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ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

It's About the Sound **Stereophonic Recording:** What Do Listeners Prefer?

By Gary Galo

Standards Review

HDMI 2.0

The Evolution of a Winner

Fresh From the Bench

Focal Spirit Professional Headphones

Designed to Please Working Ears

Audio Electronics
An Alternative to
Linear Regulators
By Bill Reeve

You Can DIY!
LuminAria: A SIT Preamplifier
By Michael Rothacher



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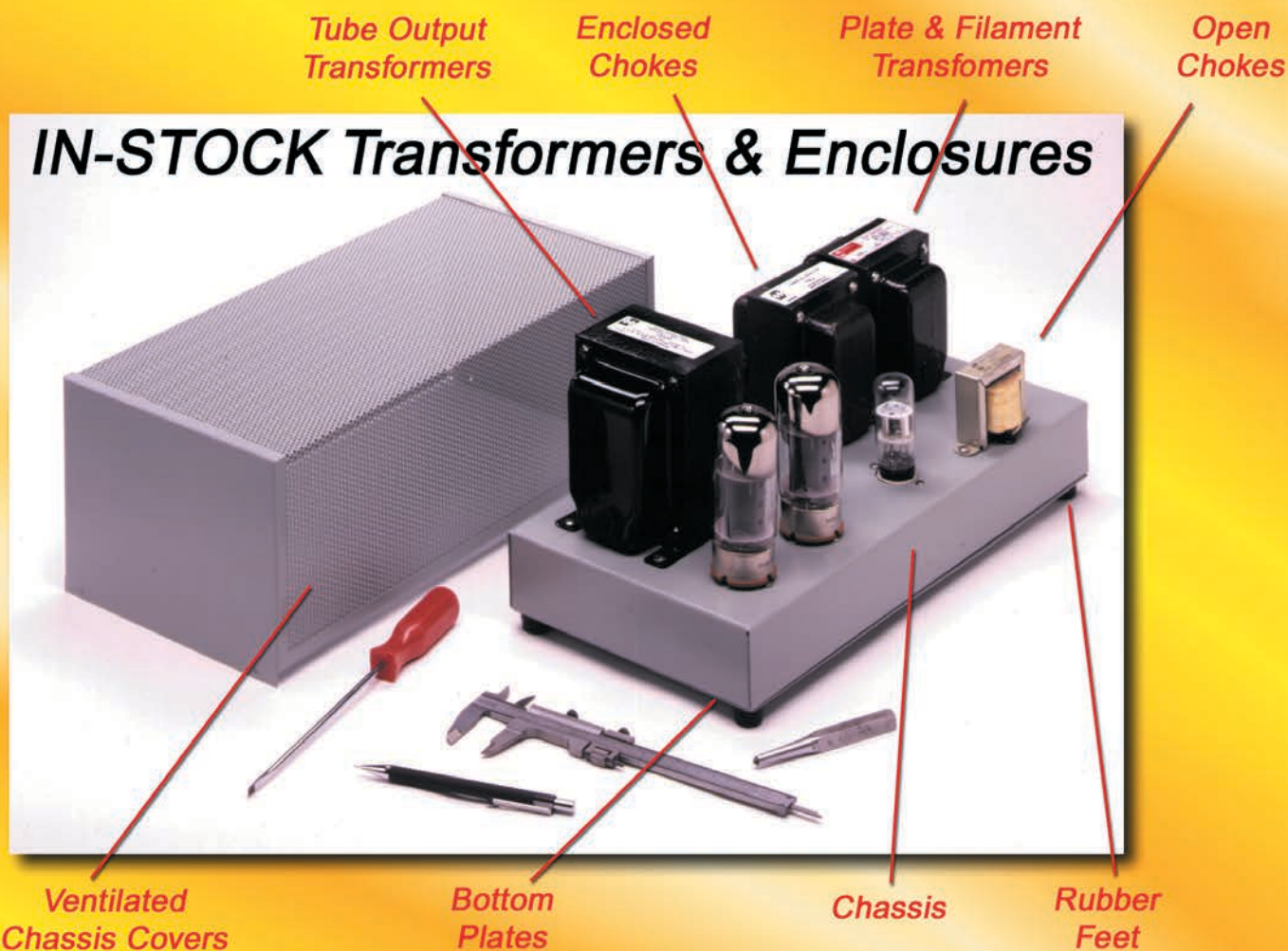
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Feeling Too Old?

With the year's first audio shows behind us, we have seen several technical breakthroughs and innovations that give us reason to be excited. Thus far, we have attended the 2014 International Consumer Electronics Show (CES), the National Association of Music Merchants (NAMM) 2014, and the Integrated Systems Europe (ISE)!

The 2014 International CES was marked by the emergence of ultra-high definition (UHD) TV. Audio professionals need to be aware of the sound implications that accompany those stunning images. Most (giant) LCD panels are so thin there's no space for high-quality speakers. The TVs may sound fine at home for sports, talk shows, and news, but there is a market for new speaker systems to complement these 4K TVs. And (sorry) I don't think consumers will opt for multichannel unless we see new speaker designs that combine great quality with convenience.

In the majority of homes, the key will be perceptual audio algorithms and a better consumer experience with less hassle. That also means wireless solutions, especially for surround sound. At the 2014 International CES, we particularly liked the Philips Fidelio E5 wireless surround cinema speakers, a "Best of Innovations" in the Home Theater category.

Also, it is critical to understand that 4K content will not arrive at homes as physical media. As we saw at the 2014 International CES, Sony's vision for 4K includes new cloud/online services, where hi-res audio and interactive gaming are also considerations. In my opinion, this means speaker and audio systems designers have good reason to work hard on new designs.

Another major trend at the 2014 International CES was wearable electronics, which will provide new opportunities, related to headsets, wireless audio, and maybe immersive/awareness experiences.

The NAMM Show in Anaheim, CA, was also a vibrant event, complete with several major announcements for the studio and stage environments and, increasingly, many incredible options for iPad and portable devices. Apps for iPad (e.g., Korg's Gadget featuring 15 virtual synths and instruments or Cakewalk's Z3TA+ iOS) sound and look amazing.

There was also a new trend that combined portable wireless speakers with guitar rehearsal amplifiers. Take a look at the VOX SoundBox Mini and IK Multimedia iLoud portable speaker and you will understand. An outstanding example came from Line 6, after its recent acquisition by Yamaha. The new Line 6 AMPLIFI combines a high-performance guitar amplifier, a streaming Bluetooth speaker, and an iOS app in one powerful solution.

The NAMM Show also saw several new studio monitors and many Thunderbolt recording interface announcements, with Zoom making a grand entrance in that category. Our favorite recording solution was the new Universal Audio Apollo Twin, a desktop 2 x 6 Thunderbolt audio interface, which enables real-time universal audio digital (UAD) processing of its high-quality plug-ins.

The greatest NAMM surprise came from QSC Audio, which revealed its first digital mixer line, the TouchMix series. Behringer also announced a new Dante network option for its X32 range of digital consoles, adding that more than 100,000 units have been sold worldwide. Perhaps inspired by that incredible success, the Music Group also announced a new 40-input Midas digital console, the M32, which is available for less than \$5,000.

If you don't feel excited by these announcements, the music is probably too loud and you may be a tad too old.

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.

NTi Audio XL2 Analyzer Updates

NTi Audio has added many new features to its XL2 Analyzer in the last few months, including several new acoustic and electric measurement possibilities.

Early in 2014, NTi launched its firmware V2.60, which offers on-site measurement of all commonly used noise curves according to American National Standards Institute (ANSI) and International Organization for Standardization (ISO) standards. The NTi XL2 Analyzer determines a room's acoustic noise spectrum and displays the results together with the standard's reference curves selected by the operator.

The XL2 Analyzer's noise curve measurements provide numerical and graphical measurement results for noise ratings (NR) in accordance with ISO 1996, noise criteria (NC) in accordance with ANSI 12.2-2008 and its 1995 legacy version, as well as preferred noise criterion (PNC) and room noise criterion (RNC) curves. The Speech Interference Level (SIL) and room criteria (RC) curve rating is also calculated in parallel. The results of all noise criteria standards are available simultaneously.

The XL2 Analyzer retains all relevant measurement values in its memory, enabling the user to switch between the different measurement standards without reacquiring the measurement data. Measurement results can be stored on the internal SD card and later integrated into user reports. They contain the applicable noise curve data and the frequency band with the highest noise band value, as well as the averaged spectrum. The XL2 Analyzer's noise curve measurements are part of the spectral limits option.

With the introduction of dBu, dBV, and volt measurement units in firmware V2.60, the XL2 Analyzer can also be able to determine the frequency response and also an audio device's intrinsic noise spectrum.

The detailed analysis of the frequency spectrum of audio signals in dBu, dBV, or volts is available with the standard XL2 instrument in octave, third-octave, and linear fast Fourier transform (FFT) resolutions. All measurement results can be stored as ASCII data for later analysis with spreadsheet programs on a Mac or a PC.

Anechoic Acoustic Measurements with the FX100 Audio Analyzer

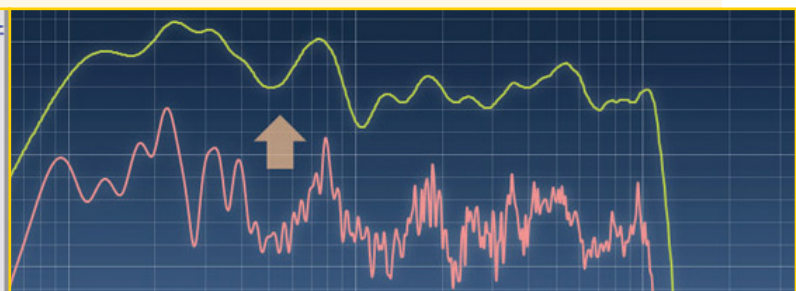
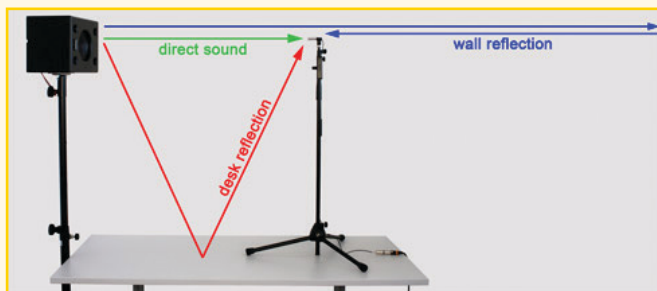
With the latest firmware and FX-Control software, NTi Audio's FX100 Audio Analyzer now supports anechoic acoustic measurements under reverberant conditions. This feature delivers free-field frequency response results, even in suboptimal acoustical environments.

The test is performed with a logarithmic GlideSweep.

The room reflections can be excluded from the measurement results by introducing a time window to the signal processing. A wizard in the FX-Control software enables localizing the reflections in an impulse-response or energy-time curve (ETC) display.

The time-window can then be graphically placed in front of the reflections, thus feeding the measurement algorithm with only the direct sound signal. The influence of the time window to the frequency response becomes visible in real time.

