**). The Lansing Shearer Horn system received a technical achievement award at the 1936 Academy of Motion Picture Arts and Sciences ceremony.

Broadcast radio and recording studio monitoring requirements also drove the need for extended bass, low-distortion speaker development. In 1937, Lansing introduced the Iconic system, a two-way

monitoring system using a 15" woofer mounted in a bass reflex enclosure with a small-format horn for the mid and high frequencies.

Jumping ahead to the 1940s, we find another notable high-fidelity (hi-fi) grade monitoring speaker with a high-performance woofer. Harry Olson, at RCA, developed the legendary LC1A broadcast monitor (see **Photo 2**). A coax tweeter enabled its full-range sound reproduction to modern audio standards. Around the same time, the Altec Lansing introduced the "Duplex" co-axial designs—including the 604, which used a 15" woofer with a 3" voice coil. Variants of the original Duplex designs were still in production into the late 1990s.

We have only mentioned a few pivotal woofer designs from US speaker companies, but I think this provides a bit of a timeline for the developments. To clarify, these movie theater, broadcast, and recording studio monitor speakers were not available to consumers. Following World War II, the home hi-fi industry blossomed and many commercial and broadcast speaker providers expanded their products for the consumer market. In upcoming articles, we will discuss the woofer's internal components, but it is worth noting that one significant transition following World War II was the use of permanent (Alnico) magnets, which replaced the field coil electromagnets found in earlier designs.

Lessons Still Used Today

The last decade has experienced a proliferation of home theater setups. Compact high-output subwoofers are increasingly more prominent in many mid- and upper-range home theater systems. Since we are both "old timers," we are often asked who devised this "sub-sat" technology. The easy answer is that the "micro" subwoofer is the evolutionary product of several innovative and creative people.

The first application was autosound, which was once a huge hobbyist's market. But today, there are only scant remnants of the autosound kilowatt, "crank-it-up" fanaticism of the 1990s and early 2000s. A vehicle's interior compartment is relatively small. However, as the appetite for intense bass grew, vehicles were the ideal environment in which to create gut-wrenching bass.

The authors admit they cannot take credit for these subwoofer contributions; however, one of them was present for a number of these breakthroughs. So here is an abbreviated history of deep bass, woofers, and subwoofers that delivered more wallop in less space than their less capable competition.

Anyone can jam a large woofer into a small



Photo 3: Bose Acoustimass 5 includes two double-driver cube speakers. (Photo courtesy of Bose)



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Photo 4: Infinity built its first speaker system, the Servo-Statik 1A, in the 1960s.



box. Reproducing satisfactory deep bass from that enclosure is considerably more challenging. As a woofer moves inward, the air in its enclosure compresses, acting like a spring. This spring shifts the in-box fundamental resonance upward and rolls off the bass. The compressed air needs to escape and seeks a pressure release. Without any pressure release, a strong magnetic system can buckle a cone and surround if they are not as stiff as the air spring.

Over the last 70 or more years, various researchers have sought ways to improve the small woofer box. Anthony Hoffman, a partner of Henry Kloss at KLH, developed and created Hoffman's Iron Laws in the 1960s. Hoffman's Iron Laws are mathematical formulas that relate to the relationship between speakers and, in particular, to sealed enclosures relating to bass extension, efficiency, and small enclosures.

The Iron Laws state that you can only have two of these three attributes in speaker systems. We cannot have all them, at least not without a lot of effort and intelligence. Ultimately, big speakers in large enclosures will tend to have good bass and efficiency. Small speakers will have less efficiency and less extended bass (and limited maximum acoustic output).

We have explored attempts to bend Hoffman's Iron Laws in previous articles about enclosures and enclosure stuffing. Hoffman's Iron Laws can

be, if not broken, at least bent a bit using unique speakers, bass reflex venting, and other box-tuning and stuffing techniques, trick amplifier servomotor feedback circuits, and signal processing.

Woofers and Subwoofers in 2.1 Systems

Many of us own or have owned 2.1 systems. These are common home theater, stereo system, and computer speaker setups that have two satellite speaker boxes or a soundbar with a separate subwoofer enclosure.

Low frequencies are not easily localized in small rooms; hence many stereo and multichannel audio systems feature only one subwoofer. A single subwoofer can be placed off-center without affecting the perceived soundstage, since the sound produced is difficult to localize. The intent with a system that includes a subwoofer is to use small main speakers (a stereo typically has two and a surround sound or movie tracks typically has five or more) and hide the subwoofer elsewhere (e.g. behind furniture or under a table). The subwoofer is also used to augment an existing speaker to save it from having to handle woofer-destroying low frequencies at high levels.

In 1988, Bose introduced its original Acoustimass 2.1 speaker system as the Bose Acoustimass 5 Series (see **Photo 3**). Its two double-driver cube speakers connected to a passive bass enclosure then to the receiver. The Acoustimass and many similar stereo 2.1 systems followed the design concepts pioneered by Paul Weathers 20 years earlier. His Compass Triphonic System featured an integrated electronics unit, a pair of satellite speakers, and a common woofer similar to a modern 2.1 system.

Although common subwoofers had previously been used, the Triphonic was the forerunner of today's modern home theater subwoofer/satellite products. This design was an early integrated audio system. The customer only needed to plug in the turntable. It was intended as the second audio system in a bedroom or den, or for use in a dorm room. Sadly, this design did not gain acceptance due to rock music's growing popularity.

Ten years after the Triphonic's launch (and 10 years before Bose's 2.1 system), Triad used tiny satellites and compact subwoofers to resurrect the concept. Much of the credit for the subwoofer becoming a stand-alone product category should go to M&K Sound, which formally started its business in 1974 and introduced subwoofers plus satellite systems in 1976.

The Infinity Servo-Statik 1 was another far

more extraordinarily ambitious 2.1 system, also conceived in the late 1960s. It became famous for its unpredictable behavior. Gene Cerwinsky's Cerwin Vega (back then just Vega) made many compact deep-bass high-excursion subwoofers, but most of them were intended for clubs and concert sound.

One of the early high-excursion subwoofers was made for the Infinity Servo-Statik home stereo. It used a single 18" subwoofer with a servo-control coil and its own power amplifier. The full-range speakers were a pair of 6' high dipole electrostatic panels. The power subwoofer provided the bass that could not be delivered by the electrostatic dipoles (see **Photo 4**)

Godehard Guenther of Analog and Digital Systems (ADS) developed another notable 2.1 system. The ADS Subwoofer-Satellite 2300 system was introduced in the fall of 1980 and was comprised of a powered subwoofer (1,500 W with dual 10" woofers, each mounted in its own separate enclosure), an electronic crossover/processor, and a pair of passive 6" two-way satellites.

Modern-Day Woofer Evolution

Let's jump to the long-throw woofer's early development. In 1954, Edgar Villchur of Acoustic Research (AR) introduced the AR-1W, which packed a high-excursion woofer into a relatively compact sealed enclosure (see **Photo 5**). This woofer-only design required a separate full-range or tweeter for full bandwidth reproduction. The AR-1W woofer's surround and spider were loose, with the sealed air in the enclosure providing a linear spring restoring force.

Stereo phonograph records, followed by multiplex stereo FM radio, were developed during the late 1950s and audio changed from a DIY construction hobby to a commercially available method to play back music. However, its success was dependent on finding a way to unobtrusively place a pair of speakers into a living room—without resulting in divorce—which, in turn, created another important (but rarely published) loudspeaker specification: the WAF or "Wife Acceptance Factor."

Amplifier power was scarce in the 1950s and prior to the AR-1W, people used gigantic highly efficient JBL, Klipsch, and Altec Lansing speaker systems derived from movie theaters sound systems. In the late 1950s and early 1960s, tube amplifier power—for example, Dynaco's 60 W unit (two of these are needed for stereo), the McIntosh MC275 (dual 75 W), Bogen, and the Harman Kardon Citation amplifiers—were widely used. To put this in perspective, amplifiers used for movie theaters in



Photo 5: The AR-1Ws are compact sealed enclosures that contain high-excursion woofers.

the 1930s and still commonly used in the 1940s were typically below 10 W.

In the early 1960s, James Novak, a talented engineer and mathematician who worked for Jensen, developed a series of equations for the bass reflex speaker. One of the reasons acoustic suspension achieved acceptance is that incumbent bass reflex designs were often poorly tuned and had "one-note bass" due to long decay times at resonance. Using Novak's turning techniques, Jensen introduced a range of compact speakers that had significant bass output at higher efficiency.

Other researchers (e.g., A. Neville Thiele, Richard H. Small, and J. E. Benson) continued to expand on Novak's ideas. Today's commonly used speaker design computer simulation programs are based on these researchers' work.

Designing a bass reflex duct long enough and large enough in diameter so the vent does not create wind noise is problematic when you are trying to get really deep bass. To keep the air velocity low, the duct can become larger than the enclosure.

Harry Olson discovered a solution and published a paper on his concept of a passive radiator as a vent substitute. Instead of a deep vent or port, a diaphragm with a suspension (but no motor) could be used. The diaphragm's mass would be equal to the mass of the column of air in a long vent.

In the late 1960s, JBL offered a range of passive radiators that DIYers could install in pre-engineered enclosure designs. JBL's passive radiators could be tuned by mass loading—the user was instructed to bolt some specified number of heavy steel washers to the dust cap to optimize the passive radiator's performance.

This completes our brief introduction and historical overview of the evolution of woofers and subwoofers, including notable milestones and iconic products. In our next article, we will explore low-end reproduction. 

We Really Have (or Want) Great Acoustics

By
Richard Honeycutt
(United States)

As an acoustical consultant, Richard Honeycutt has learned a good bit from listening to clients. Often what people tell him is not what they want to communicate. For example, “I can’t hear the lecturer” often means “I can’t understand what the lecturer is saying.” So he tries to interpret what the clients mean rather than what they actually say.

I often hear “this room has really good acoustics.” I have heard that statement made in rooms in which speech cannot be understood at a distance of 3 m, or where the program (music or speech) was completely buried in noise. The main cause of this confusion is that “good acoustics” is, for most people, a subjective judgment. Much of the focus



of acoustical research in the last century has been tying objective measures to subjective responses. Few people know how to use the objective terms that have been developed. Thus, a sound system that is beginning to develop feedback squeal may be confusingly described as “having reverberation.”

Another cause of confusion is the incorrect impression that “good acoustics” has a specific meaning. In fact, good acoustics for a classroom is not the same as good acoustics for a concert hall or a radio studio. One way to clarify our understanding—and improve communication—is to examine the meanings of some important objective acoustical measures and see what these measures represent as “good acoustics” for various venues.

Venue Acoustics

Any venue’s major acoustical concerns can be addressed by three questions: Can people hear the program (or the people across the table in a restaurant or cafeteria)? Can they understand what others are saying? Are some acoustical factors annoying? In venues with sound systems, the ability to hear depends partially on sound system design, installation quality, and user adjustments. But even the best sound system cannot compensate for serious acoustical problems. Noise is the most common reason people can’t hear the program. Often this is HVAC noise, but sometimes it is noise from outdoors or from adjacent rooms. Recently, a colleague of mine was asked to evaluate the sound system in a venue for which he had done the acoustical design. At the

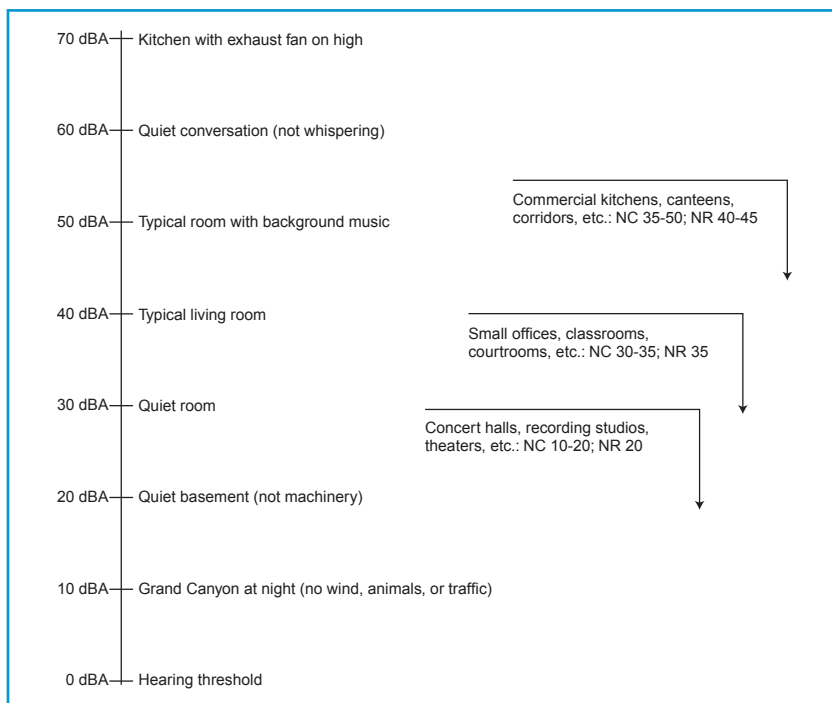


Figure 1: Appropriate background noise levels can be specified in decibel, A-weighted (dBA), or Noise Criterion (NC) in the US, or Noise Rating (NR) in Europe.

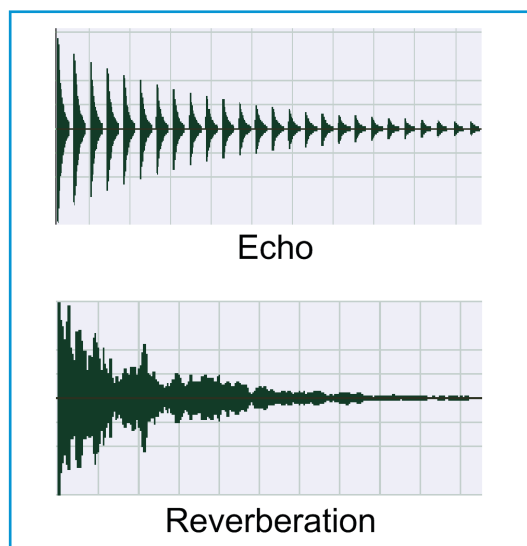


Figure 2: An echo is repeated sounds, but reverberation is prolonged sound.

beginning of the original project, he had offered to analyze the proposed HVAC design to identify potential noise issues, but the client assured him the mechanical engineer had taken care of that. My colleague was measuring the sound system's performance when the HVAC system turned on. Suddenly, he was no longer able to take meaningful measurements because of the 57-dBA noise level—about 20 dB too high. However, people had complained about being unable to hear. They had not complained about the noise.