"Being able to conduct impulse-response measurements in a non-anechoic environment extends the measurement application range for our customers. It adds free-field-like QC measurements of speaker systems directly on the production line together with available PureSound defect detection," says NTi Audio CEO Philipp Schwizer.



With the recent updates, the XL2 Analyzer also provides support for verification of audio-frequency induction loop systems (AFILS) installations in accordance with the standard IEC 60118-4. AFILS assist users with hearing aids understand spoken messages. Such systems are widely installed in churches, theaters, and cinemas for people with hearing difficulties.

New with Firmware V2.72

In February, NTi Audio announced its latest XL2 Analyzer firmware V2.72, which provides an individual under-range indication for C- and Z-weighted sound levels. This new feature ensures precise measurements and simplifies the instrument's usability.

During low-level measurements, this individual under-range indication informs users that an alternative microphone with lower noise floor should be applied.

In the past, the XL2 Analyzer indicated when readings were below the linear measurement range for the A-weighted level in accordance with the standard IEC 61672.

Another new capability enables the statistical distribution of sound pressure levels (SPLs), commonly used for environmental noise analysis (e.g., road traffic or community noise assessments). With the new firmware, the XL2 Analyzer enables users to set



variable percentile values from 1% through to 99% and define the time and frequency weightings for the statistics.

Users can select the time and frequency weightings for the broadband source between A, C, or Z and the time weighting—fast, slow, and equivalent continuous noise level (Leq)—which provide a range of new applications. Percentile measurements are part of the optional extended acoustic pack.

NTi Audio
www.nti-audio.com/XL2

LeakSeeker 89 Autoranging Fault Locator

Shorted or leaky components in a circuit can be difficult to locate because any ohmmeter will show the same shorted or low reading at every pad along the suspected trace. Wouldn't it be great if leaky or shorted parts just illuminated and whistled at us? Well, the LeakSeeker 89 Autoranging Fault Locator Can from Electronic Design Specialists (EDS) can guide us right to the culprit with whistles and flashing lights!

EDS, maker of the popular CapAnalyzer 88A, announced the new LeakSeeker 89 Autoranging Fault Locator model EDS-89, which replaces the original LeakSeeker 82B short and leaky fault locator manufactured from 1995 to 2012.

The CapAnalyzer 88A series II checks electrolytic capacitors incircuit for both equivalent series resistance (ESR) and DC resistance (DCR) with no unsoldering. It is the only capacitor tester that discharges the cap, checks it for low DCR up to 500 Ω , reads out high-frequency ESR from 0.1 to 20 Ω , and compares readings to industry standards on the three-color chart on the front panel. The included one-handed tweezer probe with beryllium copper tips enables testing of both normal and surface-mounted electrolytics and uses coaxial cable instead of normal test-probe wire for stable readings on small capacitors.

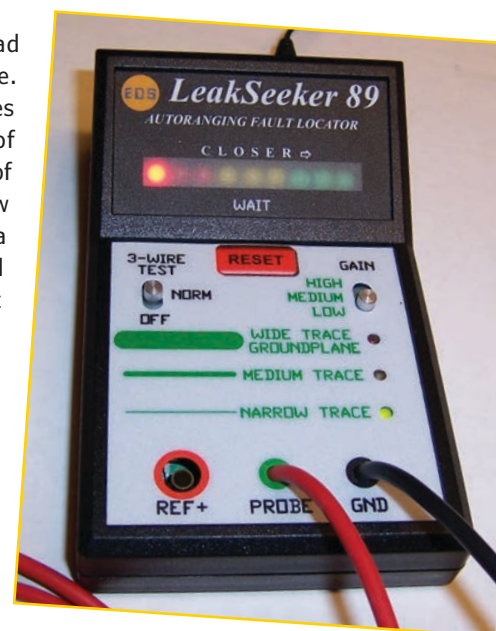
Now, the LeakSeeker 89 locates the exact spot on the PCB (to within a fraction of an inch) where a shorted or leaky component is bringing a power supply bus or data line to ground. It can locate defects from 0 to 300 Ω with no loss of resolution. It can even find active shorts that a digital voltmeter (DVM) won't show. The high GAIN mode can locate shorted components on multilayer boards with ground planes and a power layer.

With its 16-bit converter and three fully automatic sensitivity modes, the new LeakSeeker 89's gold-plated probe can be

applied to any solder pad along a suspicious trace. It automatically calibrates itself to the resistance of the defect, choosing one of 65,535 windows that show distance to the short via nine LEDs. The solder pad where the beep is highest is the location of the defect!

A special three-wire test mode isolates LeakSeeker's precision test voltage source from the probe circuitry to locate high resistance faults, even when a large good electrolytic tries to muddy up readings by trying to steal a charge from the voltage source. The thermal method of locating leaky components enables easy findings even on the most complicated boards without requiring a schematic or any knowledge of the circuit.

The LeakSeeker 89 is designed and built in the US in a ISO9001:2000-registered and certified plant. It comes with a special set of gold-plated test clips and probes and a manual with a hands-on tutorial. A technical assistance phone line is also available.



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QSC Audio in Audio Network Expansion

QSC Audio has announced several initiatives related to its audio networking solutions, starting with the confirmation that its Q-Sys Network Audio Platform will see an AES67 implementation. The company also said it will have a Dante bridge card for its Q-Sys Core processors, which have also been updated.

The Costa Mesa, CA, company's decisions confirm the market's shift toward a more interoperable ecosystem from multiple professional audio manufacturers. QSC Audio also significantly expanded its product offerings. At the Winter NAMM 2014, QSC Audio introduced its much-anticipated digital mixer line, the TouchMix Series, with the power and capability of a large-format console in a portable, fully integrated, affordable, and easy-to-use product.

At the Integrated Systems Europe (ISE) 2014 show in Amsterdam, QSC Audio confirmed that its latest software release for the Q-Sys Network Audio platform, Designer v4.0, now meets the Audio Engineering Society (AES) AES67 network clock and audio packet delivery standards. The company also stated all future Q-Sys software releases will be fully compliant with AES67 interoperability mode.

QSC Audio, as charter members of the AES's X192 task group (which has become AES67:2013), demonstrated its commitment and support of open standards that benefit the industry.

As QSC Audio explained, "The AES has long been the established voice for standards in the professional audio community, and the recent ratification of their AES67 standard for high-performance audio-over-IP interoperability is the culmination of years of bipartisan work by dedicated audio and IT professionals.

This open standard now provides the entire professional audio community with protocols that will enable A/V integrators and IT professionals everywhere to better integrate and manage their interdependent systems."

New Integrated Products

QSC Audio confirmed CDN64 Dante bridge card's availability for its Q-Sys Network Audio platform. It enables third-party Dante-enabled digital audio products to connect and interface with Q-Sys while maintaining low latency.

The CDN64 provides a bridge for 64 × 64 channels at 48 kHz (32 × 32 channels at 96 kHz) from any Dante edge network, directly into a QSC Q-Sys Core for advanced processing and redistribution over larger LAN and WAN

IT infrastructures.

This coincides with the launch of the new generation Q-Sys Enterprise Cores, which double the processing capabilities of the models they supersede (i.e., Core 1000 and Core 3000). The new Core 1100 provides up to 256× Q-LAN network audio flex-channels.

The Core 3100 provides up to 512 × 512 fixed Q-LAN network audio channels. The Q-Sys Designer 4.0 software update supports new and existing Q-Sys cores and includes many new software features (e.g., Core-to-Core paging, which enables individual cores to seamlessly transmit messages across a campus-wide installation, Timeline Playback, plus an all-new Dynamic Threshold Auto-mixer).

This software release enables greater scripting capabilities and enhancements of existing Q-Sys features including the Notch Feedback Controller, the Scheduler, the Acoustic Echo Canceller, the I/O Monitor, and Audio Playback capabilities, which have been augmented to include fast-forward, rewind, and synchronization to longitudinal time code (LTC) and GPS.

QSC Audio also introduced its new CXD-Q network amplifiers range, designed specifically for use with the Q-Sys Network Audio Platform. The CXD-Q network amplifier line combines the FAST found in the CXD/PLD amplifier platform with the same Q-Sys network connectivity and processing to create the ultimate audio routing and power solution for installed sound applications.

QSC Audio Senior Product Manager Dale Sandberg explained, "CXD-Q represents our ongoing commitment to create installed sound solutions that are incredibly versatile and integrate seamlessly with Q-Sys. As Q-Sys products, they operate like other Q-Sys peripherals utilizing the same standard Layer 3 networking technology for audio streaming, control, and monitoring. As amplifiers, they feature our new Flexible Amplifier Summing Technology (FAST) that enables the amp channels to be combined and deployed in a wide-range of installed sound applications. This allows integrators the ability and flexibility to design and install systems in a whole new way."

Adding CXD-Q amplifiers to a new or existing Q-Sys design is easily done via the Q-Sys Designer Schematic Page. The on-board Mic/Line inputs are audio on-ramps to the Q-Sys network, providing routable inputs to anywhere in the facility.



Using standard off-the-shelf Ethernet cables, the new QSC CXD-Q amplifiers can connect to any Q-Sys installation. Their Mic/Line inputs can be accessed from anywhere in the network.



QSC Audio entered the digital audio mixer market with its TouchMix Series.



The new CDN64 Dante bridge card for the Q-Sys Network Audio platform recognizes the fact that many worldwide Q-Sys installations are requesting a direct connection to Dante-enabled audio sources.

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<http://qsc.com>

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DSD

Direct Stream Digital



If these two steps seem easy enough, the free **OPPO MediaControl** app makes enjoying high resolution audio even easier. It provides complete control over your OPPO Blu-ray player. Browse music files on hard disk drives attached to the OPPO player and control playback without the need to turn on your TV. Now available for iPhone®, iPad®, Android™ and Amazon Kindle Fire. Find it in your favorite app store.

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Dante Will Support the AES67 Standard

Audinate announced plans to incorporate AES67 transport in its Dante media networking solution, currently the dominant networking solution for audio applications. The new AES67 standard, published by the Audio Engineering Society (AES) in September 2013, provides interoperability recommendations for professional-quality audio networking in the areas of synchronization, media clock identification, network transport, encoding and streaming, and session description.

"Audinate does not see networking protocols as competing technologies but more like tools in a media networking toolbox. From the beginning Audinate has incorporated and built upon standards to develop the Dante solution," stated Audinate CTO Aidan Williams. "AES67 will provide another option for interoperable Layer-3, IP-based audio networks in the future."

Dante already implements a Layer-3 IP-based transport based on the User Datagram Protocol (UDP). AES67 uses the real-time transport protocol (RTP). The addition of AES67 support provides a choice of transport protocols within Dante. Both UDP and RTP were developed by the Internet Engineering Task Force (IETF). To achieve interoperability, AES67 mandates a specific RTP payload format for delivering audio over IP (AoIP) networks and methods to exchange information about audio streams. RTP is extensively used in communication and entertainment systems that involve streaming media such as voice over IP (VoIP) telephony, video conferencing, and IP television.

Since the existing Dante transport and AES67 use routable Layer-3 IP protocols, Dante can operate in routed networks, spanning multiple IP subnets using either protocol.

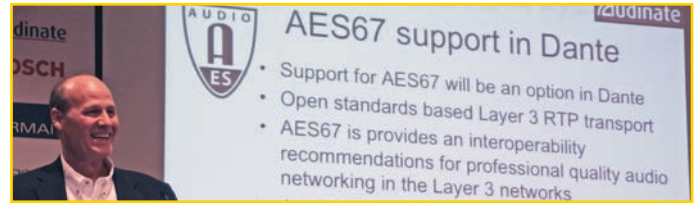
According to Audinate, "AES67 will meet the desires of customers that prefer an open IP-based transport option as part of the Dante solution. AES67 can take advantage of Ethernet switches with IEEE-1588 precision time protocol support, but unlike other network standards, does not depend on specialized switches in order to operate."

Industry Support

Audinate's announcement was endorsed by an extensive group of major pro-audio OEMs who have previously licensed Dante. Bosch Communications Systems and Yamaha Commercial Audio, both committed Dante licensees, were among the companies that applauded the move. Focusrite Audio Engineering was the first company to bring Dante technology to the studio realm with its Rednet modular Ethernet networked audio interface. Rob Jenkins, Focusrite Audio Engineering's Technical Director, stated, "Dante can interoperate with the largest ecosystem of pro-audio equipment, and having AES67 will extend that capability even further."

Solid State Logic (SSL) recently launched its first Dante-networked product and expressed support for Audinate's plan to provide an AES67 option for Dante devices. According to Niall Feldman, SSL Director of New Products, "interoperability was a primary decision in SSL adopting Dante for network I/O products and we are pleased the addition of AES67 will add further devices to the hundreds of compatible products already available."

Audinate confirmed its AES67 support as a firmware update to OEMs in initial Dante products within 12 months. During the National Association of Music Merchants (NAMM) Show 2014 and Integrated Systems Europe (ISE) 2014, Audinate confirmed new license agreements with several audio manufacturers. The companies demonstrated their



Lee Ellison, Audinate's CEO, confirmed the strong momentum for Dante during the AV Networking World conference in Amsterdam, where companies such as Allen & Heath, Atterro Tech, Audinate, Bosch, Bose, BSS Audio, Focusrite RedNet, Harman, Nexa, Open Access, Peavey Commercial Audio, Shure, Soundcraft, Stewart Audio, Symetrix, TC Group, 2N Telecommunications, and Yamaha Commercial Audio took part in discussion panels and major interoperability networking demonstrations.

latest Dante networking implementations during the AV-Networking World event, which was promoted in parallel with the ISE show in Amsterdam.

At the NAMM Show 2014, German company ESI Audiotechnik GmbH, a recent licensee, demonstrated a prototype of its planet88 Dante network audio interface with eight balanced inputs (e.g., Line, Microphone, and Hi-Z) and eight balanced outputs.

Behringer also announced three new high-performance expansion cards, the X-ADAT, the X-MADI, and the X-Dante. The cards expand the capabilities of its successful family of X32 Digital Mixing Consoles. The new X-Dante expansion card brings a 32-channel, 48-kHz bidirectional audio interface for integration with Dante-enabled IP networks using either standard 100 Mbit/s or Gigabit Network infrastructure with DSCP-based QoS, with support for mixed sample and bit rates coexisting on the same network.

At the ISE, several more companies announced licensing agreements with Audinate. Crestron, a well-known name in the installation market, was one of the most relevant. Crestron's association with Audinate enables it to integrate Dante into any of its products, implementing digital audio and control via standard Ethernet networks with near-zero latency and perfect synchronization.

"We have selected Dante because it has become the most widely adopted media networking technology," stated John Pavlik, Crestron's Director of Architecture & Design. "Dante is a proven networking technology that already has an ecosystem of hundreds of OEM products in the market, with tens of thousands of installations worldwide."

Bose Professional Systems has begun shipping three new cards that bring Dante networking interoperability to the company's latest ControlSpace engineered sound processors and PowerMatch configurable professional power amplifiers. Each user-installable Dante option card provides 48 kHz/24-bit digital audio network connectivity over standard IP networks for connecting with Bose and other manufacturers' Dante-enabled products.

Meanwhile, Extron Electronics, which is another strong name in the integration market and a longtime Dante licensee, confirmed the launching of its AXP 50 C AT, a five-input audio expansion processor with Extron ProDSP for integrating remote microphones and other sources onto a Dante-enabled Extron DMP 128 audio system.

Audinate
www.audinate.com

The Bruce Swedien Recording Method

Learning from a Master

With a career spanning 60 years, Bruce Swedien has recorded and mixed the biggest-selling albums of all time. He worked with Quincy Jones for decades and he recorded and mixed six Michael Jackson albums including *Off the Wall* and *Thriller*. He was also nominated for 13 Grammys and received five. Swedien's contribution to popular music is undeniable and his recordings have become a reference for generations of audio engineers.

The Bruce Swedien Recording Method is not only a technical book about his technical skills, it is also a valuable reference for understanding the art of music recording and popular music in general.

Swedien's career in recording began early in his life in Minneapolis, MN. In 1957, he moved to Chicago, IL, to work for RCA Victor and later for the Universal Recording Studios. He recorded Jimmy Dorsey, Count Basie, Duke Ellington, Woody Herman, Oscar Peterson, Sarah Vaughn, Dinah Washington, and many other musical greats. With Quincy Jones, he developed a creative relationship that has lasted to the present day. In the 1970s, Swedien was already working independently in New York and Los Angeles, where he recorded several albums and motion picture scores.

He became an industry legend after recording some of Michael Jackson's greatest albums, and began conducting master classes, lectures, and seminars worldwide. Swedien now lives in Florida where he started the "In the Studio With Bruce Swedien" workshops, which basically demonstrate the lessons/lectures contained in his new book.

The Bruce Swedien Recording Method details the knowledge he gained that enables him to capture the best musical recordings. He learned from observing premier orchestras and musicians. His years of technical knowledge years have enabled Swedien to reach a level of creativity and the confidence required to bring his recording to the next level, establishing some true "landmarks" of contemporary music productions.

This book is complemented by a DVD-ROM containing five HD videos, which enable us to see and hear Swedien in his studio. In the videos, he explains how he works with music, technology, recording, and mixing, and he provides practical examples. These are techniques you don't get from reading a book. The DVD also includes an interview with Bea Swedien, his wife of more than 60 years. During the interview,

she explains her husband's "positive" way of relating to artists and understanding the needs of musicians, which makes them feel comfortable and confident. The insight provided in those interviews and throughout the entire book is truly unique.

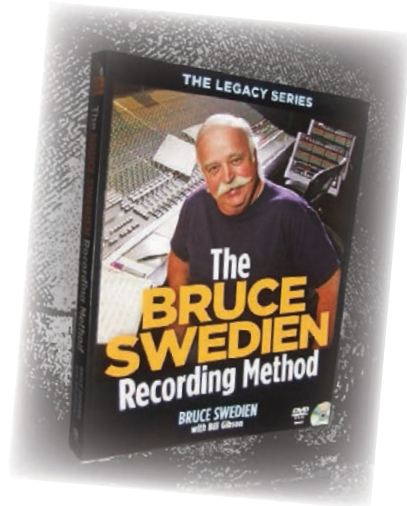
Book of Revelations

The Bruce Swedien Recording Method book is well organized and illustrated and it reveals valuable techniques for recording and mixing. Swedien's testimony explains why certain things are important (e.g., translating a mix to a master and how analog and digital recording can be different tools). The book provides lessons on music production, which is essential to the listening experience, and details the hard work involved.

The great existing books about professional audio are usually encyclopedic works—*Mastering Audio* by Bob Katz (Focal Press) and *The Art of Digital Audio* by John Watkinson (Focal Press) are two of my favorite examples—or biographies that are entertaining to read, revealing many interesting things we would otherwise never have guessed.

A great autobiography in the audio field is always a must read, but very few exist. Normally they are "assisted books," written by someone else, which can be good. Then, there are the more recent published books about audio, music, and studio work that are a hybrid version—mainly a compilation of interviews, published articles, and all kinds of testimony.

The Bruce Swedien Recording Method falls into this category. It is a valuable compilation of different materials and reflects Swedien's most recent activities in his workshops and master classes, where he generously shares what he knows. It's a unique reference book—meaning we can turn to it every time we need enlightenment. Swedien's book is a must have for anyone interested in understanding popular music and how some of the 20th century's most iconic recordings were made. 📺



By
João Martins
(Editor-in-Chief)

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HDMI 2.0

The Evolution of a Winner



Evolving from the digital visual interface (DVI) specification and competing with rival interfaces such as DisplayPort, the High-Definition Multimedia Interface (HDMI) is now one of the most successful consumer electronics standards.

By
João Martins
(Editor-in-Chief)

Fourteen years have passed since the original HDMI 1.0 specification was published. In 2009, the market widely adopted the latest HDMI 1.4a standard published, updated to 1.4b in 2011. Now, the HDMI Forum has announced the new HDMI 2.0 specification, which is designed to meet the bandwidth requirements of the forthcoming Ultra HD or 4K televisions while using existing cabling for backward compatibility. The new specification significantly increases bandwidth to 18 Gbps, supports Ethernet on the same cable, adds support for 32 audio channels, and features up to 1,536-kHz audio sample frequency and simultaneous delivery of video and audio streams to multiple users.

The HDMI was developed in the late 1990s to solve many of the problems originated by the DVI. Maybe one of the least successful initiatives in the consumer electronics (CE) and IT industries, DVI was intended to replace the dated analog VGA connection, which unfortunately remains in use today.

The HDMI 1.0 specification, published in April 16, 2002, tried to solve the connections needs of HDTV equipment, maintain backward-compatibility with the DVI connector, and respect the high-bandwidth digital content protection (HDCP) and digital rights management technologies primarily imposed by Hollywood studios. Video was always the focus for the HDMI's initial efforts. Its widespread adoption in television receivers and many types of LCDs and set-top boxes explains its success.

Another reason for the HDMI specification's success has to do with the standard connector design (Type A, 19 pins) and features, even though several variations have been proposed. This was the first truly "digital age" connector and its implementation history could be a topic for another article.

One of the best examples of the HDMI connector's success is its recent implementation by Apple, which used the connector in its MacBook Pro laptop computer series. Apple has never been a company to settle for standard connectors, especially if they compromised the design of its own products. That was the reason why Apple was a strong promoter of the smaller DisplayPort rival interface, from the Video

In 2002, the High-Definition Multimedia Interface (HDMI) was promoted as a solution for HDTV needs, replacing old analog standard connectors such as VGA and SCART and solving the many issues created by the first digital attempt, the DVI.