Excess Noise

The reason that excess noise is often unrecognized is that our ear/brain system tends to ignore steady sound. Thus, we don't notice steady noise, but we do notice that we can't hear or understand speech. This can apply not only to HVAC noise, but also to noise from other equipment. Noise from external sources often has a more noticeable character: you hear thumping bass from a nearby night club or "boom car," you notice the growl of an air compressor, you are distracted by the drone of a lawn mower, or the neighbor upstairs seems to have hobnails in his combat boots. As a result, when we hear these sounds, we complain about the noise. The bottom line is that noise can cause a client to complain about being unable to hear or about the noise itself.

Other types of venues in which excess noise is often not directly noticed are cafeterias and restaurants. A hundred people talking in a room that has too little acoustical absorption can create surprisingly high sound levels. This noise causes people to talk louder to be heard over the noise, which of course compounds the problem. Since we unconsciously adjust our talking volume, we may not notice the noise level. Hearing-challenged people have a particularly difficult time in such environments since hearing aids cannot distinguish between noise and desired sound: they amplify everything. The solution is to add enough acoustical absorption to keep the internally generated noise under control. **Figure 1** shows the maximum background noise level in dBA for various venue types.

"I can't understand what people are saying" is a complaint usually uttered only by fairly perceptive listeners. More often, the complaint will be worded, "I can't hear." For now, let's discuss only cases where the sound level and signal-to-noise ratio (SNR) are adequate.

Echoes and Reverberation

In some cases, speech intelligibility problems can be caused by an inadequate high-frequency

response in the sound system or by the listener having some recognized or unrecognized hearing challenge. In such situations, the solution is obvious, although it may not be straightforward to implement. Common acoustical issues that can harm speech intelligibility include echoes and excess reverberation. Probably the easiest way to explain the difference is that echoes are discrete repetitions, whereas reverberation is a continuous extension of the sound. Echoes are like, clap, clap, clap, but reverberation is cla.a.a.a.p. **Figure 2** shows waveforms representing echo and reverberation applied to a single bongo beat.

The objective measure of reverberation is "reverberation time," often abbreviated "RT60." This is the amount of time required for sound at the normal conversation level to fade to inaudibility in a very quiet environment. There is an ideal reverberation time for each venue, ranging from under a half second for small speech rooms and control rooms to as much as 7 s for a concert hall for medieval music such as Gregorian Chant and slow organ music. While too much reverberation hurts speech intelligibility as well as the articulation of fast-tempo or percussive music, too little reverberation makes music sound dry. So the optimal RT60 heavily depends on the type of program for which a venue is used.

There are other important factors as well. We expect a longer RT60 in larger rooms, and if this expectation is not met, the room will sound dry.

Figure 3: Optimal RT60 varies according to purpose and room volume.

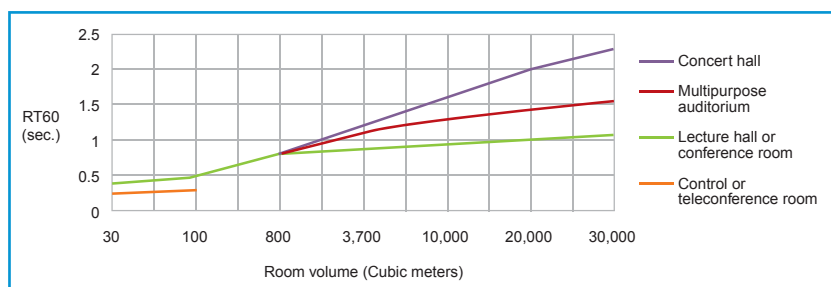
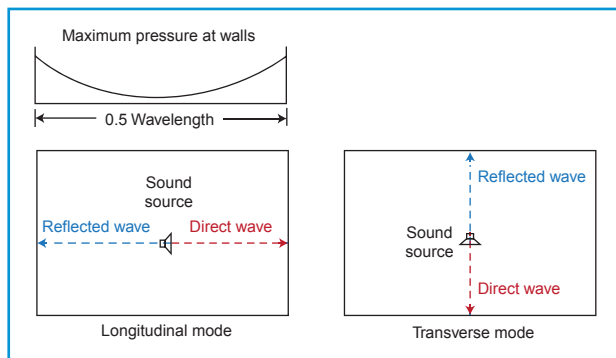


Figure 4: Room modes occur at frequencies where a reflection's round-trip path is a multiple of the wavelength.



Recording studios are usually designed to have RT60 similar to concert halls of the same size, used for the same type of music. Most control room designers prefer that the room add no character of its own, so they design for minimal reverberation. Multipurpose venues are often designed a bit on the dry side, because while a too-dry musical performance may be less than satisfying, poor speech intelligibility from high RT60 can render the venue useless.

Figure 3 shows ranges of optimal RT60 for various purposes and room volumes.

To be recognized as an echo, a reflected sound must be delayed at least 50 ms behind the original sound. More delay, up to about 200 ms, progressively decreases intelligibility. Probably the most commonly recognized venue in which echo problems are epidemic is non-acoustically-treated gymnasias. Echo problems result from the combination of acoustically reflective walls and large distances. This combination creates high-level echoes with long delay times. While reverberation is useful in the correct amounts for specific venues used for particular purposes, acoustical echoes should almost always be avoided.

One additional acoustical annoyance pertains mostly to small rooms (e.g., control rooms, production rooms, and small listening rooms, and home studios): boominess caused by room modes or resonances. Any time a sound wave with a particular

frequency bounces off one or more reflecting surfaces and returns to its point of origin in phase, a “standing wave” exists. It creates locations in the room where sounds of that frequency will be emphasized and other points where they will be weak. The simplest example of a path that can create a standing wave is that of a sound source located near the center of a rectangular room, with the sound repeatedly bouncing back and forth as it travels the length of the room again and again. This is called a “longitudinal mode.” Modes involving reflection between the long walls are called “transverse modes.” Other paths can include more bounces involving just the two long walls, all four walls, or the four walls plus the ceiling and floor.

Room Modes

All rooms have room modes, but in large rooms such as auditoria, the modal frequencies are too low to affect what we hear. That is why room modes are only problematic in small rooms. Figure 4 shows the formation of longitudinal and transverse room modes.

Table A lists some of the modes for a small (4 m long × 3 m wide × 3 m high) room and a large room (33 m long × 20 m wide × 13 m high). Notice that as frequency increases, the spacing between modes decreases.

Above a certain frequency, the “Schroeder frequency” modes are so closely spaced that they cause no significant frequency-dependent boosting or cancellation. This frequency can be calculated by the following equation:

$$f_s = 2,000 \sqrt{\frac{RT60}{V}}$$

where V is the volume of the room in m³.

Since the Schroeder frequency is determined by the room volume, which, in turn, is usually determined by nonacoustical requirements, you must control modal effects. To do so, arrange the room shape to avoid creating multiple low-frequency modes near the same frequency and by weakening the reinforcement or cancellation of particular frequencies caused by the inevitable modes. Avoid floor-plan shapes that include circles and all regular polygons (i.e., squares, pentagons, etc.), since these will produce “modal pile-up”: groups of modes at or near the same frequency.

Controlling modal effects can be accomplished by proper selection, placement, and installation of the correct type of acoustical absorption. The absorption must be effective at the low frequencies where the modes exist. Helmholtz absorbers and other bass traps are often used for this purpose.

Table 1: These are the nine lowest modal frequencies for two different rooms.

4 m × 3 m × 3 m Room	33 m × 20 m × 13 m Room
43.1	5.2
57.4	8.6
57.4	10.1
71.8	10.4
71.8	13.2
81.2	13.5
86.1	14.2
91.9	15.7
103.5	15.8

In venues for critical listening, especially auditoria, several other important acoustical criteria should be considered. Most of these involve helpful early sound reflections that reach the listener's ears within 50–80 ms of the original sound. They can be very important perceptually.

Intimacy

Intimacy refers to the sensation that the talker, singer, or instrument(s) is located physically close to the listener. Intimacy is important for speech and chamber music. Intimacy depends on the initial time-delay gap, the time that elapses between the original sound, and the first reflection. Although intimacy is usually considered in connection with music venues, I have been involved in the design of at least one gymnasium in which the client wanted an intimate sound. Probably familiarity with television and pod-type music players has made our society accustomed to much more intimacy than would be experienced in most natural venues.

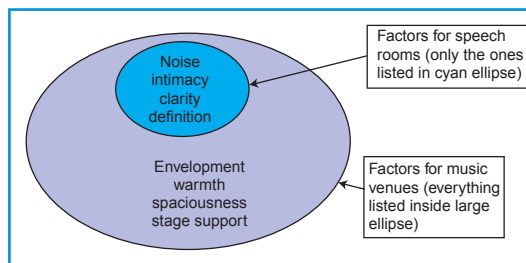
Envelopment

Envelopment is the sense of being surrounded by the music, and is important for almost any type of musical performance. It depends mainly upon the presence of diffuse reflections from the side and rear of the audience. Diffuse reflections are spread out in time, unlike the coherent reflections that are perceived as echoes.

In addition to the need for side and rear diffuse reflections, envelopment requires that the sound striking the left and right ears be dissimilar. A measure of the similarity of left-ear and right-ear sound is called the inter-aural cross correlation (IACC). Lower values of IACC (less similarity between left and right) give improved envelopment and are generally preferred. Envelopment is most important for organ, symphonic and choral music; less so for piano, small solo instrument and solo voice; probably detrimental for speech.

Warmth

Performance spaces are sometimes described as "brittle," "thin," "full," or "warm." These terms describe the room's timbre, and technically are considered as descriptions of the property called warmth. In small rooms, such as recording studios, the warmth is controlled by the modal frequencies. In larger spaces, warmth depends primarily on the way in which RT60 varies with frequency. Typically, for most acoustical music, the RT60 in the octave containing middle "C" (261.6 Hz) can be about 10%–15% longer than the mid-frequency



RT60 (an octave above middle "C") and the RT60 of the octave below middle "C" can be about 25% longer than the mid-frequency value. More variation than this can make the room sound too warm or boomy. Less variation can make it sound thin.

Spaciousness

Spaciousness, sometimes called the "apparent source width" or "sound stage width," is the perception that the sound source is widely distributed across the front of the room, as opposed to a sense of constricted width. Spaciousness depends on the prevalence of sidewall (lateral) reflections that are heard within the first 80 ms after the original sound. Thus, the factors that contribute to spaciousness also promote envelopment.

Clarity and Definition

Clarity and definition are closely related, in that both are quantified by the ratio of early sound reflections to total sound. Clarity specifically refers to reflections from all directions arriving within 80 ms of the original sound and is used to assess music halls. Definition refers to reflections coming from all directions, but arriving within 50 ms of the original sound, and is often used to predict speech intelligibility.

Ensemble and Stage Support

In any musical venue, it is essential that the individual performers be able to hear themselves and each other well, and in proper balance. The acoustical piece of the puzzle involves reflective surfaces that bounce the sound of the individual back to him or her, while also bouncing the sound off the other performers back to each individual as well. These beneficial stage reflections must come from surfaces close enough to the performers that the reflections arrive within an interval of 35–50 ms. Otherwise, the sound is perceived as an echo that can confuse the timing and synchronism of the performance.

For groups with just a few members, monitor speakers can substitute for stage support. However, providing stage support electronically for large groups is only technologically possible if in-ear monitors are used, so acoustical means are often preferred. **Figure 5** shows which acoustical factors are important for which type of venue. 

Figure 5: Certain specific acoustical factors are important for each type of venue.

Acoustical Society of America Meeting

The North Carolina Acoustical Society of America's fall general meeting will be held in the Wingate Hotel in Cary, NC, on November 15, 2013. The meeting will feature the Royster Competition in which graduate students in acoustics participate for scholarships. The students' work must involve noise and/or hearing conservation, and will include presentations of papers as well as a poster session. Additional papers will be presented as well, covering a broad spectrum of acoustical topics. The program is yet to be finalized.

Attendance is open to anyone, whether a member of the ASA or not. The registration fee is \$40 and includes lunch. A late registration penalty will be added after November 8, 2013. Special room rates are available at the Wingate Hotel for attendees. For more information, visit, www.nc-asa.org.

A 12-V Trigger Supply

By
Gary Galo

(United States)



Photo 1: The trigger supply is built in a Hammond 1455-series enclosure. The black anodizing and plastic end bezels give the enclosure a professional appearance.

This simple project provides a convenient method to remotely power audio equipment. Many manufacturers make audio equipment and power products that can be remotely switched with a low DC trigger voltage. PS Audio's power products, including the Power Plant AC regenerators and the Quintet line conditioner, can be turned on remotely with 5 to 15 VDC. A 12-VDC trigger can directly switch on many NAD and Parasound audio products. Preamps made by both companies will supply a 12-V trigger to their power amps, but if you're using another manufacturer's preamp you may be out of luck.

It's convenient to be able to turn your entire system on with one switch. But, power amplifiers are high-current devices and should never be plugged into a switched power outlet on the back of a preamp. Plug them into their own line conditioners, AC regenerators, or directly into an AC wall outlet.

I use three PS Audio Power Plant Premier AC regenerators, one for my low-level equipment and one for each mono power amp, each fed by a dedicated AC line.^[1] With the trigger supply, my entire system can be turned on and off with one low-current switch (see **Photo 1**).