Electronics Standards Association (VESA) and the reason why the company pushed the Mini-DisplayPort connector version forward, to manufacture even thinner computers. Currently, the DisplayPort connection is even integrated in the Thunderbolt interface developed in collaboration with Intel. But still, Apple understood that the consumer wanted a standard option to connect to any television set or common display. The company does this with its Apple TV device. In 2012, Apple decided to bring back an HDMI connector on its MacBook Pro computers.

An entire book could be written about the story of the HDMI connector and its implementation in the industry and another book could be written about HDCP. The truth is, HDMI is an interface we all have to deal with and we'll be doing so for quite some time. Even though its evolution has been primarily dictated by video, its audio capabilities are worth mentioning, especially now that HDMI 2.0 has introduced a new set of specifications and there is increasing interest in high-resolution audio.

HDMI's Evolution

With HDMI 1.0, the specification considered the integration of digital audio in line with the same CD standards as a foundation, requiring a minimum of two pulse code modulation (PCM) channels. Other audio formats (e.g., Dolby Digital 5.1) were an option,

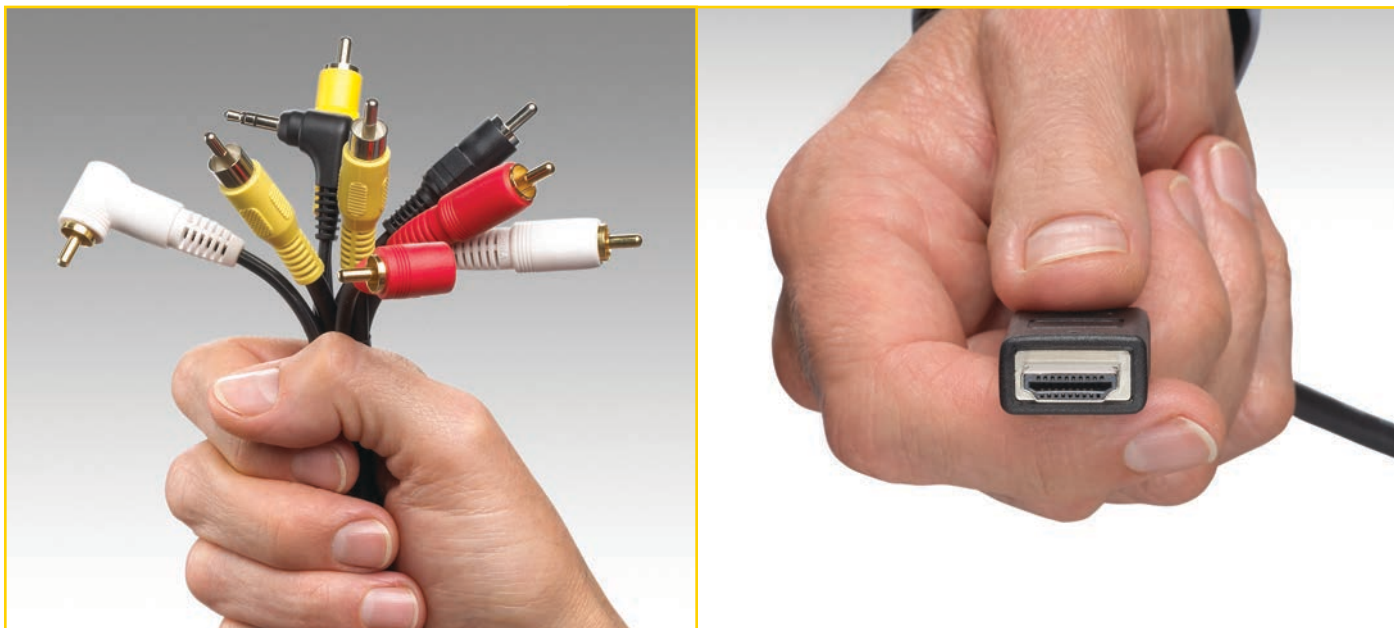
supported by up to eight-channels of 192-kHz/24-bit audio PCM. Basically the interface was designed to enable the transmitter and receiver end to define the type of format they would be exchanging, creating a two-way communication port. The Display Data Channel (DDC)—based on the I²C bus specification—exchanges data about what audio/video formats they are supporting/exchanging. The same DDC is also used for HDCP protection, including content-protected audio and content protection for prerecorded media (CPPM) and content protection for recordable media (CPRM) licensing.

The HDMI 1.2 update, announced in 2005, added support for one-bit delta sigma audio or direct stream digital (DSD), which coincided with the need to support the ongoing Super Audio CD (SACD) industry efforts. Certainly the one-bit audio and expanded sampling frequency introduced by that update made HDMI an extremely valid and robust solution for the current needs of the high-resolution audio applications—independently of the SACD format demise.

Next in line was the expansion of multichannel (e.g., 6.1 and 7.1) audio formats, with Dolby Digital Plus, Dolby True HD, and DTS-HD being supported with high bit rate and a set of specifications that finally enabled a single-cable solution for any audio source.



All HDMI cables are made of 19 individual wires. Many of these wires perform multiple tasks and transmit large amounts of data, so the quality of the conductors, the precision of the geometry, and the quality of the termination all affect the cable's ability to properly implement its tasks.



HDMI is considered the best generic interconnection for audio surround formats and one of the only reasons to invest in best cable materials, enabling for reduced jitter and lower error rates.

Another important decision was to mandate that HDMI 1.2 support low-voltage sources, enabling an important convergence with the PC market and in particular converging with the PCI Express interface.

In 2006, the HDMI 1.3 specification introduced support for the lossless audio codec formats Dolby TrueHD and DTS-HD Master Audio output streams for external decoding by AV receivers. The audio used on HD-DVDs and Blu-ray Discs could be externally decoded in uncompressed audio for up to eight channels.

It took almost another four years to bring HDMI 1.4 to the market, but this version focused on support for Ethernet over HDMI and the all-important Audio Return Channel, which means that an HDMI-connected TV with a built-in tuner can actually send audio data “upstream” to a surround audio system. The specification eliminated the need for a separate audio cable or enabled users to access something not previously available, since many TVs don’t even have a dedicated audio output. The CE industry promoted 1.4 mainly for the 3-D stereoscopic support, and the arrival of 4K x 2K resolution and additional expanded-color standards. Effectively, it was the Ethernet over HDMI support that received interest from the AV install community, which was already interested in proposals from rival DisplayPort and networked technologies (e.g., HDBaseT).

HDMI 1.4 received an additional revision “b” in 2011, a time when the HDMI Forum was already examining valid commercial requirements for another “major” update. In 2013, it initiated the HDMI 2.0 specification.

HDMI 2.0

The HDMI 2.0 specification enables the support of 4K-resolution video at 50/60 frames per second (fps) and up to 32 audio channels along with dynamic auto lip sync, among several other new features.

As with previous standard revisions, HDMI 2.0 is backward compatible with earlier HDMI versions, and doesn’t require new plugs or new cables (existing high-speed Category 2 cables with 19-pin connectors support the increased bandwidth).

The new HDMI specification is the result of intensive work carried out by the HDMI Forum member companies, expanding the support of 4K pictures beyond the 24/25 fps limit to 50 or 60 fps and increasing bandwidth from 10.2 to 18 Gbps. And it will probably not stop there, since another specification revision is expected to support the Japanese 8K ultra-high definition (UHD) standard and up to 120 fps, which will probably require a major revision in terms of hardware support.

It is important to note that, as with previous iterations, the upgrade of existing equipment from HDMI 1.3 or 1.4 to 2.0 will require dedicated firmware and some form of hardware upgrade. Therefore, the HDMI Forum is aware of the need to quickly define a stable set of industry specifications to avoid unnecessary consumer confusion.

With HDMI 2.0, the increased bandwidth can already be used to send high-quality dual-video streams (good quality 3-D in 50/60 fps), expand color depth to 10 or 12 bits, and even support the 21:9 aspect ratio video in high-quality and high-frame rates.

In the audio domain, the HDMI 2.0 specification introduces a margin to grow and implements new services (e.g., new multichannel formats—there is support for up to 32 channels, instead of the previous 8). It also enables different multichannel signals to be sent to different devices and multiple users (up to four).

The support for 1,536-kHz audio is obviously extremely important to accommodate future high-resolution audio options without any compromise. Even though such an extreme sampling frequency may seem hard to understand until we see practical industry implementations. However, it is already obvious that DSD and DXD format support will benefit from those capabilities. In simple terms, any manufacturer can implement eight uncompressed audio channels at 192 kHz with this new specification.

The HDMI 2.0 specification opens a world of new possibilities in line with the current market trends. In the same way, the SACD support provided additional capabilities that became useful for completely different reasons than initially considered, now that HDMI 2.0 has been completed, we can expect to see practical implementations and they don't necessarily only have to do with 4K televisions.

While HDMI is clearly a consumer-oriented technology, HDMI 2.0 makes it possible to also implement some specific new production applications and exciting new signal distribution possibilities, particularly for residential and some commercial installations. As long as great lengths of cable are not involved, which is the reason why the installation industry is moving to Cat5e/6 cable and fiber, HDMI 2.0 can always be considered. HDMI and technologies like HDBaseT complement each other and the 2.0 specifications provide many complementary approaches for consumer-oriented applications.

According to the HDMI Forum website, HDMI Specification Version 2.0 with bidirectional 100-Mbps Ethernet support will inspire manufacturers to implement many features currently implemented in a proprietary way, such as finally expanding the Consumer Electronic Control (CEC) application between systems from different brands, as originally intended. 



The Ethernet feature, used primarily for audio control, is the only one in the HDMI specification that requires a special HDMI cable. The HDMI Ethernet Channel (HEC) uses two pins in the standard connector plus a special cable—a single 100-Mbit/s Ethernet channel is implemented.

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Fresh From the Bench

Focal's Spirit Professional Headphones

By
Miguel Marques

(Portugal)

Focal is a well-known French speaker manufacturer of professional audio products as well as high-end speakers and car audio systems. Last year Focal released its first headphones. Among the models intended for the hi-fi and consumer markets, Focal also introduced the Focal Spirit Professional, a closed-back headphone design built for the professional musician or recording studio engineer. We recently tested the Focal Spirit Professional to discover what this model brings to an already crowded and very competitive headphone market.

Focal has been developing high-end hi-fi speakers and drivers for more than 30 years. It has established itself as one of the most prominent and valuable brands in that industry. For years, Focal also manufactured drivers and parts for other brands (e.g., KRK, L-Acoustics and Wilson Audio) in part because the company has manufacturing capabilities to build almost anything in its Saint-Étienne facilities.

Focal presents itself as a very engineer-driven company, with a focus on its R&D department. It has provided the industry with several inventions (e.g., the "inverted dome tweeter," the "W" sandwich cone, and the beryllium tweeter). The beryllium tweeter used by Focal is manufactured by Materion Electrofusion using the Truextent diaphragms, as is the case with the majority of transducer manufacturers that utilize genuine beryllium in their designs. Beryllium is 10 times more expensive than gold and seven times more rigid than titanium or aluminum, without their weight. Because of its high rigidity and low mass, it is a perfect material for tweeters. However, Focal only uses it on its premium products because it is expensive to manufacture.