Supply Circuit

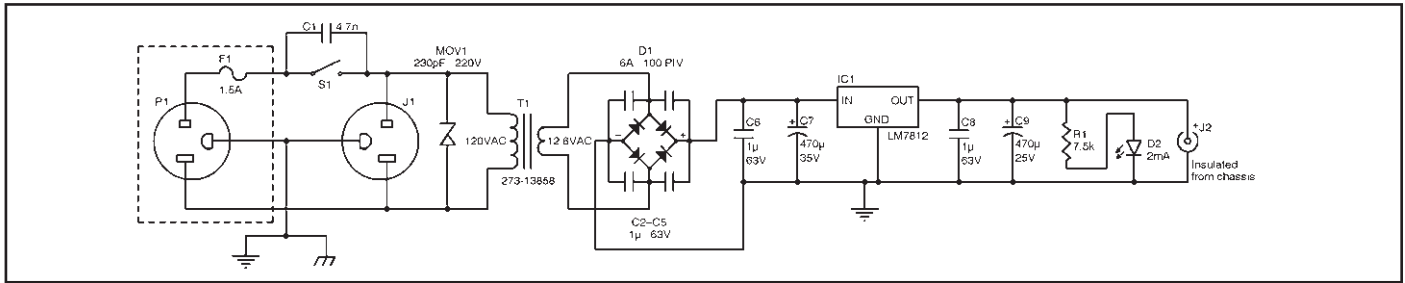
Figure 1 shows a simple 12-V trigger supply that uses a Kobiconn AC Inlet with an International Electrotechnical Commission (IEC) AC connector and a built-in 5-mm × 20-mm fuse holder. I included a Kobiconn AC outlet right after the power switch to trigger older line conditioning equipment (e.g., the Adcom ACE-515), which requires a 120-VAC trigger voltage. This AC outlet is only intended for low-current triggering applications. Don't attempt to power any audio equipment directly with this outlet. I included an MOV for modest protection against power surges and lightning.

The power transformer can be purchased at any electronics parts store. It also has a 12-VAC/300-mA secondary. I used a full-wave bridge rectifier—6 A is obviously overkill, but it's the lowest current rating you can normally find on an encapsulated bridge. The three-terminal IC regulator is a 7812 type with a TO-220 case. Many semiconductor manufacturers make these—the brand isn't critical and a heatsink is not necessary. The input and output electrolytics are bypassed with 1-μF film caps. Use a 35-V electrolytic on the input side, and 25 V on the output.

Some builders may consider the film bypasses overkill in a supply strictly intended for trigger application, but this is a sensible practice in any power supply and the cost is minimal. The same is true of the four snubber caps across the rectifier diodes. The 2-mA LED and series resistor form the pilot lamp. The output connector, J2, is a standard DC power connector. Normally these have three terminals: chassis, sleeve (supply negative), and tip (supply positive). I recommend not connecting supply ground to the chassis connector—use the sleeve connection only.

Hammond Enclosure

For decades, audio builders and amateur radio operators have relied on Hammond Manufacturing



as a prime source for high-quality enclosures and transformers. For this project, the Hammond 1455-series enclosures seemed a natural choice. These nicely finished, extruded aluminum enclosures are available in a clear- or black-anodized finish, with either aluminum or plastic end pieces. The enclosures have U-shaped bodies and slide-in bellies that can function as the top or the bottom plate, depending on your personal preference and project layout. The enclosures are supplied with a pair of black ABS plastic bezels for mounting the end pieces, but their use is optional.

Hammond includes all the hardware and four adhesive rubber feet. It sells replacement end pieces separately, which is nice if you make a mistake or have second thoughts on the layout. The 1455N2201BK model—with a black anodized finish measuring 8.66" x 4.06" x 2.09"—is just right for this project.

PS Audio specifies a range of 5 to 15 VDC for its AC regenerators and line conditioners. The voltage requirements may vary with products from other manufacturers. Although 12 V is fine for most products activated with a DC trigger, you can easily adjust the output voltage by substituting a different 78xx-series regulator and power transformer. The transformer secondary AC RMS voltage should be the same as the regulated DC output.

The PS Audio Power Plant products and the Quintet line conditioner have 3.5 mm (0.125") mini-phone trigger input jacks. You will need a cable to get from the trigger supply's output to the triggered device's input. RadioShack sells a 6' patch cord with a 0.125" plug on one end and stripped wires on the other. These ends can be soldered to P2, the DC power plug (P2 is not shown in the schematic). Although each PS Audio power product has a pair of jacks, they are not feed-through—both are trigger inputs. If you want to daisy-chain more than one unit, you'll need the 6" Y-adapter and 0.125" cables. Since the trigger inputs on these products draw a minuscule amount of current, several can be daisy-chained together. Most Parasound products use 2.5-mm sub-mini jacks. There's an input jack and a feed-through loop for daisy-chaining several Parasound products, so there's no need for a Y-adapter. Use a cable with a 2.5-mm (0.094") plug for the Parasound products.

Construction

I used the Hammond enclosure's underside as the bottom plate (see **Photo 2a**). The power transformer, the rectifier bridge, the capacitors and the three-terminal IC regulator are mounted on a small grid-style prototyping board. Don't rely on the transformer's solder terminals to hold the transformer in place. Use a pair of 4-40 machine screws to secure the transformer to the board. Use insulated standoffs to mount the board to the bottom plate.

Mount the IEC-type AC power inlet, AC outlet, and DC power jack on the rear panel (see **Photo 2b**). Mouser's website (www.mouser.com)

Figure 1: The 12-V trigger supply's complete circuit is shown. The output voltage can be easily adapted for equipment requiring other trigger voltages. A switched AC outlet is included for older line conditioners (e.g., Adcom's ACE-515).

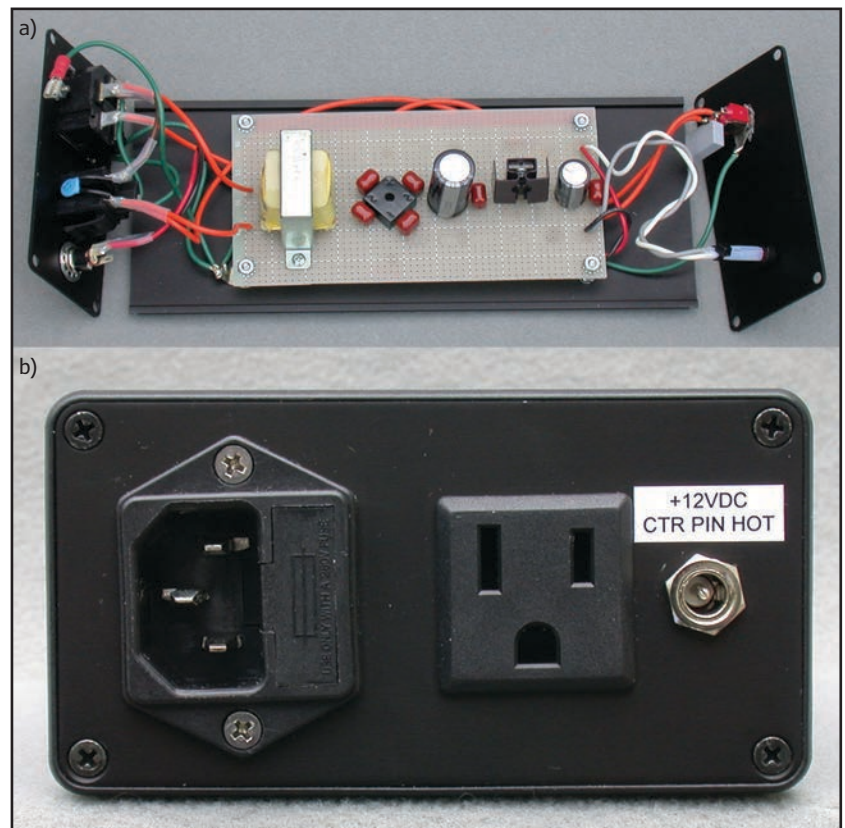


Photo 2: The trigger supply's inside view shows the power transformer, the rectifier bridge, the filter capacitors, and the IC regulator, which are mounted on a small prototyping board. A ground wire fitted with a push-on connector (visible on the left) facilitates a solid ground connection to the U-shaped body (a). The AC inlet, the AC outlet, and the 12-VDC outlet are shown in the trigger supply's rear view. Cutouts for the AC inlet and outlet were made according to the dimensions supplied on the Kobiconn datasheets (b).



Photo 3: The Hammond 1455-series enclosure's U-shaped body uses a ground lug which, when mated with a slip-on connector from the main chassis, will ensure a proper ground connection. The plastic bezels shown at the bottom are supplied with the enclosure, but their use is optional.

has datasheets for the Kobiconn power receptacles available as downloadable PDF files. The datasheets include panel cutout dimensions. MOV1 can be soldered directly to the AC outlet's terminals using insulated sleeving on the leads. Mount the

main power switch S1 and the pilot LED on the front panel with capacitor C1 soldered directly to S1's terminals. I recommend using heat-shrink tubing on all soldered terminals where 120-VAC line voltage is present.

One of the problems with modular, anodized enclosures (from any manufacturer) is that the anodized finish is an insulator, making electrical connection of the various chassis pieces a hit-or-miss proposition. I always ensure all the pieces have chassis ground potential by installing a chassis ground lug on each piece and connecting them with soldered ground wires.

The ground lugs can share mounting holes with one of the AC Inlet mounting screws on the rear panel, one of the prototyping board mounting screws on the chassis's underside, and the AC power switch. (You'll need one with a 0.25" mounting hole for the power switch. Mouser has a large assortment of ground lugs.) Grind away the anodizing to ensure a good connection to the chassis. A Dremel-style hobby tool (mine is actually a Sears Craftsman) with a small grinding wheel works well. The U-shaped body can also be fitted with a ground lug which, when used with a push-on connector, facilitates ground connection with the rest of the enclosure, and enables easy removal, if necessary (see **Photo 3**). After assembling the supply, stick the supplied adhesive rubber feet to the bottom of the enclosure.

Parts List

Capacitors

- C1 = 4-n7/250-VAC Interference suppression capacitor (Vishay), Mouser 75-F17102471000
- C2-C6, C8 = 1- μ F/63-V stacked metal film capacitor (Panasonic), Digi-Key P4548-MD
- C7 = 470- μ F/35-V aluminum electrolytic capacitor (Nichicon), Mouser 647-UKT1V471MPD
- C9 = 470- μ F/25-V aluminum electrolytic capacitor (Nichicon) Mouser 647-UKT1E471MPD

Hardware

- Hammond enclosure, Mouser 546-1455N2201BK
- 5-mm LED holder, RadioShack 276-079
- Ground lugs, Mouser (various)
- Insulated standoffs, Mouser (various)
- Hookup wire
- Grid-style prototyping board, RadioShack 276-147
- Heat-shrink tubing, RadioShack 278-1610
- 18-AWG detachable IEC power cord, RadioShack 61-205
- 6' shielded cable, 0.125" plug to stripped wires, RadioShack 42-2434 (Mates with J2; substitute as necessary)
- 6" Mono Y-cable, RadioShack 42-2542 (Optional, see text)
- 6' shielded cable, 0.125" plug to 0.125" plug, RadioShack 42-242 (optional, see text)

Miscellaneous

- P1 = IEC-Type AC Inlet with 5-mm $\times$ 20-mm fuse holder (Kobiconn),

Mouser 161-R3013-E

- F1 = 1.5-A/250-V fast-acting 5-mm $\times$ 20-mm fuse, RadioShack 270-1050
- J1 = Snap-in NEMA AC outlet (Kobiconn), Mouser 161-RU0200-E
- J2 = 5.5-mm O.D./2.1-mm I.D. DC power jack, RadioShack 274-1582
- P2 = 5.5-mm/2.1-mm DC power plug (mate for J2), RadioShack 274-1569
- S1 = SPDT toggle switch, Digi-Key EG2367-ND
- T1 = Power transformer, 120-VAC primary, 12-VAC, 300-mA secondary, RadioShack 273-1385B
- MOV1 = 230 pF/220 V (Bourns), Mouser 652-MOV-07D221K

Regulator

- IC1 = 7812, 12-V, 1.5-A three-terminal IC regulator (NJR), Mouser 513-NJM7812FA

Resistor

- R1 = 7.5 k/0.25 W (Yageo), Digi-Key 7.50KXBK-ND

Semiconductors

- D1 = 6-A/100-PIV Rectifier bridge, Digi-Key PBPC602DI-ND
- D2 = 2-mA red LED, Digi-Key 516-1325-ND

Sources

- Digi-Key | www.digikey.com
- Mouser | www.mouser.com
- RadioShack | www.radioshack.com (and local stores nationwide)

Testing

Once the supply is finished, it is ready for testing. Before connecting the AC to the unit, ensure the wide contacts on the AC inlet and outlet are connected to neutral and the narrow contacts are hot (see **Figure 1**).

Turn the main power switch off and install a 1.5-A, fast-blow, 20-mm fuse in the Kobiconn AC inlet. Then, connect an IEC power cord to the AC inlet and insert a standard AC outlet tester (available at any hardware store) into the AC outlet. Plug the other end of the power cord into a live AC outlet and use the main power switch to turn the unit on. Check the outlet tester's lights to ensure the AC outlet is properly wired. Then, use a digital multimeter (DMM) to check the DC voltage at the IC regulator's output and the DC power connector's center pin. It should read 12 V in each location. The front panel LED should also be illuminated.

If the unit passes all the tests, turn it off, disconnect the power cord, and assemble the enclosure. Slide the enclosure into the slots on the U-shaped body. Slip the plastic bezels over the end pieces and onto the enclosure's ends. Use the supplied screws to fasten the end pieces and

About the Author

Gary Galo retired in October 2010 after 34 years as an Audio Engineer at the Crane School of Music, SUNY, in Potsdam, NY. Not ready to quit entirely, he works half-time in that capacity. He is the author of more than 260 articles and reviews on both musical and technical subjects. He is an active member of the Association for Recorded Sound Collections (ARSC) and has been a regular presenter at ARSC conferences. Several of his conference papers have been published in the *ARSC Journal*.

bezels to the chassis. The trigger supply is ready to use. The Hammond 1455-series enclosure looks professional and should make an attractive companion to the rest of your audio components. 

Reference

[1] G. Galo, "Keep It Clean," *Multi-Media Manufacturer*, July–August 2010, and *audioXpress*, January 2011.