Initially known as JMLab, Focal also has several products in the car audio industry. When the company opened in the late 1980s, it developed and manufactured car audio speakers, amplifiers, and systems. In 2003, Focal launched a range of professional monitors, increasing its brand strength among audio professionals. Now, it's fairly common to see models such as the Focal Twin 6be and the

Focal SM9 in studios all over the world. As part of

its expansion effort, Focal released its first headphones, the Spirit line, in 2013. The circumaural headphone line includes the Spirit One, which was created as an audiophile headphone for mobile users; the Spirit Classic, a full-size high-quality product for the hi-fi market; and the Spirit Professional headphones, the model we reviewed for this article.

Features and Specifications

The Focal Spirit Professional headphones are sold with a high-quality case filled with enough foam to secure the headphones and all the accessories in place. Focal carefully designed the case to serve as a perfect storage place when the headphones are not in use. The headphones are bundled with a detachable cord, which is obviously a

The Focal Spirit range of headphones is the first of its kind from the French speaker manufacturer.



major requirement for professional headphones. The low-impedance oxygen-free copper (OFC) coiled cable measures 13' and comes with a standard tip/ring/sleeve (TRS) connector.

Focal also included a smaller straight cable with a remote control and a built-in microphone to use the headphones with a smartphone, tablet, or laptop. This is a nice feature, as it makes the headphones usable in more applications. It also makes these professional headphones functional in daily life. The headphones also come with a carrying bag in case you want to use them outside the studio, which is likely since the headphones include another personal "mobile" cable.

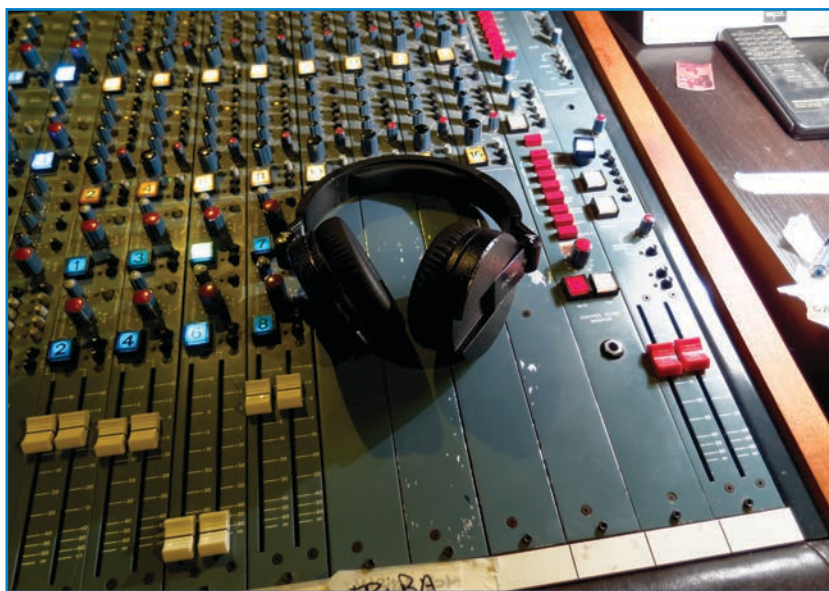
All the contents included with the Spirit Professional headphones are well made, with tremendous attention to detail that provides a high-quality feel to the product. Even though this is a professional product, it feels like a premium one.

The Focal Spirit Professional headphones have a stylish, modern look even though the all-black design makes it a very discrete, sedate pair of headphones in contrast with the two other models in the Spirit line. The only non-black elements are the Focal logo and name, which are embedded on the headphones in white. The reason these headphones aren't as ornate as most hi-fi headphones, including the Focal Spirit Classic, is because they were designed to withstand the abuse that recording studio routines usually imply. They also have a textured finish that is said to be shock-and-scratch resistant. Only time will tell if the finish on these headphones is as good as it is advertised, but they do look robust.

The handling feels very solid and the build quality is very good. We are aware that Focal cooperates with specialized manufacturers for specific components and uses external manufacturing facilities when it needs to be competitive. This enabled Focal to bring these headphones to the market at very competitive prices, in direct comparison with similar models (e.g., the Ultrasone Pro Series, using the same driver choices).

The Spirit Professional is actually one of the few products Focal isn't manufacturing in France. Yet we still are confident that they should stand the test of time and the daily recording studio use.

The Spirit Professional headphones feature a mylar/titanium dome in their transducers, which is a key element in their performance. These drivers use a mylar and titanium alloy to produce a thin and extremely light cone that's also very stiff and rigid. In theory, it's the same principle achieved with the beryllium tweeter, only with a different application. Combined with a strong neodymium magnet, these



The Spirit Professional headphones' closed-back circumaural design was developed for professionals.

40-mm mylar/titanium matched transducers are marketed as being very dynamic transducers with an extended frequency response.

The headphones are also touted to reproduce any audio source in great detail. The headphones are specified with a low 32- Ω impedance, which, along with a 102-dB SPL (1 mW at 1 kHz) sensitivity means they should provide enough sound pressure for most users regardless of the amplifier used. That also means the headphones will pair nicely with almost any battery-operated device (e.g., a laptop or smartphone).

The drivers were also designed to produce very



The headphones use memory foam on the ear cushions and in the headband to ensure a comfortable fit even when listening for long periods of time. A high-quality OFC corded cable is included for the best audio quality and a shorter straight cable with remote control cable is included for personal use with smartphones and laptops.



Designed to withstand the rough life in the studio, the Spirit Professional headphones feature a shock and scratch resistant textured finish.

About the Author

Miguel Marques is a full-time mastering engineer who works in his own mastering studio in the north of Portugal. After earning a bachelor's degree in Music Production and Technologies, Miguel worked in commercial recording studios as a recording engineer. He has also written several technical reviews and articles for pro audio magazines. He recently finished his first book about audio engineering, which was released in Portuguese-speaking countries.

low distortion numbers, which according to the brand will help prevent ear fatigue in professionals who may use them for extended time periods. To also ensure the comfort of these headphones over time, the Spirit Professional contains memory foam in the ear cushions and in the headbands.

The headphones are closed-back circumaural headphones, which obviously presents a compromise on sound quality. Closed-back designs provide sound to the user without letting in exterior sound and without letting out the inside sound, which is a mandatory feature for recording music. Because of that, closed-back headphones typically have an inaccurate and exaggerated low-end response and they're usually not detailed on the highs. With that said, Focal said thanks to a perfectly adjusted acoustical load on the transducers, the Spirit Professional headphones behave similar to an open-back pair of headphones. According to the brand, users should expect an articulated and detailed bass response.



Care has been taken when designing the box for the Spirit Professional headphones. Inside the box a lot of foam protects everything. Separate compartments hold the manual and the two supplied cables.

In Use

The Focal Spirit Professional headphones are very comfortable to wear. They are large enough to cover most users' ears and yet they aren't too bulky to feel uncomfortable. The headphones also aren't heavy as far as professional closed-back headphones go, which is a big plus. The memory foam on the headband and in the ear cushions is another big advantage. It enables the headphones to sit tight on the head without hurting or fatiguing the user.

We used the Spirit Professional headphones extensively during this review and never found them to be heavy or tiring. However, headphone comfort is subjective. We can only share our own experiences. Nor can we guarantee everyone will feel comfortable using the Spirit Professional headphones for extended time periods.

As we mentioned earlier, closed-back headphones are not usually known for their high fidelity. Their purpose is not to achieve the highest sound quality possible; that's a task typically assigned to open-back designs. The main goal with a closed-back headphone design is to assure a certain degree of sound isolation, which the Focal Spirit Professional headphones certainly achieves.

Even though no specification is provided regarding ambient noise isolation, the Focal Spirit Professional headphones seem to be as good as any professional closed-back models standard in the studio, except possibly a few models targeted for drummers that achieve even greater isolation.

When used as monitoring headphones in a recording session, we must confess that we're not fans of the supplied spiral cord. It seems to add some weight to the headphones and it may also get in the way of playing. We'd much prefer to see a longer straight cable to use when recording but since it's easy to swap the cable, there's really no problem with the bundled one. This is obviously a personal preference, and some professionals may enjoy the corded cable vs a straight one.

Despite being closed-back headphones, after a reasonable amount of usage, we were surprised to find that the Spirit Professional headphones sound great. They are probably the best closed-back headphones we've ever heard. Their level of

Manufacturer Specifications

Impedance: 32 Ω
Sensitivity: 102 dB SPL/1 mW at 1 kHz
THD: less than 0.3%/1 kHz/100 dB SPL
Frequency response: 5 Hz to 22 kHz
Drivers: 40-mm mylar/titanium
Net weight: 0.6 lb (280 g)



A soft carrying bag is included with the headphones.

detail, accuracy, and translation to the real world can only be compared with open-back designs. This is an unfair comparison, they serve different applications and most of the time different price ranges. However, it indicates their level of sound quality.

However, these are closed-back headphones. They are a bit heavy on the low end, even though the bass response is accurate and detailed. It's just slightly louder than you would expect, though it may actually be a feature instead of a flaw since it enables musicians to hear in greater detail what's happening in the low end when recording. Because of that, the Spirit Professional headphones don't come out as being bright headphones, even though they have just the right amount of top-end information to provide very detailed and fast highs.

Their amount of detail and accuracy makes them good headphones to do quality control and to check if a mix is well balanced when working in untreated rooms. The fact that they provide a decent amount of isolation yet still maintain clean, detailed sound should appeal to most professionals, because most closed-back headphones fail in that respect. And that's really what sets the Focal Spirit Professional headphones apart from its main competitors.

Quality Headphones

The Focal Spirit Professional headphones cost \$349, which is more than most closed-back headphones on the market but it is still less than most comparable European-made models. They are also better than most closed-back headphones on the market. The headphones are a safe investment for audio professionals who need a closed-back pair of headphones that serves many applications.

For the best audio performance, there are better options on the market, especially if open-back headphones are considered. However, open-back headphones are usually not an option in the studio. The Focal Spirit Professional headphones close the gap between open-back and closed-back designs. Focal created an excellent product with its Spirit Professional headphones, which will certainly find a place in a lot of professional studios and with sound engineers everywhere.

For more information, visit www.focal.com. 

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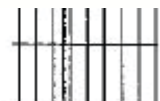
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Stereophonic Recording

What Do Listeners Prefer?

Since I first became interested in audio back in the 1960s, I've witnessed many debates about various issues that affect this field, some of them hotly contested. The audibility of differences between amplifiers, preamplifiers, interconnect cables, and even power cords still provokes controversy. Yet one issue that was the topic of debate in years past seems to have ebbed, and that is the subject of recording techniques for unamplified, acoustic music, particularly classical music. A recent study has brought this issue back to the front burner, but before describing this most interesting research project a bit of review and historical perspective is in order.