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1455-Series enclosure
Hammond Manufacturing Co., Inc. | www.hammondmfg.com

Power Plant AC regenerators
PS Audio | www.psaudio.com

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- Develops an understanding, from a theoretical and practical view, of the typical materials used or suggested for use in the construction of transducers
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Tips to Resurrect a Classic Speaker or Design a New System (Part 1)

By
Thomas Perazella
(United States)

Use a Heil Transformer and an Updated Woofer



Photo 1: The Eton 7" drive has a light rigid cone made of a Nomex honeycomb sandwiched between two Kevlar layers for true pistonic behavior.

This article is the first of a three-part series that will describe the history, construction, testing, problem solving, and voicing of an old speaker brought back to life by new technology. Although the original goal was to breathe new life into a classic, many of the issues addressed here are also critical when designing a new speaker or even getting the best performance from commercial speakers and associated equipment.

When it comes to musical taste, you could probably classify me as an omnivore. Depending on my mood, I may lean toward classical, jazz, rock, pop, country, or other genres. Like most of us, there is one special piece that goes directly into my heart, bypassing all logic to create an intense emotional involvement. For me, it is Gustav Mahler's Symphony No. 2, "The Resurrection." It is also a fitting title for my latest completed project.

What is an AMT?

Several years ago, I became interested in the Heil Air Motion Transformer technology. Created by Dr. Oskar Heil, this device uses an unorthodox mechanism to quickly accelerate air, resulting in a very interesting tweeter. There is a lot written about the actual mechanism, including several patents leading up to the AMT version (see Resources).

But there is also a lot of misinformation on how it really works. It has been said that the diaphragm, which is a vertically aligned array of pleated material behaves like an accordion squeezing the air in and out. Actually, the whole diaphragm does not move in and out like an accordion, rather alternate folds move either in or out so the air is squeezed out one side and sucked in the other. The folds stop the diaphragm from moving back and forth so the result is a dipole pressure wave. For a better understanding of this device, please refer to the Resources section.

Heil AMT History

The AMT was originally produced by the California company ESS, but the final complete speaker received mixed reviews. It was a two-way design that suffered from running the AMT at too low a frequency and the 12" woofer at too high a frequency. The two drivers' radiation patterns were different because of their physical shapes and

the wavelengths reproduced vs. the size of each throughout their crossover frequency ranges. Also, one was used in open air as a dipole and the other in a box as a monopole. The AMT's crossover point and slopes caused it to operate at the low end in a range that did not work with the given slopes. The resulting speaker did not live up to the AMT's true performance capabilities. In spite of this, people loved the speaker so much it almost became a cult classic because of the AMT's crystalline response.

I do not remember exactly when I decided to use the AMT as the basis of a DIY project, but it was probably in the late 1980s or early 1990s. I obtained a used pair of the tweeters and immediately knew I would have to use a three-way configuration if I was going to be able to get a wide enough dynamic range without exposing the tweeter to undue stress. I would also have a chance to better control the integration of the drivers' radiation patterns.

Although a three-driver three-way is a common configuration, some severe compromises in a one-driver-per-frequency-range configuration must be addressed. Primarily, they have to do with sensitivity, linear displacement capability, and power requirements, which I recalled during the implementation. It also reminded me why I moved on to other solutions. However, the concept has validity because of its smaller size and lower cost.

My Original Project

The speaker concept used a sealed box holding a dynamic driver for the bass, a dipole mounted dynamic driver for the midrange, and the AMT as a dipole for the high frequencies. I located a woofer box and grille used in an exceptionally well-performing speaker, the Acarian Alon IV. It was designed for a sealed-box woofer and a dipole-mounted midrange and tweeter (see Resources for more information). I then had to choose the bass and midrange drivers.

When I designed the original speaker, the German company Eton had developed a woofer/midrange driver that used what it called "Hexacone" technology. The cone was made from a Nomex honeycomb sandwiched between two Kevlar layers (see **Photo 1**). The result was a cone that was very light but extremely rigid. Within a certain range, it acted like the ideal piston.

In addition to the sandwich cone, the Eton woofer/driver had an inverted soft rubber surround. The result was a relatively smooth, low-distortion output, even when driven hard. There were some ripples in the passband and if you went up in frequency, an anomaly in the frequency response

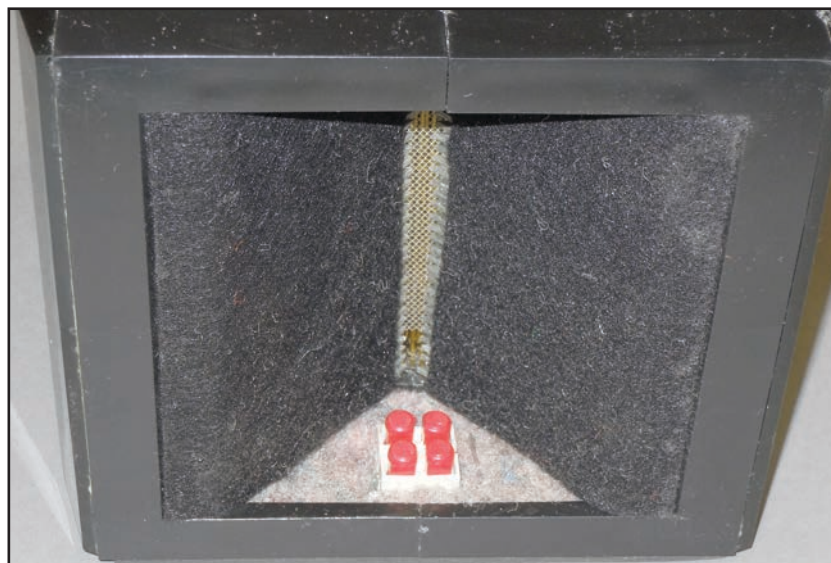


Photo 2: The cavities in the Heil AMT housing were filled with deadening material in an attempt to reduce cavity resonances.



Photo 3: The baffle housing the midrange and tweeter is isolated from the bass cabinet by a rubber sheet and rubber tubing in the holes that receive the indexing pins.

would appear. I have been told that the ragged response at higher frequencies was due to a difference in impedance between the rigid cone and the compliant surround that resulted in reflections of energy back into the cone at the higher frequencies. I cannot verify the actual mechanism myself, but response problems do appear.

The solution to using a driver that has great performance in most of its range but problems at frequency extremes is simple. Don't use the driver in the problem areas. That may be problematic when using passive crossovers, but with the flexible electronic crossovers now available, it has become a non-issue. At the time, I decided to use the Eton woofer/driver with a home-built 12-dB/octave electronic crossover. I mounted it in a quasi-dipole configuration with an open back but heavy damping. As it turns out, that was okay, but not the best solution.

The original AMT driver was mounted in a plastic housing that had a ridge around the front and back that created triangular-shaped cavities at the top and bottom in the front and rear of the radiating element. Although I had no way to measure those cavities' effects, I altered them by filling the cavities with wool carpet underlayment

Photo 4: The original speaker used an HSU 12" bass driver, which is shown without the grille.



material. In addition, I fastened some thin black felt material over the faces of the pole pieces to improve the tweeter's looks and possibly reduce some reflected energy (see **Photo 2**). It is interesting to note that the AMT's current version has the cavities removed from the housing and has added what appears to be a black surface-finish treatment on the pole pieces.

Both the AMT and the Eton drivers were mounted on a flat baffle that sat on top of the woofer box in positions to time align the three drivers. I added mass to the baffle base in the form of a box containing cement. I used sand-filled steel tubes to brace the baffle's upright section to the base. The baffle was located on the woofer housing with steel pins fastened to the housing and slip fitted into holes in the baffle's base. I used surgical rubber tubing inserts to isolate the base from any vibration that may have come up the pins. The baffle's base also had a rubber sheet on the bottom for isolation. **Photo 3** shows the layout of the baffle's base.

For the bass range, I originally used 12" drivers that I already had. However, they did not have much linear excursion, so I switched to 10" Peerless drivers. They were better, but the additional excursion hardly made up for the loss in cone area. Ultimately I used 12" drivers from HSU Research that had a higher X_{MAX} and a larger cone area. **Photo 4** shows the completed speaker.

I used those speakers for many years with good results. However, as time went on, I realized they had limitations, including limited bass excursion and a radiation pattern that resulted in more floor and ceiling reflections than I preferred. Ultimately, I replaced them with my current main system that uses Bohlender-Graebener RD75 drivers for the high frequencies, Peerless 831727 10" drivers for the upper bass/lower midrange, and sealed box

subwoofers. (More information about these drivers and midranges is available in my article series, "On Angels Wings," Part 1 and 2, *audioXpress*, January–February, 2001 and "A Dipole Midbass," Part 1 and 2, *audioXpress*, June–July 2004.) The Heil-based speakers were put into storage. With the AMT tweeters available to the public once again, I decided to see if I could improve on the original design, especially since a range of powerful reasonably priced digital signal processing (DSP) equipment is available.

Updating the Design

Although the AMT and Eton drivers have undergone revisions over the years, both versions I had are still adequate for the job so I did not replace them. Woofer designs, however, have progressed dramatically in the intervening time. The search was on to find a suitable substitute.

A major advantage of current speaker control devices is that trying to achieve the woofer's critical tuning in an enclosure is not as much a necessity as in the past. For example, if you use a passive crossover with an incorrect speaker parameters and box volume combination, you may wind up with a Q_b that is too high. The result would be a bump in the frequency response at resonance. In the past, that could result in an overly bass-heavy voicing. Most DSP-based electronic crossovers enable you to create a bandpass filter with any center frequency, Q , and any amplitude you desire to flatten out the response bump. Because of this, if you already have an enclosure as I did or are restricted by severe space limitations, your driver choices are greatly expanded. You do not have to worry about achieving critical damping by choosing from a limited range of drivers with the correct parameters for the given box size.

The Woofer

When it comes to speakers with drivers covering separate frequency ranges, the woofer is the group's heavy lifter. For each octave the reproduced frequency decreases, four times the volume displacement of air is necessary to maintain the same sound pressure level (SPL). If you need "X" amount of volume displacement to achieve your desired SPL at 20 kHz, at 20 Hz you need more than 4 million times "X" to achieve the same SPL. Fortunately, music does not produce the same SPLs at the highest frequencies as in the bass, but it quickly becomes apparent why so few speakers can reproduce high-amplitude organ pedal notes at realistic listening levels without excessive distortion, if even at all.



Photo 5: The Dayton Ultimax UM12-22 woofer has several great features, including 19-mm X_{MAX} .

To achieve the high linear volume displacements necessary for clean bass, you need a lot of radiating surface combined with a lot of linear excursion. It is similar to an internal combustion engine where displacement is a product of the cylinder bore times the piston stroke times the number of cylinders. If the engine needs more displacement, you must have some combination of more pistons, bigger pistons, or a longer stroke for the pistons.

In this speaker's case, the "bore" was limited to a 12" driver and the number of "cylinders" or woofers in each speaker was one. To get the volume up to a reasonable level, a high linear "stroke" or excursion was needed.

The specification for a dynamic driver's maximum linear one-way excursion is X_{MAX} . It is the excursion where either the motor force factor (Bl) falls to 50% of its peak value or the suspension stiffness (K_{MS}) increases to double the initial value. X_{MAX} is defined as the direction of travel either in or out that has the worst of either of the limiting values.

For example, if a driver is limited to 5-mm travel in one direction by force factor reduction and is limited in the other direction by increased stiffness at 4 mm, the driver's X_{MAX} will be 4 mm. Unfortunately, some manufacturers quote maximum total excursion in both directions regardless of either parameter's nonlinearity as the excursion instead of true X_{MAX} . *Caveat emptor!*

In previous woofer designs, I used drivers from various companies that had different linear excursion vs. price combinations. Some have incredibly high linear volume displacements in small drivers but at a high price. Over the years, my best results for drivers with high linear displacement at reasonable prices have come from Dayton Audio, the house brand of Parts Express. My current reference system uses two Dayton 15" DVC woofers that provide yeoman service for a two-driver arrangement.

Looking at the Parts Express website, the sub-woofer section brought me to a relatively new series of drivers called "Ultimax." This series consists of three drivers in 10", 12", and 15" diameters. They all have similar features, which can be viewed on the website. One with a 19-mm X_{MAX} (which is huge for a driver in this price range) was really impressive. I decided to use the Dayton Audio UM12-22 12" driver. It was not so many years ago that drivers with these specifications would have been unheard of, let alone affordably priced. Anyone who refers to "the good old days" obviously didn't build speakers. I purchased two units (see **Photo 5**).

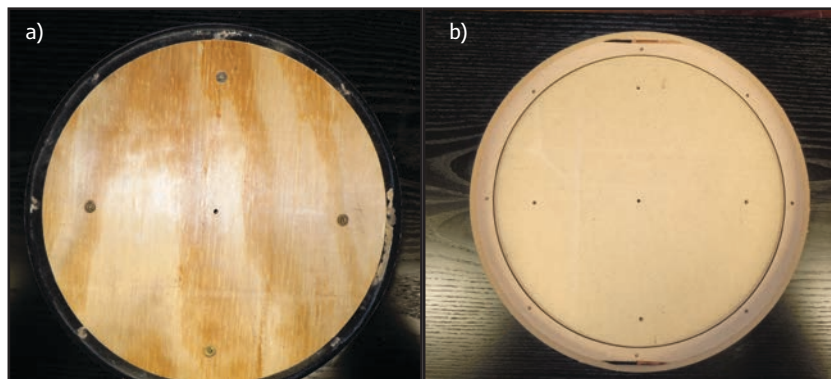


Photo 6: A pivot plate was constructed and inserted into the original woofer mounting hole to provide the alignment point necessary for the Jasper jig and router to properly enlarge the hole (a). The enlarged hole in the bass cabinet is shown with the backer plate and pivot hole before routing of the lower basket clearance hole (b).

Mounting the Woofers

After receiving the new woofers, I removed the old ones from the enclosures to see what modifications were needed to fit the new drivers. The new drivers turned out to be 0.25" larger in outside diameter than the old ones, so I had to enlarge the original mounting holes to accept the baskets. In addition, the mounting flanges and surrounds were a bit thicker. To provide clearance for the grilles and give it a more finished look, I recessed the drivers further into the housing.

This combination presented two challenges. The first was that I no longer had a center hole to use with my Jasper jig and router to enlarge the hole. The second was that, even though the enclosure's front panel was 1" thick, the amount of recess I desired would leave too little material after routing to securely fasten the drivers.

To solve these problems, I decided to use a backing plate mounted to the inside of the enclosures. First, I cut out two pieces of 0.75" MDF to mount inside the enclosures. The pieces covered the existing holes and extended far enough to provide the necessary strength to mount the drivers. They also acted as a support for the router I used to enlarge the hole's diameter. Two pieces were cut to fit inside the enclosure at the front.

A small problem cropped up when I tried to insert the backers. The enclosure's walls had additional 0.5" thick MDF plates with deadening material fastened to the inside to reduce vibrations. Because of the backers' size, the edges of the plates attached to the sides interfered with the backers. I removed, trimmed, and set them aside to be refastened inside the housing after all the work on the backers was completed. Then, I temporarily screwed the backers into place against the front plate.

Next, I fabricated a temporary pivot plate to fill the existing hole flush with the enclosure front.

About the Author

Thomas Perazella is a retired IT director. He received a BS from the University of California, Berkeley campus. He is a member of the Audio Engineering Society, the Boston Audio Society, and the DC Audio DIY group. He has written for *Speaker Builder* and *audioXpress* magazines. He has authored several articles in professional audio journals and taught commercial lighting at the Winona School of Photography.

Photo 7: The woofer terminals were connected to the terminal plate using 12-gauge zip cord. A 12-gauge jumper was used to tie the two voice coils together in series.



This plate enabled the router to smoothly slide over the enclosure and provide the necessary pivot hole for the Jasper jig. I used a scrap piece of plywood for the plate that was not quite thick enough to present a totally flush surface by itself. I used several temporary shims of the proper thickness on top of the backer to solve the problem. I screwed the plate to the backer with four flat head wood

screws and I drilled not only through the plate but on through the backer to make a pilot hole for the jig. This assured me that the enclosure's newly routed outside clearance hole would be exactly in line with the inside driver frame clearance hole that I later routed (see **Photo 6a**).

Then, I set the proper spacing on the Jasper jig and routed the new mounting hole in the enclosure. Once the hole was finished, I removed the pivot plate and checked its fit with the driver. Note that the backer plate has the four holes that were used to fasten the pivot plate and also the pivot hole that was then used to route the clearance hole in the backer. Because the new driver needed extra depth, the additional routing opened two small holes into the original enclosure's inside (see **Photo 6b**). Although they would be covered by the driver frame and gasket, I decided to use the wood filler to ensure the enclosure was sealed.

Next, I removed the backers and routed the clearance holes through which the rear of the driver baskets would fit. When that was done, I glued the backers and screwed them to the enclosure's inside using the same holes to ensure alignment. I replaced the damping plates. Then I masked, primed, and painted the front of the enclosure.

To maximize the enclosure's apparent inside volume, I stuffed it with the fiber material that had been used in the original speaker. Next I wired the drivers using 12-gauge zip cord and a single piece of 12-gauge wire as a jumper between the two voice coils. To achieve 4-Ω impedance, the two 2-Ω coils on each driver were wired in series with one coil's negative terminal wired to the other's positive terminal. I wired the remaining positive and negative terminals from each of the two coils to the input terminals on the enclosure (see **Photo 7**). I used McFeely's #8 black-oxide square drive wood screws to fasten the two woofers. I have been using square drive screws in critical applications for years as they eliminate the "camming" that often happens with Phillips screws. McFeely's has a good selection. The results provided a secure mounting for the woofers and, with the additional depth to the recess, a clean look. Next I moved on to the midrange and treble sections.

I will cover the remainder of the construction phase plus some problem solving in Part 2 of this article series. I will continue with problem solving plus voicing and music correction curves in the final article in this series. 

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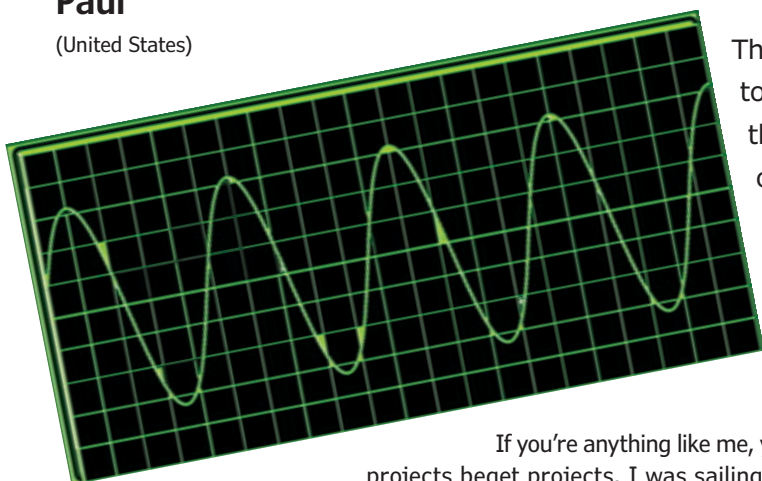
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Simple Sineman

A Sine Wave Generator for the Rest of Us

By
Christopher Paul

(United States)



This article presents a design for a sine wave generator that achieves a THD of better than 130 dB below the fundamental output. Designs that obtain 140 dB or better performances are referenced, but these are noticeably more complex.

If you're anything like me, you find that projects beget projects. I was sailing along, putting the finishing touches on a comparison of the performances of several single-stage triode voltage and current gain amplifiers. All I needed to do was perform some distortion measurements. I had my spectrum analyzer ready. How hard could it be to whip up a high-quality sine wave generator?

Always looking for the opportunity to do something new, I built a 1-Mbps, 1-bit DAC followed by some modest analog filtering. But I could only get the THD to around -85 dB or so. Maybe I'll go back to it someday. Finally, I had to look at the literature so I could steal, uh, benefit from, what others had achieved. There is some great stuff out there about oscillators in general. (See Resources for more information.) If you're looking for the simplest possible circuit to build, there is one that consists of an op-amp, four resistors, two capacitors, and a Christmas tree light bulb(!). It reportedly has "second- and third-harmonic distortion products lower than -120 dBc."^[1]

More complex designs are available that use multiple op-amps, references, transistors, diodes, semiconductors, and several more capacitors and resistors. (See Resources for more information.) These circuits claim performances ranging from -120-dBc THD to an eye-popping better than -140 dBc. But they were all a bit too complex for my taste. I really wanted to design one for myself. Then I found a fairly simple design with "only" 0.01% (-80-dB) THD, published more than 35 years ago by veteran audio designer and *audioXpress* author Walt Jung ("Gain Control IC for Audio Signal Processing," *Ham Radio*, 1977). I thought this basic design had the potential for some significant but simple improvements.

Build an Oscillator

To make an oscillator, you just build a circuit that, only at the oscillation frequency, has a stable loop gain of exact unity and a phase shift of 0° or a multiple of 360°. **Figure 1** shows the preferred method of the more complex referenced designs. C1, C2, R1, R2, and U1a form a band-pass filter. The filter is centered at:

$$\frac{1}{\sqrt{(4\pi^2 \times R1 \times R2 \times C1 \times C2)}} \text{ Hz}$$

where the phase shift is 180°. If the resistors and capacitors have the same values, then the gain of the Rvar1, R3, R4, R5, and U1b network must be -2 for oscillation to occur. In his article, "Low Distortion Oscillator Tests Measurement Circuits," (*EDN Network*, 2012), Vojtěch Janásek claims that a better choice from a noise point of view is to make C2 = 2 × C1 and R1 = 2 × R2, in which case the remaining network must have a gain of -1. In either case, the control element must perform such that higher output signal levels increase and lower ones decrease

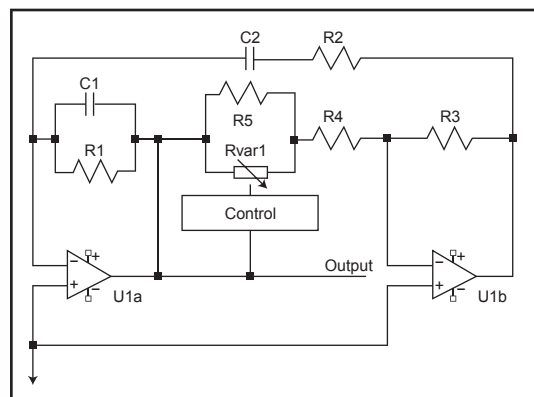


Figure 1: This is a block diagram of a more complex oscillator design.

Rvar1's value to drive the gain around the loop to unity, thus stabilizing it as a condition of oscillation.

Because Rvar1 and its control form a nonlinear element, they introduce distortion into the circuit. Part of the strategy is to ensure that the amount of signal through Rvar1 is as small as possible in comparison to that through R5. Often this leads to adjustment with a potentiometer (not shown) to ensure that the range of achievable Rvar1 values will include one that establishes unity loop gain. (This potentiometer can also become a convenient means of establishing the oscillation signal level.)

The other part of the strategy is to ensure that the control signal is clean, eliminating as much as possible any vestiges of the oscillation signal that could modulate Rvar1. It is this demand on the control circuit's performance that causes most of the complexity in these designs.

The One IC Oscillator

One thing that is so interesting about Jung's circuit shown in **Figure 2** is that it uses only one IC (!) to perform the functions provided by Rvar1, control, and the op-amps shown in **Figure 1**. The heart of this circuit is that IC, ON Semiconductor's SA571 compandor. **Figure 3** shows a block diagram of the SA571. Jung's article discusses other applications of the SA571 and provides a more detailed description of how the part functions. And yes, this 35-year-old IC is still manufactured by ON Semiconductor, not by Phillips, which bought the original manufacturer, Signetics.

Referring to **Figure 3**, the part of the SA571 that performs Rvar1's function is called the ΔG cell. The "control" block is the rectifier, whose output is filtered by a grounded capacitor connected to the rectifier capacitor. And the op-amp is, well, an op-amp, that works with an R-C network shown in **Figure 2**, which is different from that shown in **Figure 1**.

The generator of **Figure 2** functions as follows. Infinite attenuation at a notch frequency of f_0 Hz occurs when:

$$R1 \times R2 \times (C1 \times C2 + C2 \times C3 + C1 \times C3) = \frac{1}{(2\pi \times f_0)^2}$$

and:

$$RV = [R1 \times (C1 + C2) + R2 \times (C2 + C3)] \times \left[\frac{1}{\frac{C1 + 1}{C2 + 1} \times 3} \right]$$

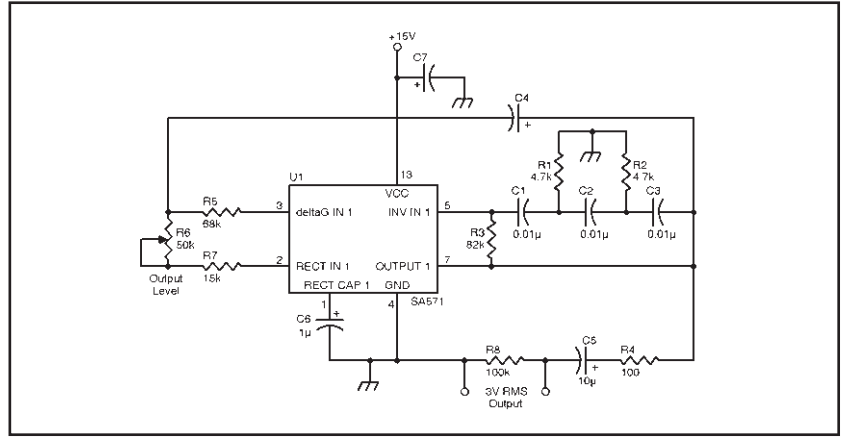


Figure 2: Walt Jung's schematic shows a one IC sine wave oscillator (Image courtesy of Walt Jung)

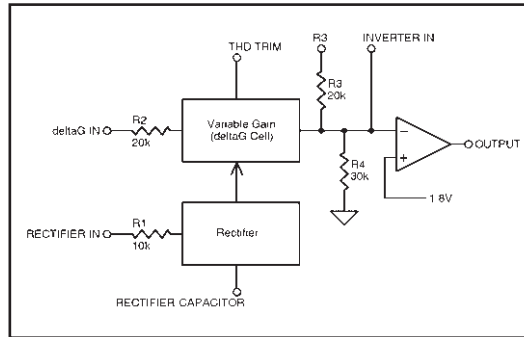


Figure 3: The SA571 compandor IC is shown in a block diagram.

These requirements simplify if $C1 = C2 = C3 = C$ and $R2 = R1 = R$, as shown in **Figure 2**. Now:

$$R \times C = \frac{1}{(\pi \times f_0 \times \sqrt{12})}$$

A variable resistor Rvar (the ΔG cell) within the SA571, in series with R5, is connected in parallel with resistor R3 to form the total resistance R_v . R_v , R1, R2, and C1–C3 form a notch filter. At some frequency slightly below the notch, when R_v is slightly larger than $12 \times R1$, this network exhibits a relatively slowly changing phase shift in the vicinity of -90° along with a relatively rapidly changing large amount of attenuation.

The notch network is connected between the internal op-amp's output and inverting input. The op-amp and the network form a loop. Already, over a broad range of frequencies, the op-amp supplies a -180° phase shift (because the signal goes through its inverting input) plus -90° (because of its internal compensation capacitor). Its gain is large and its gain and phase change more slowly with frequency than the network's do.

Because the loop has two degrees of freedom—the oscillation frequency and the Rvar's value—a point can always be found to satisfy the two conditions (gain and phase) for oscillation. The point is stable because if the output tries to grow, R_v

Figure 4: The ΔG cell input and distorted portion of its output, the AC part of the filtered rectifier output, and the rectifier output before filtering are shown with signals *in phase* at the ΔG cell and rectifier inputs.