The AKG C24 remains a prized microphone for many studio-recording applications. It is shown here during a Bob Clearmountain piano session. (Photo courtesy of Apogee Digital)

By
Gary Galo
(United States)

Life was simpler prior to the introduction of commercial stereophonic recording in the 1950s. When electrical recording was introduced in 1925, a microphone replaced the horn that had been used for acoustical recording. Although the single microphone ruled for some time, as soon as primitive microphone mixers became available, recording engineers began adding microphones. Columbia introduced the modern

33 1/3-rpm microgroove long-playing (LP) record in 1948, which was also the year Ampex marketed the first commercial magnetic tape recorder in the US. By 1950, most LP records were mastered on magnetic tape and recordings made with a single microphone were the exception, rather than the rule.

The Mercury Record Corp., whose recording team was headed by producer David Hall and engineer C. Robert Fine, resisted the trend toward multi-miking. One could argue that Mercury launched the hi-fi era with its orchestral recording made with the Chicago Symphony Orchestra under its newly-appointed conductor Rafael Kubelik: the Ravel orchestration of Mussorgsky's *Pictures at an Exhibition* (see **Photo 1**). The music was recorded in Orchestra Hall, Chicago, IL, in April 1951 with a single Telefunken U-47 omnidirectional condenser microphone suspended above the podium.

The utterly natural sound on this recording prompted *New York Times* critic Howard Taubman to comment that the recording put the listener in the living presence of the orchestra. The term "Living Presence" would soon appear on all Mercury classical recordings. Mercury used the single-microphone technique for the remainder of the mono era, producing many outstanding recordings

Photo 1: This is a CD reissue of Mercury's first Living Presence recording, Rafael Kubelik and the Chicago Symphony Orchestra performing the Ravel orchestration of Mussorgsky's *Pictures at an Exhibition* (434 387-2).



with Kubelik and the Chicago Symphony, Paul Paray and the Detroit Symphony, Frederick Fennel and the Eastman Wind Ensemble, Antal Dorati and the Minneapolis Symphony, and Howard Hanson and the Eastman-Rochester Orchestra (see **Photo 2**).

Blumlein Stereo

Stereophonic recording is only slightly more complicated since, in its purist form, stereo recording involves only two microphones. But, there are a variety of microphone characteristics, along with numerous possible configurations for the two microphones. Although stereophonic recording became a viable commercial reality in the mid-to-late 1950s—first on prerecorded reel-to-reel tapes and then on LP disc records—stereo recording is actually considerably older.

In 1931, Alan Blumlein, an engineer for EMI, patented what is still today one of the purist stereo microphone techniques. The technique uses a pair of microphones in figure-8 directional patterns, with the diaphragms placed one on top of another and crossed at a 90° angle. Since the microphone diaphragms are in the same lateral plane, there are no phase differences between the two microphones. This “single-point” method creates the stereo image entirely from a difference in intensity (amplitude).

Blumlein’s method is also known as pure coincident recording, or intensity stereo. It offers precise localization within the soundstage. But, depending on the recording venue, Blumlein recordings can lack spaciousness. Blumlein’s recording method is also sensitive to microphone placement and often requires patience in finding the best location.

Several manufacturers have offered stereo microphones with the two figure-8 elements in the same housing, including the classic AKG C24 condenser microphone (see **Photo 3**). A version of this microphone with custom-tube electronics was used for a series of direct-to-disc recordings issued by Sheffield Lab, including excerpts from Prokofiev’s ballet *Romeo and Juliet*, with Erich Leinsdorf conducting the Los Angeles Philharmonic Orchestra, recorded in 1977 (see **Photo 4**).

Despite the rather dry acoustic, this disc remains one of the most spectacular orchestral recordings ever made, with phenomenal clarity, dynamics, and soundstage reproduction. The X-Y recording, a variation on the Blumlein method, uses a pair of microphones in cardioid directional patterns with the angle often increased to as much as 135°.

Spaced Omnis

At the opposite extreme of pure coincident stereo recording, the A-B method consists of two

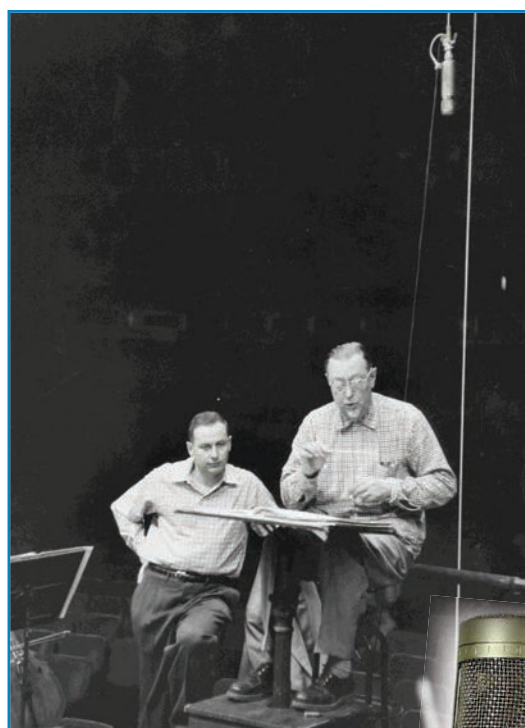


Photo 2: Mercury engineer C. Robert Fine is shown with conductor Howard Hanson during a recording session in the Eastman Theater, Rochester, NY. The single Telefunken omnidirectional condenser microphone is seen above the conductor. (Photo courtesy of Tom Fine)

spaced omnidirectional microphones. With spaced microphones, the stereo image is created by phase and intensity differences. The spacing between the two microphones can considerably vary, depending on the recording engineer’s preference, the ensemble’s size, and the recording venue’s acoustics. Schoeps, in its one-page *Overview of Stereophonic Recording Techniques*, suggests 40 to 80 cm or greater. Another worthwhile discussion can be found on the Erich Thienhaus Institute at the Academy of Music’s website (See Resources). The omnidirectional microphones (e.g., the classic model



Photo 3: The AKG C24 Blumlein classic tube microphone contained two figure-8 condenser capsules in a single housing. (Photo courtesy of AKG)



Photo 4: Sheffield Lab 8, a spectacular direct-to-disc LP of excerpts from Prokofiev’s *Romeo and Juliet*, was performed by the Los Angeles Philharmonic Orchestra conducted by Erich Leinsdorf. The recording was made with a single-point Blumlein stereo configuration.

Photo 5: Denon's 1985 recording of Mahler's Symphony No. 4 with the Frankfurt Radio Symphony Orchestra conducted by Eliahu Inbal was made with a pair of spaced B&K 4006 omnidirectional microphones (33C37-7952).



4006 manufactured by Danish Pro Audio (DPA), the successor to B&K) are among those favored by recording engineers who use this method. DPA also makes the UA0837 and UA0838 Stereo Booms for mounting a spaced pair. These booms are 54 cm in length and enable adjustment of the microphone spacing and angles (wider spacing usually involves two microphone stands).

Recordings made with two spaced omnidirectional microphones offer a spacious soundstage, but can suffer from a lack of precise localization. This technique can be very sensitive to placement, and finding the best spot can be time consuming.

An excellent example of the successful use of this technique is Denon's 1985 recording of Mahler's *Symphony No. 4* with Eliahu Inbal conducting the Frankfurt Radio Symphony Orchestra (see **Photo 5**). This recording was made at the Alte Oper in Frankfurt, Germany, using a pair of B&K 4006 microphones, (some of the other recordings in this excellent Mahler cycle used a small number of accent microphones, in addition to the main B&K omni pair). Although the soundstage localization lacks the precision associated

with coincident techniques, the sonic picture is wonderfully natural and spacious, with a lot of depth.

Office de Radiodiffusion Télévision Française (ORTF)

Broadcasters and disc-cutting engineers tend to dislike recordings made with two spaced microphones. Mono compatibility has been a major concern of broadcasters since the introduction of stereo. Because of the large phase differences between the two channels, recordings made with spaced microphones often do not sum well to mono.

Out-of-phase low-frequency information can be particularly problematic for disc-cutting engineers. It can cause large vertical excursions in the record groove, making the record difficult to cut and play back.

Around 1960, the Office de Radiodiffusion Télévision Française (ORTF) developed a near-coincident microphone technique consisting of two cardioid capsules spaced 17 cm apart at a 110° angle. The ORTF method offers much of the precise localization associated with pure coincident recording, but the spacing between the capsules introduces just enough phase difference to make the sound more spacious without seriously compromising mono compatibility. Although the ORTF was primarily concerned with broadcasting, recordings made using this technique are also easy to cut onto a disc record.

Schoeps has manufactured ORTF stereo microphones for several decades, most recently the MSTC-64, which includes a pair of MK4 cardioid condenser capsules mounted on an MSTC-6 dual amplifier body (see **Photo 6**).

Strangely, the ORTF recording method never gained any popularity in the record industry. I have favored it for most of my concert recording at The Crane School of Music, SUNY Potsdam, NY, for more than 30 years. I also use the Schoeps ORTF microphones. For me, one important attraction is that ORTF works well under a variety of circumstances. In a busy music school, we literally crank out recordings of live concerts on a daily basis, in two different performance venues. I don't have 2 h to spend finding the best spot for the microphones for each concert. ORTF has provided very good results under a variety of circumstances.

Phase Coherence

Recording engineers often use Lissajous patterns on an oscilloscope to visually check the phase relationships between the left and right channels, with the left channel normally fed to the oscilloscope's X axis and the right fed to the Y axis. If the signals in the two channels are identical in amplitude and



Photo 6: A Schoeps MSTC-64 ORTF stereo microphone is shown. Two MK-4 cardioid capsules are held at a 17-cm spacing and a 110° angle by the MSTC-6 dual amplifier body. (Photo courtesy of Schoeps)

phase, a straight diagonal line stretching from the lower left to the upper right will be displayed. Reverse the phase of one channel and the diagonal line will extend from the upper left to the lower right. A circular pattern indicates a 90° phase difference between the two.

Most digital editing programs include an oscilloscope to evaluate the phase differences between the left and right channels. Although my primary digital editor is a Sound Forge Pro 10, I like the phase oscilloscope supplied with Adobe Audition.

Photo 7 shows the climax at the end of Act II of *Romeo and Juliet* in the Sheffield Lab recording I mentioned earlier in this article. I took this from the Sheffield CD reissue, which was made from the analog back-up tapes because I didn't want random noise from the record groove to distort the image. Typical of pure coincident stereo recordings, the Lissajous shows a diagonal elliptical pattern stretching from the lower left to upper right on the display.

Photo 8 shows one of the passages for full chorus and orchestra in the "Dies Irae" portion of Benjamin Britten's *War Requiem*, which I recorded last spring with a single Schoeps MSTC-64 ORTF stereo microphone. (I had a second ORTF microphone in the balcony for the boys' choir and organ, but that microphone was not active for this passage.)

The pattern is still elliptical and shows good phase coherence between the two channels. However, the slightly wider ellipse indicates that some phase differences are part of the stereo picture, relative to the pure coincident recording shown in **Photo 7**.