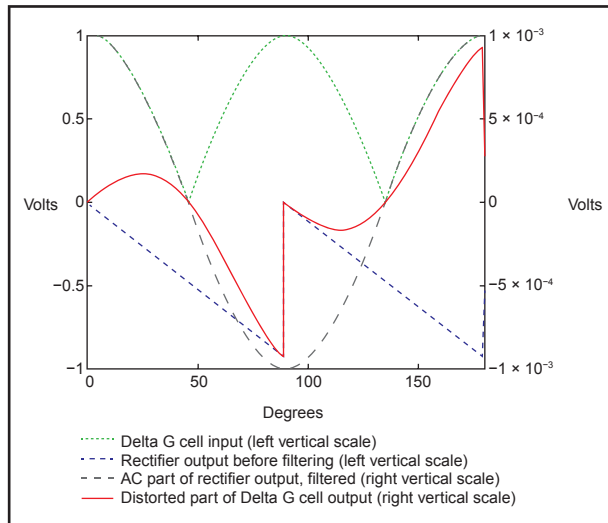
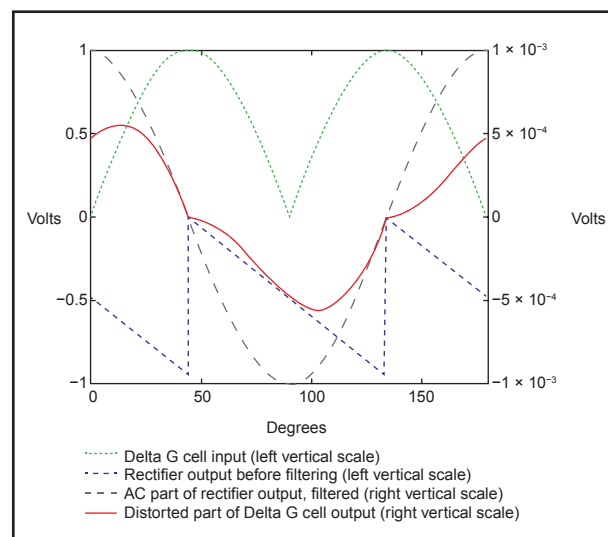


Figure 5: The ΔG cell input and distorted portion of its output, the AC part of the filtered rectifier output, and the rectifier output before filtering are shown with signals *90° out of phase* at the ΔG cell and rectifier inputs.



will decrease and loop gain will fall to counteract the effect. And if it tries to drop in level, R_v will increase, again producing a compensatory effect.

Understanding the constraints on R_v , how is R_3 chosen? The main requirement is that R_3 is greater than $12 \times R$ so the variable resistor in the IC can reduce R_v to just above $12R$. In this circuit, $12R = 12 \times 4.7k = 56.4k$. So the value of 82K used for R_3 fits the bill.

However, the circuit gives a THD that is only 80 dB below the fundamental. This is not very impressive compared to the performances of the other referenced circuits. Obviously, there is some work to do.

Revving Up the Original

One of the simplest ways to improve performance is to increase the C_6 's value, filtering out more of the control signal's AC portion, which generates modulation distortion at the ΔG cell output. You may think this value could increase without limit. But the higher the value, the longer the circuit takes to stabilize.

There is another strategy I mentioned to reduce distortion: Minimize the signal that goes through R_{var} (the ΔG cell in this case) in comparison to that which goes through R_v (R_5 plus the ΔG cell in parallel with R_3) as a whole. This can be down by reducing R_3 to a value only slightly larger than $12 \times R_1$. Now R_{var} must be very large, reducing the current through the ΔG cell. To make R_{var} large, increase the values of R_5 and $R_6 + R_7$. In the process, R_6 is eliminated (replaced with a short). Now, regardless of the signal at the op-amp output (which drives the ΔG and rectifier inputs), a small ΔG signal is ensured. To regain control over the output level, break R_3 up into a fixed resistor R_{3a} and a variable resistor (made from a potentiometer) R_{3b} . The bulk of the resistance is in R_{3a} . Because the ΔG contribution is so small, R_{3b} affords a sensitive adjustment. It's best to use a 10-turn potentiometer here.

But I'm not quite finished with R_7 yet. It's worth

Parts List

Capacitors

- C1, C2, C3, C101, C102, C103 = polystyrene, 0.01- μF , 160-V, axial 1%
- C4, C8, C9, C11 = ceramic, 0.1- μF , 50-V, axial, X7R type
- C5 = 330- μF , 25-V, aluminum electrolytic
- C6, C7, C10 = 22- μF , 25-V, aluminum electrolytic
- C7r = 100- μF , ceramic axial, NPO type

Miscellaneous

- SKT1 = IC socket, DIP package, 16 pin
- SKT2 = IC socket, DIP package, eight pin
- CLP1, CLP2, CLP3, CLP4 = 9-V battery clip
- Bt1, Bt2 = 9-V alkaline battery
- LED1, LED2 = LED, red, T1 package (leaded)
- Breadboard with groundplane

Regulator

- IC1 = Compador
- IC2 = Dual audio op-amp

Resistor

- R101 = 3,740- Ω , 1% 0.125-W metal film resistor, axial
- R1, R2, R102 = 3,830- Ω , 1% 0.125-W metal film resistor, axial
- R3a, R103 = 43.2-k Ω , 1% 0.125-W metal film resistor, axial
- R3b = trimmer, potentiometer, 5 k Ω , 20 turn, through hole
- R104 = trimmer, potentiometer, 200 Ω , one turn, through hole
- R105 = trimmer potentiometer, 5 k Ω , one turn, through hole
- R5 = 681-k Ω , 1% 0.125-W metal film resistor, axial
- R8 = 100- Ω , 1% 0.125-W metal film resistor, axial
- R9 = 4,990- Ω , 1% 0.125-W metal film resistor, axial
- R10 = trimmer, potentiometer, 20 k Ω , 25 turn, through hole
- R11 = 1,000- Ω , 1% 0.125-W metal film resistor, axial

taking a look at and understanding the relationship between the waveforms of the rectifier and the ΔG cell. **Figure 4** shows the ΔG cell input in phase with the (as yet unfiltered) rectifier output signal, the AC portion of the filtered rectifier output (the signal across C6), and the product of the first and last of these signals (the distorted part of the ΔG cell output), which finds its way into the oscillator output. The sharp change in the control signal coincides with the ΔG cell input signal's peaks, which it multiplies, yielding gross distortion.

Fortunately, a simple change can significantly improve this situation. If I replace R7 with a capacitor whose impedance at the oscillation frequency is much larger than that of the rectifier's built-in 10k input resistor, I will have shifted the phase of the rectifier's AC output signal by almost 90°.

Figure 5 shows how much the product signal is cleaned up when the sharp changes in the rectifier signal are shifted to coincide with the ΔG input's zero-crossing portions rather than its peaks. A harmonic analysis of the distortion signals shows a 10-dB reduction in the third-harmonic levels and even greater improvements in the higher odd harmonics. (This distortion in all cases generates minimal even harmonics. Those present in the oscillator output are due to other mechanisms.) Let's call this new capacitor C7r (since it replaced R7) and connect it directly to the op-amp output. C4 now drives only the ΔG cell input. C7r should be approximately:

$$\frac{1}{(2 \pi R5 f)}$$

where f is the oscillation frequency in Hertz.

The op-amp within the SA571 is not a high-quality unit. But it can be ignored and a higher quality external op-amp can be connected in its place. The bad news is that in doing so I am doubling the number of ICs in the design! Such is the price of progress. These days, I believe the gold standard for low-distortion op-amps is the Texas Instruments (TI) LME49710 op-amp. The LME49710 has a ± 15 -V supply voltage, and a 3-to 5- V_{RMS} output level into a 600- Ω or more load. It claims a typical THD+N of -134 dB (!).

When I started the Simple Sineman design, I thought I might need two op-amps, so I used TI's dual LME49720. The LME49720 has a slightly poorer rating of about -131 dB. So you should use the LME49710 if you build this circuit. As with any op-amp, you should decouple its power supply pins to ground. I used 0.1- μF ceramic capacitors.

The SA571's datasheet points out there is one more thing that can be done to minimize distortion.

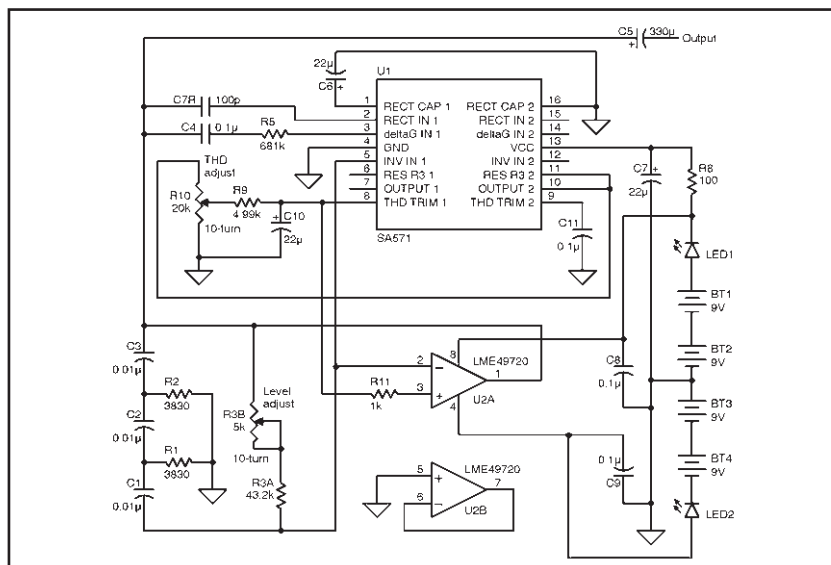


Figure 6: The Simple Sineman circuit is shown.

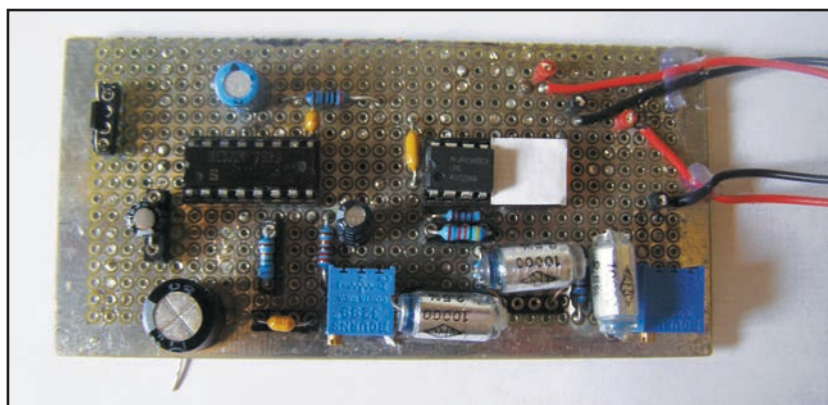


Photo 1: The Simple Sineman breadboard seemed to be the best development platform. The batteries, battery clips, and LEDs are not shown.

Pin 8 is a THD trim pin. You can carefully adjust the DC current flowing into or out of this pin while watching the oscillator output's second-harmonic distortion on your spectrum analyzer. To create this adjustable current source, you need a stable voltage reference to drive the top of a ground-referenced potentiometer. The reference must be higher than the 1.8 V present on Pin 8. A series resistor connected from the potentiometer arm to Pin 8 and a capacitor from Pin 8 to ground are needed to complete the adjustable current source.

This is what's so great about the SA571: It's a dual unit, intended for stereo. If on the second unit, I don't feed AC signals into the ΔG and Rect inputs, I ground the Crect pin and tie the IC's R3 and output pins together to have a stable 3-VDC source. Just add a capacitor to decouple the second THD adjust pin to ground.

Unlike the LME49720 op-amp, which has a better than 110-dB power supply rejection ratio (PSRR) on each of its rails, the SA571's PSRR is not specified.

About the Author

Christopher Paul is an electrical engineer who has written articles for *The Audio Amateur*, *audioXpress*, and several other electronics industry magazines.

Harmonic Number	Decibels
2	-11.81 dB
3	-6.54 dB
4	-5.14 dB
5	-3.78 dB
7	-2.78 dB
9	-2.49 dB

Table 1: This shows the notch filter's frequency response.

Harmonic Number	Level, Decibel V_{RMS}
2	-136.0
3	-134.2
4	-148.0
5	-141.3
7	-151.7
9	-149.4

Table 2: The oscillator harmonic signal levels are recorded at the notch filter output.

Harmonic Number	Decibels
2	-133.7
3	-137.2
4	-152.4
5	-147.0
7	-158.4
9	-156.4

Table 3: Oscillator harmonic signal levels are shown with respect to their fundamentals.

I thought it best to use an R-C network to decouple its positive supply from that of the LME49720, on which significant fundamental and harmonics of the oscillation can be seen.

Regarding the power supplies, I chose batteries for mine. I used four 9-V batteries to provide a quiet and portable supply. Since a new 9-V battery is typically 9.6 V, and the two ICs' absolute maximum voltage ratings are 18 V and ± 18 V, I had to knock down these 19.2-V (maximum) supplies. What better way than with two 1.9-V red LEDs to produce a 17.3-V maximum? I added a switch and had a visual power-on indicator.

The last "rev up" is to ensure that the capacitors chosen contribute negligible distortion. I have a few concerns about the qualities of Cr7 and C6. The former is in the rectifier input path, where the signal is going to be purposely distorted through full-wave rectification and the latter has a distorted signal imposed upon it. I selected C5 to be a high-voltage, high-capacity electrolytic, so there is a negligible voltage drop across it when loads are attached. The small drop ensures that minimal distortion is incurred.

Although C4 carries a signal that will eventually find its way directly into the oscillator output, the signal is so small that any distortion it adds is swamped by the residual distortion in the rest of the circuit. I can make it a small and inexpensive ceramic.

High-quality film capacitors are used for C1, C2, and C3. This is critical. I've tried both polystyrene and metalized polypropylene. Polystyrene provides the best results. I used capacitors that I had laying around for this test. Because polystyrenes can be difficult to find, I have provided a source for them in the parts list.

Figure 6 shows the up-revving results. R1, R2, R3a, R3b, C1, C2, and C3 were selected for a 2,400-Hz frequency. It should be possible to divide the values of C1, C2, and C3 by F and change the frequency of oscillation by a factor of F.

Construction

When I started this effort, I had a number of ideas I wanted to explore. I was uncertain not only about component values, but also about circuit topologies. I decided the best platform for development was a breadboard. While I didn't believe layout was critical, I did want to have a ground plane. The breadboard I selected, while a bit expensive, provided that option. (I have since discovered that RadioShack stocks some acceptable, less expensive alternatives.) **Photo 1** shows the Simple Sineman as a working model (minus batteries, battery clips, and LEDs).