Finally, **Photo 9** shows the passage for full orchestra at the end of the 3rd Movement of Mahler's Symphony No. 4, from the Denon recording mentioned earlier. The circular phase pattern is typical of recordings made with two spaced omnidirectional microphones. As the recording plays, the circle will sometimes move slightly toward an out-of-phase condition. At other times, it will move slightly toward an in-phase condition. Relative phase is not nearly as stable as it is with coincident and near-coincident techniques.

Which Method is Best?

Oscilloscope images are visually interesting, but they tell us nothing about how the recording will sound. So, which stereo recording method sounds best? Wei Lim, a young recording engineer and researcher, conducted an interesting study in which he attempted to answer that question. Lim completed two BS degrees at the University of Michigan, one in Sound Engineering and another in Brain Behavior and Cognitive Science. At the 135th Audio Engineering Society (AES) Convention, he presented a poster

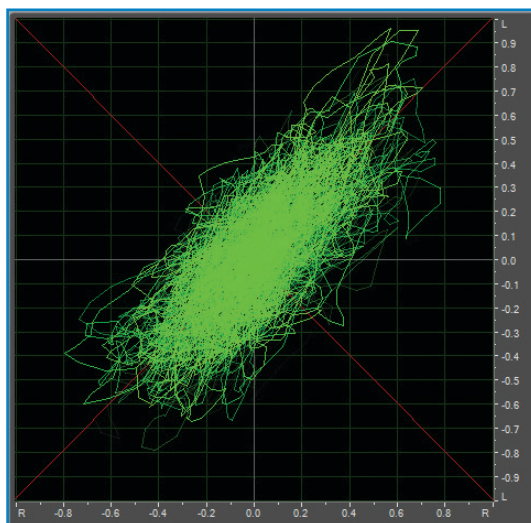


Photo 7: Lissajous pattern displayed on Adobe Audition's phase scope, showing the climax at the end of side 1 of Sheffield's *Romeo and Juliet* recording. The second CD release of this recording, made from the analog backup tapes, was used to avoid having the image distorted by noise from the record groove (10043-2-G). The pure Blumlein recording shows excellent phase coherence between the left and right channels.

session titled *An Objective Comparison of Stereo Recording Techniques Through the Use of Subjective Listener Preference Ratings* (see **Photo 9**).

Lim notes that in the existing literature there is a shortage of direct comparisons of how various stereo microphone techniques differ from one another, including the advantages and disadvantages of each. Lim conducted blind listening tests comparing X-Y, Blumlein, ORTF, NOS (similar to ORTF, but with 30-cm spacing and a 90° angle), modified Faulkner (a 60-cm spacing between figure-8 microphones), and A-B (a 60-cm spacing). Lim notes that these techniques can be classified into three categories: coincident (X-Y and Blumlein), near coincident (ORTF and NOS), and spaced configurations (modified Faulkner and A-B).

Lim began his study with three hypotheses: First, there would be a preference for near-coincident miking, because these methods combine the advantages of coincident and spaced microphone

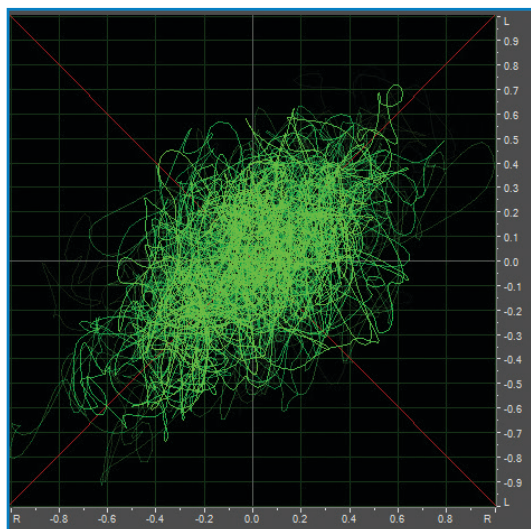
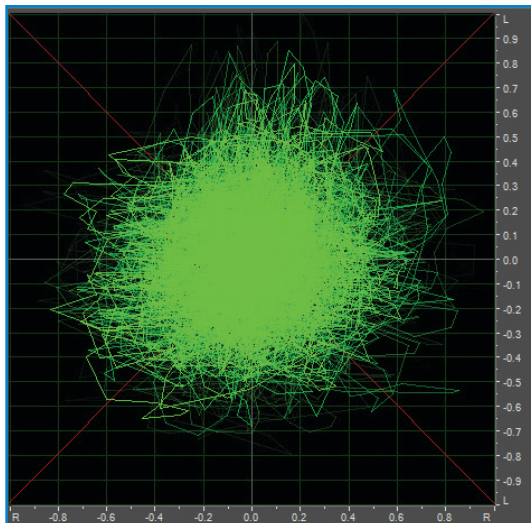


Photo 8: A passage employing the full chorus and orchestra in the "Dies Irae" portion of Britten's *War Requiem*, recorded by the author using a single Schoeps MSTC-64 ORTF stereo microphone. The slightly wider ellipse, compared to Photo 6, reflects the small phase difference produced by the 17-cm spacing of the cardioid capsules.

Photo 9: This is the passage for full orchestra at the end of the 3rd Movement of Mahler's Symphony No. 4, recorded by Denon with two spaced B&K 4006 omnidirectional microphones.



techniques; second, widely spaced microphones do not necessarily result in greater perceived spaciousness; and third, listeners generally prefer clarity over spatial content in recordings.

Lim selected listeners with two to 19 years of critical listening experience. None were compensated for their time. They were all told that their participation was completely voluntary. He conducted a blind experiment in which users submitted their relative preferential ratings for each sample played. Six questions, presented two at a time, were asked for each of the sample recordings. They covered representation of the instrument recorded, perceived wideness of the space, quality of localization, clarity of notes (in terms of attack and articulation), perceived depth of space, and general preference. The music was performed on a Yamaha Disklavier MX100A and recorded at the Rolston Concert Hall at The Banff Center in Alberta, Canada.

Lim used an upright piano that was fed MIDI information from a digital audio workstation. Using the Disklavier ensured an identical performance

for each recording, but the use of an upright piano rather than a concert instrument strikes me as rather odd. (Lim admitted that it was probably not the best choice for classical piano music.) Lim also ensured that all variables were kept to a minimum, with the locations of the piano, microphone stands, cables, and microphone preamplifiers maintained throughout the testing. Only the microphone positioning changed.

Lim found there was a significant difference in general preference ratings between coincident, near-coincident and spaced microphone techniques. The results also showed there is no significant correlation between the perceived spaciousness of a recording and the physical distance between the microphone capsules. This upholds Lim's second hypothesis.

The statistical results showed that listeners preferred near-coincident techniques, which Lim had predicted in his first hypothesis. Lim notes that near-coincident techniques provide good localization, which tends to confirm his third hypothesis—that listeners prefer clarity over spaciousness.

He also notes that while stereo microphone techniques are generally defined by the distance between the microphone capsules, the results of this study suggest that other factors (e.g., microphone polar patterns) could also influence the results. Lim says that future tests may be expanded beyond the six microphone techniques used for this test and believes it would be interesting to compare binaural recordings to traditional stereo recordings.

Additional Possibilities

Should Lim decide to conduct additional listening tests, I would also suggest adding three-spaced omni and the Decca Tree to the equation. Although they use more than two microphones, the three-spaced omni and Decca Tree are still considered purist techniques when used without addition spot-miking.

When Mercury began recording in stereo, Fine added left and right side microphones to the single, center microphone he had used for monaural recording, creating the three-spaced omnidirectional setup, which he normally used without any accent microphones (see **Photo 11**).

Fine recorded the three microphones on three separate tracks. The center microphone was fed to both channels when the stereo LP record was cut, to provide a stable center fill. Fine's stereo recordings for Mercury remain highly prized by audiophiles and music lovers to this day, for sound that is both natural and spectacular, as well as for the many outstanding performances they have preserved. Fortunately, many have been remastered on both CD and SACD. Some of the early Telarc digital recordings engineered

Photo 10: An enthusiastic Wei Lim at the New York Audio Engineering Society (AES) Convention in October 2013. Lim presented a poster session on research he has conducted on listener preference ratings of various stereophonic recording techniques.



by Jack Renner also used the three-spaced omni configuration, and are well worth hearing in the excellent SACD transfers.

The Decca Tree was invented by Roy Wallace, an engineer for The Decca Record Co., in 1954. Wallace's "Tree" consisted of three Neumann M50 omnidirectional microphones, two rather widely spaced for the left and right channels, and a center microphone mounted forward of the other two, which was fed to both channels to create the center fill.

Several companies are making hardware for mounting microphones in a Decca Tree configuration, including Telefunken. The Telefunken Decca Tree spaces the left and right microphones 2 m apart, with the center microphone 1 m forward. But, it's important to note that, for British Decca, the Tree was constantly evolving, with spacings and microphone types varying over the many years it

Recommended Recordings

Single-Microphone Mono

M. Mussorgsky and M. Ravel, *Pictures at an Exhibition*, Chicago Symphony Orchestra conducted by Rafael Kubelik, Mercury Living Presence CD 434 378-2 (Amazon or ArkivMusic).

Blumlein

S. Prokofiev, Excerpts from *Romeo and Juliet*, Los Angeles Philharmonic Orchestra conducted by Erich Leinsdorf, Sheffield Lab Direct-to-Disc LP, Lab 8 (used copies are also available on eBay). CD from analog back-up tapes, SL10043G or SL10043-52 (Amazon or ArkivMusic).

Two Spaced Omnis

G. Mahler, Symphony No. 4 in G, Frankfurt Radio Symphony Orchestra conducted by Eliahu Inbal, Denon CD 33C37-7952; Also available in the complete set of Mahler Symphonies, Brilliant Classics CD 92005 (Amazon or ArkivMusic).

Three Spaced Omnis

I. Stravinsky, *The Firebird*, London Symphony Orchestra conducted by Antal Dorati, Mercury Living Presence CD 432 012-2 or SACD 470 643-2 (Amazon or ArkivMusic).

I. Stravinsky, *The Rite of Spring*, Cleveland Orchestra conducted by Lorin Maazel, Telarc SACD 60563 (Amazon, ArkivMusic, or Telarc).

Decca Tree (with additional accent microphones)
R. Wagner, *Der Ring des Nibelungen*, Kirsten Flagstad, George London, Gustav Neidlinger, Birgit Nilsson, Hans Hotter, James King, Wolfgang Windgassen, Gottlob Frick, et al, Vienna Philharmonic Orchestra conducted by George Solti, Decca Deluxe Limited Edition LC 0171, 14 CDs plus 48-kHz/24-bit Blu-ray audio disc (Amazon or ArkivMusic).



Photo 11: A Mercury stereo session with Antal Dorati and the London Symphony Orchestra is shown in Watford Town Hall. The three spaced Telefunken M201 omnidirectional microphones are clearly visible. (Photo courtesy of Tom Fine)



Photo 12: This is a Decca recording session in Vienna's Sofiensaal showing the Decca Tree in the center, above the conductor's podium. Decca invariably used the Tree along with accent microphones, but engineers have also used it as a purist, stand-alone configuration. (Photo courtesy of John Dunkerley, The Decca Record Co., Ltd.)