Measurement

Of course, you'll need a spectrum analyzer. Some of these can reliably distinguish signals that have dynamic ranges of as much as 140 dB, but I don't have access to one. My Stanford Research SR770 can manage an 85-dB spurious-free range. So I resorted to putting the signal generator output through a notch filter to significantly attenuate the fundamental and reduce the dynamic range of the signals I needed to measure. A suitable notch filter can be made from the R1, R2, R3a, R3b, C1, C2, and C3 shown in **Figure 6**. But this time, R3b can be a one-turn potentiometer.

Replace R1 with a 3,740- Ω resistor and put it in series with a 200- Ω , one-turn potentiometer. This filter's input and output are its R3b-C3 and R3a-C1 junctions. It's essentially symmetrical, so its input and output can be swapped. The two adjustable resistors are repetitively tweaked to achieve the fundamental's maximum attenuation. In the parts list, I've added a value of 100 to reference designators listed here to cover parts for the filter. R104 is the 200- Ω potentiometer mentioned earlier.

Power up the oscillator, place a load of 10 k Ω or greater on its output (600- Ω loads significantly increase distortion) and adjust R3b for a 3-VRMS (9.5-dB VRMS) output level. This is in the range of lowest THD values for the LM49710/20 op-amps powered from ± 15 V.

It will take several seconds for the signal level to stabilize. Note the frequency to four decimal places. Why? Sometimes the analyzer will show spurious signals from the "aether" (power line harmonics, etc.) that have nothing to do with the oscillator. You'll only know by their frequencies that they are not one of the harmonics. Put the signal through the notch filter and adjust the filter's two variable resistors for the minimal fundamental level at the filter output. You should be able to easily get 60 dB of fundamental attenuation.

Now you can measure and record the notch filter characteristics. Almost any variable frequency sinusoidal generator with an output impedance of 50 Ω or less will do. Connect the generator to the filter input. Use your spectrum analyzer or decibel meter to record the filter output in decibels minus its input in decibels for each of the harmonics you want to measure. **Table 1** shows my filter's gains (less than unity, so they are negative) at the harmonics I wanted to measure.

With the filter calibrated, you're ready to measure distortion. Connect the oscillator to the filter input and the filter output to the spectrum analyzer. If the analyzer has a feature that automatically adjusts the spectrum analyzer's gain so that

the signal it is analyzing is just below the level where the analyzer adds distortion, now is the time to use it. To accurately measure the harmonics, I set the span of my analyzer to about 50 Hz and centered it on the frequency I calculated for each harmonic. When looking at the second harmonic, adjust R10 for the minimum level. Record the levels of all the harmonics you want to measure. **Table 2** shows my measurement results. (Some of the even harmonics were too low to measure.)

To determine the levels of the harmonics below the fundamental, start with the harmonic level at the oscillator output in decibels. For each harmonic, add the (negative) "gains" from **Table 1** and subtract the signal levels from **Table 2**. My results are in **Table 3**. The THD for the harmonics I measured can be determined by taking

each of the **Table 3** harmonic levels h (in decibels), calculating $10^{h/10}$, adding them to arrive at a sum S , and calculating $THD = 10 \times \log_{10}(S) \text{ dB} = (100 \times \sqrt{S}) \%$. My results are $-131.9 \text{ dB} = 0.000025\% \text{ THD}$.

Overall Impression

Several designs are available for ultra-low distortion sine wave generators. I think that the Simple Sineman design occupies a "sweet spot" between simplicity and performance, but you can be the judge of that. 

Author Acknowledgment: I'd like to thank Walt Jung for the helpful comments he provided from his review of this article.

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Resources

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67WR5KLF Trimming potentiometer
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3296P-1-203LF Trimming potentiometer
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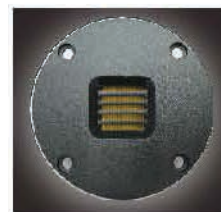
SR770 FFT Spectrum analyzer
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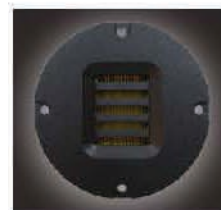
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Classic Approaches to Dynamics Processing

By
**Richard
Honeycutt**
(United States)

Fundamentally, there are two ways to build a dynamics processor: by using a voltage-controlled attenuator or by using a voltage-controlled amplifier (VCA). In either case, the control voltage is derived from the audio signal via a rectifier and an integrating filter. For a compressor, the gain decreases as the control voltage increases once the signal exceeds the threshold.

For an expander, the gain increases as the control voltage increases above the threshold. **Figure 1** shows the operation of a voltage-controlled attenuator and a voltage-controlled amplifier (VCA). The two differ in performance in that the amplifier provides gain and the voltage-controlled attenuator does not.

Voltage-Controlled Attenuator

A voltage-controlled attenuator is just a voltage divider, one arm of which changes resistance according to an applied control voltage. This can be accomplished by using a triode, a pentode, or a light-dependent resistor (LDR). Since the 1970s, field-effect transistors (FETs) have also been used for this purpose.

Figure 2 shows a rectifier and LDR configured for use in a compressor. It makes little difference whether the rectifier is solid or hollow state. To use the circuit (shown in **Figure 2**) in an expander, the positions of the fixed resistor and the LDR could be interchanged, or the rectifier diode's polarity could be reversed.

In this article, I'll focus on downward compressor circuits. The operating principles of expanders are the same; only the control voltage's polarity is reversed. The amount of compression can be controlled by adding a potentiometer between the input and the rectifier diode. Depending on the

light source in the LDR, an amplifier stage and/or current-limiting resistor may be needed between the rectifier and the LDR input.

The use of triodes in a voltage-controlled attenuator is based on the fact that a triode's plate resistance depends on the grid bias voltage, changing from a low value at normal bias voltages to a high value as the bias approaches the cutoff value. **Figure 3** shows a triode used in a voltage-controlled attenuator. Essentially it works like the LDR-based voltage-controlled attenuator, except the triode takes the LDR's place. Depending on the type of lighting device used in an LDR, the triode's attack time can be faster. On the other hand, a triode is more expensive.

The simplest LDR-based compressor uses a small incandescent lamp as the LDR. The amplifier feeding the lamp must be able to supply enough current to drive the lamp: typically about 40 mA. Fed by voltages ranging from 0 to 4 V, a nominal 0.04-A metal-filament lamp will have a resistance of about 10 Ω cold (0 V), increasing to 80 Ω hot (4 V).

Build a VCA

The two most common ways to build a hollow-state VCA are to use a triode or to use a pentode or pentagrid tube. **Figure 4** shows the effect of grid bias voltage on the gain of a 12AX7 triode amplifier stage. By applying a control voltage derived from a

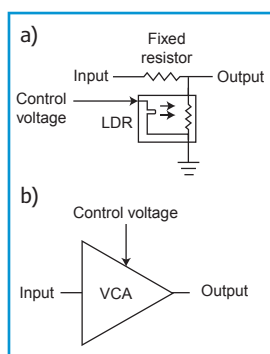


Figure 1: Either a voltage-controlled attenuator (a) or a VCA (b) can vary signal level in accordance with a control voltage.

rectifier, you can vary the gain of an amplifier stage: this is a VCA. **Figure 5** shows a triode VCA stage's circuit. Pentodes can be similarly used.

The maximum output is about 34-V peak. For this circuit, about 13.5 dB of gain control is possible by varying the control-grid bias between -1 and -5 V. Additional control could be provided by cascading additional stages. The Fairchild model 670 compressor used this principle, incorporating a quad of triodes connected in parallel push-pull.

A pentode's gain can be varied by applying a control voltage to the control grid, as described for the triode. In the case of a pentode, the remote-cutoff version is usually used. A remote-cutoff pentode uses a control grid structure that provides more linear control of plate current as grid bias voltage is varied. These have also been referred to as "variable- μ tubes." **Figure 6** shows the difference between a sharp-cutoff and a remote-cutoff tube.

Figure 7 shows an RCA MI-10234C compressor amplifier's circuit. A similar circuit was used in the Westrex RA-1593A compressor amplifier. The remote-cutoff pentodes are connected in push-pull to reduce even-order harmonic distortion.

For the circuit shown in **Figure 7**, V1 and V2 are 6K7 remote-cutoff pentodes.^[1] Their cathodes are connected together and their combined current flows to ground through a single 2.2-k Ω resistor. The voltage drop across that resistor sets the voltage of the cathodes and the suppressor grids. A 100-k Ω balancing potentiometer helps minimize distortion. A compression metering circuit is fed via the plate supply for the two 6K7s. The output from the push-pull VCA stage feeds two 6J6 amplifier stages. The signal is fed from the output transformer primary's high side through a two-stage 6SN7 amplifier (the "side amplifier"). It drives the 6H6 full-wave rectifier, which provides the control voltage through the input transformer secondary's center tap. (Note that signal flow in the side amplifier is from right to left, since it performs something like a feedback function.)

In this way, the 6K7's control grids are fed the input signal added to the DC control voltage. Since the cathodes have a small positive bias from the 2.2-k Ω resistor mentioned earlier, the control grids are somewhat negative with respect to the cathodes when no signal is present. With a signal applied, the negative control voltage biases the control grids more negative, in proportion to the signal level. (Notice that the control voltage is taken from the full-wave rectifier's anodes, making the control voltage negative.)

If two of these circuits were used for stereo

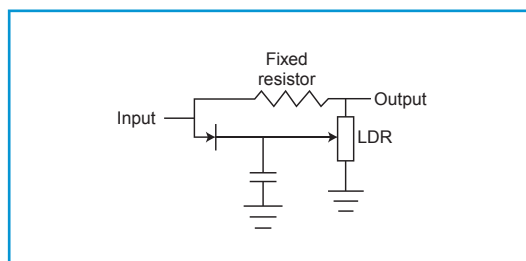


Figure 2: A rectifier feeding an LDR provides compression.

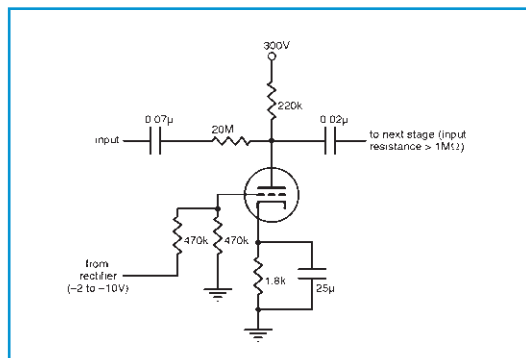


Figure 3: A triode can serve as a voltage-controlled attenuator.

operation, the same control voltage would be used for both the right- and left-channel compressors, a process called "side-chaining." Side-chaining can also be used on microphones for individual musical instruments, to identically control all instruments' compression.

Pentagrid Tubes

Pentagrid tubes were originally developed to make it possible to combine the oscillator and mixer stages of superheterodyne radio receivers. Pentagrids are heptode tubes, with five grids. Grid 1 serves as part of the oscillator circuit. Grids 2 and 4 are connected together, often inside the tube envelope, with grid 2 acting as the oscillator's screen grid and anode. In some schematics (e.g., the one shown in **Figure 7**), only one grid is shown for both grid 2 and grid 4, since they share an external connection. Grid 3 receives the RF signal. Grid 4 and grid 5 act as the screen and suppressor grids for the mixer stage, respectively.

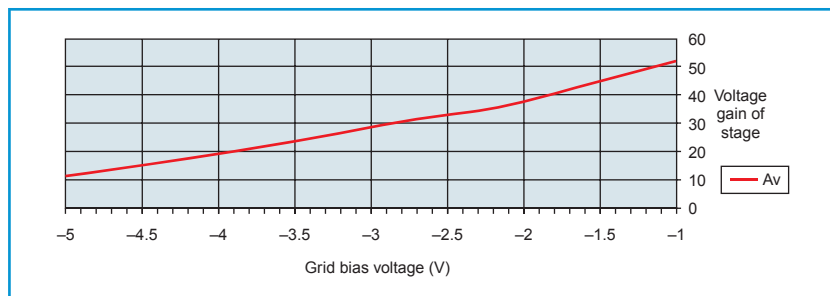


Figure 4: Biasing the control grid less negative increases the triode stage's gain.

Figure 5: This triode stage can provide about 13.5 dB of compression.

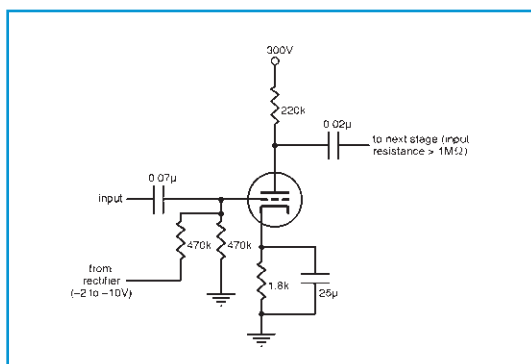


Figure 6: A remote-cutoff pentode has a less sharp change in plate current as grid voltage approaches cutoff.

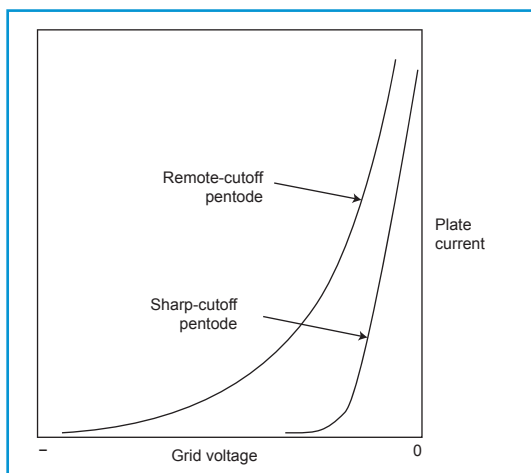


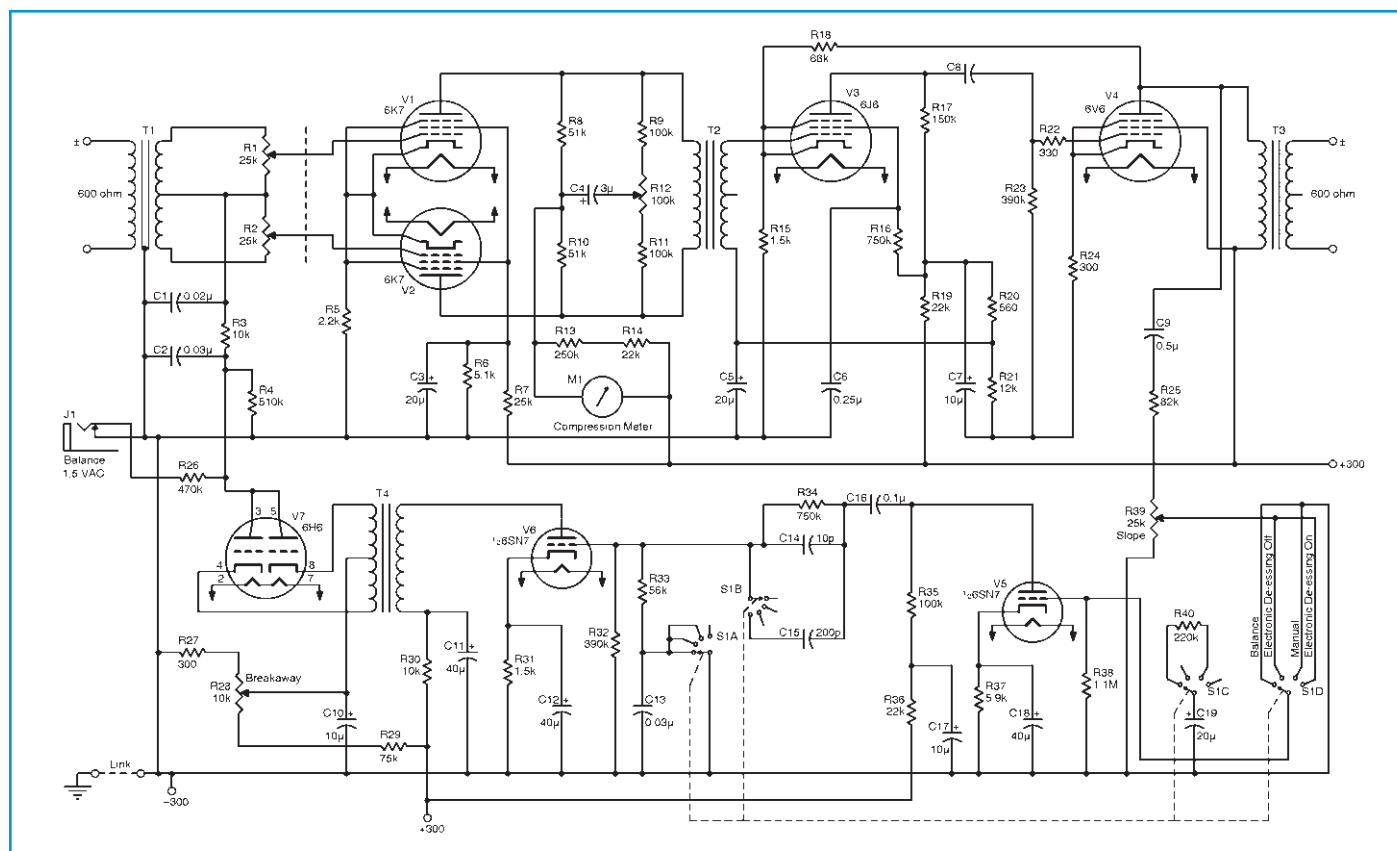
Figure 7: This RCA compressor amplifier uses remote-cutoff pentodes in its VCA.

When pentagrids are used in compressors, the signal is usually applied to grid 1, and the control voltage is connected to grid 4. **Figure 8** shows a compressor using a pentagrid.^[2]

Compressor Controls

Compressors can include a variety of controls including: input level, output level, threshold, slope or compression ratio, attack time, release time, and de-essing. An input-level control, if present, is simply a potentiometer located between the input connector and the first stage of amplification. This can be linear amplification or a VCA. The input-level control's setting affects the amount of compression that occurs. When you increase the signal level, the rectifier ultimately puts out more control voltage, and the VCA turns down the gain.

Normally, the output level control does not affect the amount of compression, but a few compressors have been manufactured in which this is not true. Thus, the input and output level controls can interact with the compression-ratio control, making it imperative that the operator thoroughly understand the equipment if predictable results are to be obtained.



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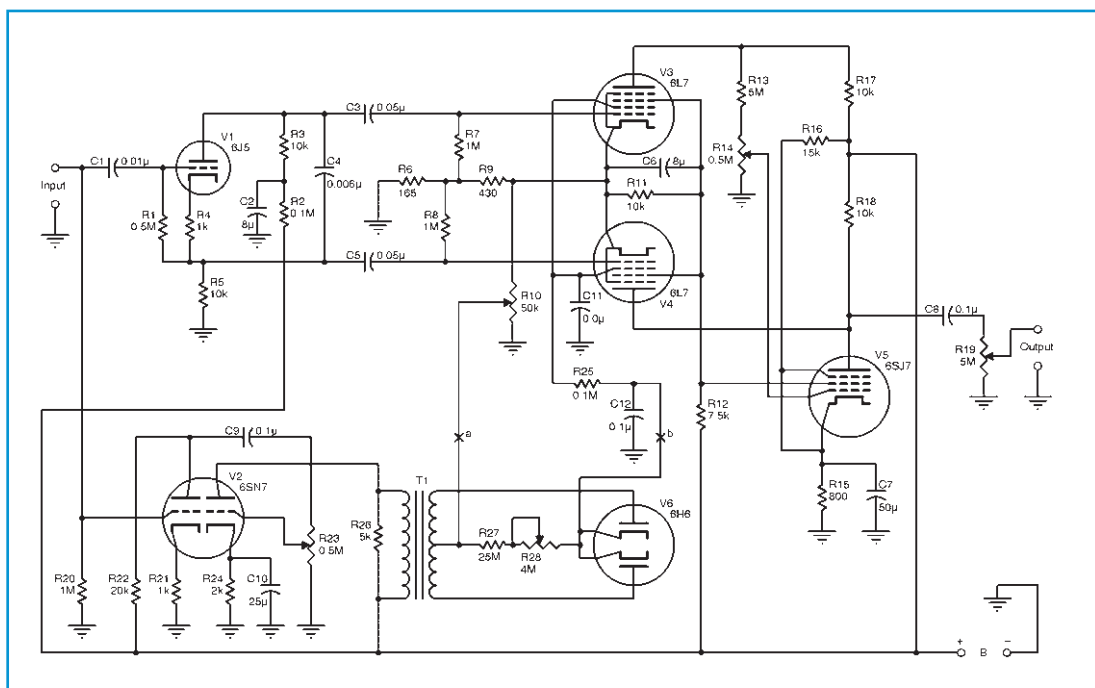
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Figure 8: The VCA in this compressor uses 6L7 pentagrid tubes.



The threshold control affects the “back bias” applied to the rectifier diodes so no appreciable control voltage is produced until the signal level exceeds a certain voltage. The slope or compression ratio control determines the side amplifier’s gain. More gain provides a higher compression ratio.

Attack time is determined by the filter capacitor’s time constant at the rectifier’s output, and the total equivalent charging resistance in series with the capacitor as a signal peak causes it to charge. Release time is determined by that same capacitor value and by the equivalent resistance in series as the capacitor discharges. In general, the charging and discharging resistances are not equal. This is because the charging resistance includes equivalent resistances on the rectifier’s input side, but the rectifying action blocks those resistances from the discharging current.

De-essers

When an audio signal is processed by a downward compressor, trailing sibilants (e.g., the “ss” in “hiss”) have such a low level that the compressor often releases while these sibilant sounds are occur-

ring. Thus, less compression is applied to the trailing sibilant than to the rest of the word. The result is that such sibilants—as well as fricatives (“f” sounds, etc.) are overemphasized in the output signal from the compressor. This makes for unnatural-sounding speech, singing, and music performance.

Simply using an equalizer to reduce the amplitude of the high-frequency sounds is not the answer, since the problem is not constant with time. The answer is to use a compressor that only operates on the high frequencies. Such a device is called a “de-esser,” which makes a lot of sense, considering what it does. De-essers have found use in audio for broadcasting, recording, and motion pictures. Some compressors (e.g., the RCA model whose circuit is shown in **Figure 7**) incorporate de-essers. In **Figure 7**, the circuitry involving the two switches below T3 controls the de-essing operation.

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Member Profile



Member Name: **Bruce Heran**

Location: Sierra Vista, AZ

Education: Bruce has a BS in Biology and he has taken some master's levels courses.

Member Status: He has been reading *audioXpress* for a number of years.

Occupation: Bruce co-owns and is vice president of design for Oddwatt Audio. The company specializes in high-quality audio kits (mostly valve designs). Prior to that, he was a project manager for a government defense contractor.

Audio Interests: Bruce is a fan of anything analog. He enjoys designing high-end valve equipment and listening to music on vinyl.

Most Recent

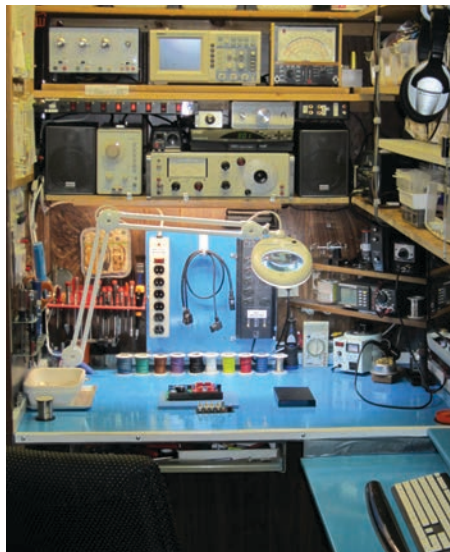
Purchase: Because he is willing to design and build almost anything, Bruce never knows what he might buy. However, his most recent purchase was an Audio Technica AT-33PTG/II phonographic cartridge.

Current Audio

Purchases: Bruce is working on a digital remote control integrated

valve preamplifier with a valve phono stage. It also has a solid-state phono preamplifier and an economy integrated valve stereo amplifier.

Dream System: Bruce's dream system contains Martin Logan Montis speakers (he currently owns the Vistas); a pair of Martin Logan Grotto or Depth subwoofers;



KT120 mono block power amplifiers of his own design (he already has them); a remote-control pre-amplifier of his own design with phono stage (he already has this, too); an OPPO BDP105 Blu-ray player (to replace his OPPO 83SE); a Sota Millennium Turntable (to replace the Sota Moonbeam he currently owns); and several phonograph cartridges.



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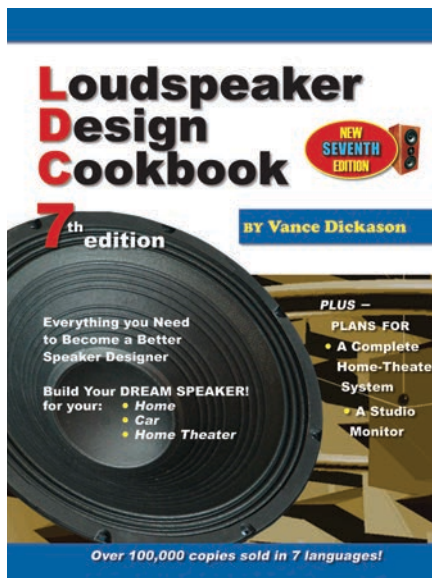
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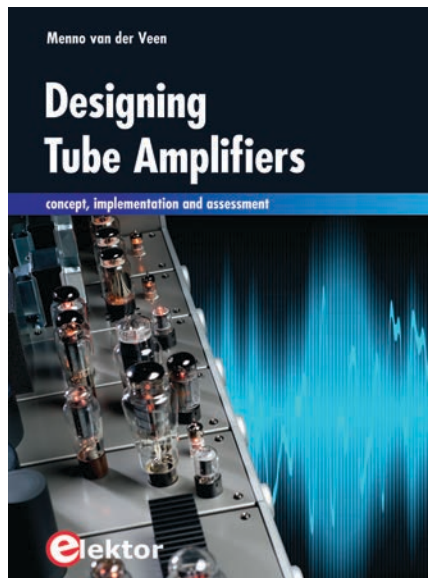
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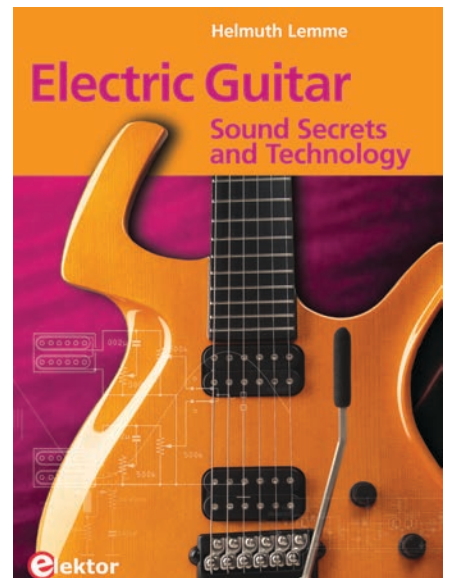
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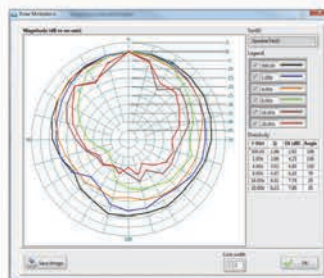
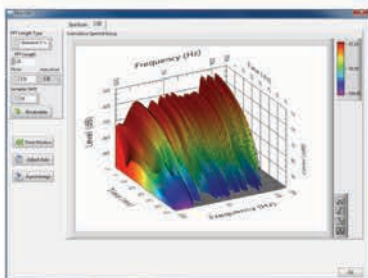
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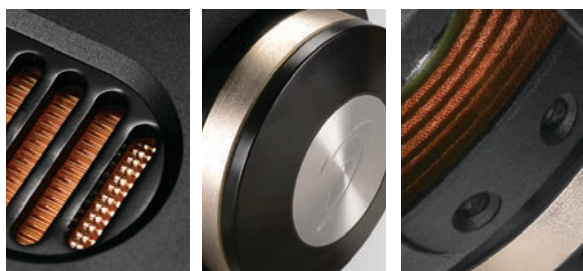
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