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*.

was in use. Wallace initially placed baffles between the microphones, but they were removed within a few years. It's also important to note that Decca invariably used the Tree with additional spot microphones, though other engineers have made use of the Tree as a purist recording setup.

Photo 12 shows a Decca recording session in Vienna's Sofiensaal in the early 1970s, with the Tree clearly visible in the center, above the conductor's podium. Since the three-omni and Decca Tree methods require a mixer, the Benchmark PRE420 microphone preamplifier would be ideal. This microphone preamp has four inputs, each with panning capability, making a switch between two and three-microphone techniques simple, eliminating multiple microphone preamplifiers as a variable. It also has a much simpler signal path than a conventional mixer.

Lim may also wish to include Mid/Side (M/S), a coincident technique with a forward-facing cardioid microphone and a side-facing figure-8. The L+R and L-R signals from the two microphones are converted into L and R using a matrix decoder specifically

designed for this purpose. Many digital editors also have a plug-in for M/S decoding.

Lim has conducted an interesting piece of research on subjective listener evaluation of purist stereophonic recording techniques. Were he to conduct another round of tests with another instrument—perhaps a grand piano or even a small ensemble—in a different venue, listener preference may differ.

Although I tend to believe in purist recording techniques for classical music—or at least those that minimize the number of microphones to that which is necessary to get the job done—there is no single best recording method. Each engineer has his/her own preferences, and those preferences often change depending on the venue and the forces being recorded. Since the early days of commercial stereo, one can find many excellent recordings made with a wide variety of techniques, and my own collection of reference recordings reflects a wide variety of approaches.

More information on Lim's project is available on the AES website. 

Resources

Erich Thienhaus Institute at the Academy of Music, "Microphone Techniques for Stereo Recording," http://old.hfm-detmold.de/eti/projekte/semesterarbeiten/hauptmikro/hauptmikro_en.html.

W. Lim, "An Objective Comparison of Stereo Recording Techniques Through the Use of Subjective Listener Preference Ratings," Audio Engineering Society (AES) Convention Paper 8997, www.aes.org/e-lib/browse.cfm?elib=17045.

Schoeps, *Overview of Stereophonic Recording Techniques*, www.reddingaudio.com/schoeps-stereo-series-list.php.

Telefunken, www.telefunken-elektroakustik.com.

Sources

PRE420 Microphone preamplifier

Benchmark | www.benchmarkmedia.com

DPA 4006 Microphone and UA0837 and UA0838 stereo booms

DPA Microphones, Inc. | www.dpamicrophones.com

MSTC-64 ORTF Stereo microphone, MK4 cardioid condenser capsules, and MSTC-6 dual amplifier body

Schoeps, GmbH | www.schoeps.de

Disklavier MX100A

Yamaha Corp. | <http://usa.yamaha.com>

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Bass Shakers (Part 1)

Enhancing the Deep Bass Experience with Tactile Energy

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



Our perception of live music and the auditory world in general includes not only airborne sound energy, but also structure-borne vibrational energy. At a concert you see the performers, hear the music, and feel the venue resonate with the bass. For home theaters this “tactile” component adds a dimension (sometimes referred to as 4-D) to reproduce the impact of explosions, engine sounds, thunder, and other bone-rattling low-frequency content. This creative impact can be powerful, and as subtle as the anticipation created when *Jurassic Park* viewers first see the dinosaur’s footfall as waves in a puddle, feel the first tremor, and finally hear the thundering vibration.

Phonograph records did not quite capture the deepest bass, as this was too much for most phonograph cartridges to track. However, modern CDs, DVDs, Blu-ray Discs, and even the MP3s and other compressed audio formats can reproduce these low frequencies with ease. Still, the challenge remains in the playback. Getting your soundbar, home theater in a box (HTiB), or even a decent surround-sound system to realistically reproduce the feel of concert bass is an exercise in optimism.

We hear airborne music through our ears and we perceive structural bass energy through tactile perception. Reproduction of this structural component of live sound can be achieved through the use of a tactile transducer or “bass shaker.”

Bass shakers are physically coupled to various surfaces and pass low-frequency vibrations into them so that they can be felt by the listener. Tactile transducers may augment a subwoofer, but they cannot be substituted for a subwoofer. Tactile transducers are used in home theaters, commercial

movie theaters, arcade games for special effects, amusement park rides, and aerobic studios, and so forth. Some innovative applications have also been developed for the hearing impaired, bypassing the hearing system in a manner similar to a cochlear implant.

As for actual installation, a bass shaker is firmly attached to a surface (e.g., a seat, a couch, or a floor). The bass shaker houses a small weight (typically a moving magnetic structure) that is driven by a stationary voice coil. The voice coil uses a conventional power amplifier to reproduce a low-frequency audio signal (e.g., the effects from a Dolby boom channel). The voice coil exerts force on the weight and the bass shaker’s body, with the latter forces transmitted into the mounting surface.

When it comes to seating design for bass shakers, it is important to fool the senses. If the bass shaker only drives a chair, your senses won’t be fooled, particularly if it is apparent that only the chair is vibrating. But if the chair, ottoman, and floor

Photo 1: The dbx 100 Boom Box recreated the audio spectrum’s lower frequencies by seizing information in the 50-to-100-Hz range, creating a signal one octave lower (25 to 50 Hz) and mixing this new signal back into the program.

have shakers, then the entire room will appear to vibrate. There is also a preferred balance of vibration distribution between the back, waist, hips, and buttocks.

Program Processing for Tactile Reproduction

Poor bass content factored into the slow acceptance of these devices. We already discussed the limitations of vinyl LPs and cassettes. The bottom end tended to be a bit irregular due to tape speed, tape-head geometry considerations, and related compromises for good tape-to-head contact. Other magnetic tape formats (e.g., the open reel) provided acceptable bass reproduction for tactile reproduction but were not commonly found in most consumer hi-fi systems. Also, very few record labels produced any reel-to-reel program material.

An early kluge was the use of a subharmonic synthesizer (e.g., the dbx 100 Boom Box). This device synthesized extreme bass from the existing bass signal (see **Photo 1**).

Another issue with using stereo two-channel program material was that any dialog with bass content (e.g., a deep male voice) would be included with and detract from the tactile effect.

CDs and Dolby surround, which has a dedicated “boom channel,” changed all this. The Dolby effects channel separated the dialog from the signal sent to the subwoofers (and tactile devices). Response optimization benefits from the most realistic tactile contribution and adds to the overall experience.

In reading comparative bass shaker reviews and attempting to measure output, we want to offer some caveats. As with most electroacoustic transducers, there is a gain vs bandwidth tradeoff with just about everything. It is easy to design an efficient bass shaker with huge output. Think of a bass shaker as having a Q, so a high-Q design would have a lot of output over a very narrow band of bass frequencies—similar to a one-note subwoofer.

Alternatively, we can design for a wider Q and more linear response over the operating range, but at the expense of efficiency. Products that are reviewed by unsophisticated hobbyists and found to be super-efficient may just be a “one-note” design that wants to vibrate at their favorite but narrow frequency range.

Some History

An entire generation of moviegoers first encountered tactile effects when *Earthquake* debuted in 1974. In this disaster film, a catastrophic

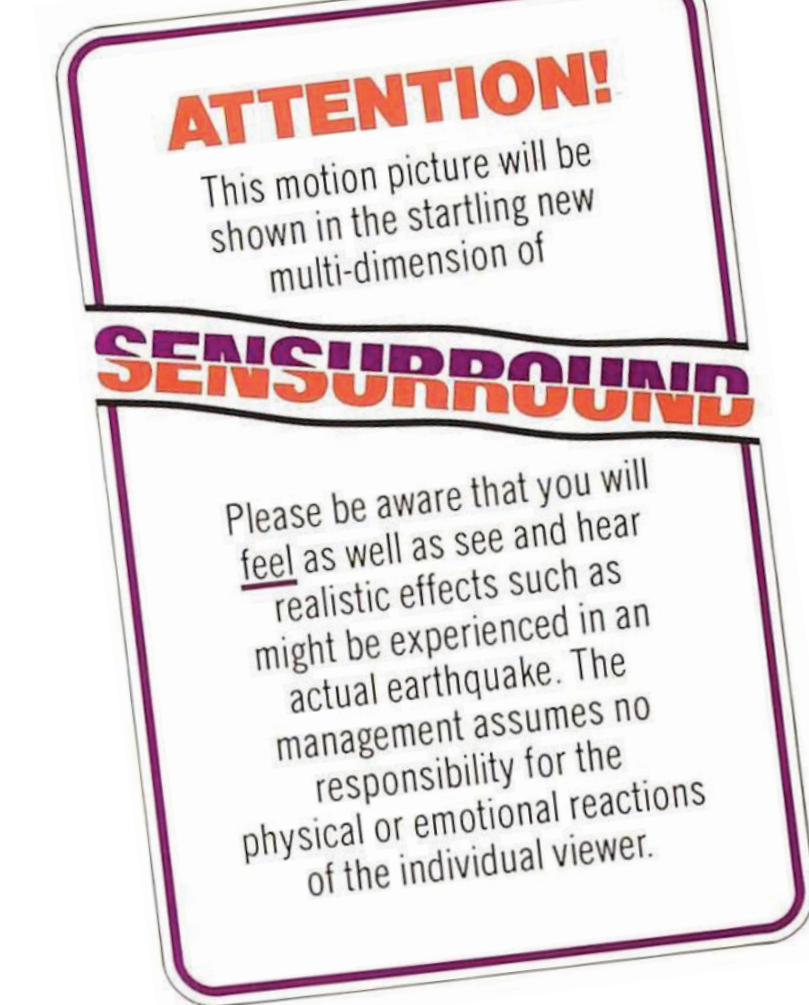


Photo 2: Movie theaters displayed this Sensurround disclaimer when showing the movie, *Earthquake* in the late 1970s.

earthquake destroys most of Los Angeles, CA. In a previous article, we discussed how the speaker company Cerwin-Vega, in conjunction with Universal Studios, developed deep bass horns driven by high-power amplifiers and used gated noise generators as the signal source to shake the theaters at the appropriate moments. The experience rattled moviegoers and was a huge commercial success.



Photo 3: The Aura Model AST 11-4 is an early bass shaker that literally sent sonic vibrations through the material to which it was attached. (Photo courtesy of Aura Systems)



Photo 4: Aura System's Interactor display box is shown. The Aura Interactor, which was sold as "virtual-reality gamewear," connects to the computer's sound output to let users feel the sounds and vibrations. (Photo courtesy of Aura Systems)

Clearly, the entertainment consumer had an appetite for *Earthquake's* tactile experience, and several films featuring "Sensurround" quickly followed (see **Photo 2**).

Just two years later in 1976, Nozomu Matsumoto founded Bodysonic.

Matsumoto is best known as the founder of Pioneer Electronics in Japan.

Bodysonic was the first "bass shaker," which Matsumoto supposedly developed in his hobby shop.

Sometime between 1982 and 1984 author Mike Klasco experienced tactile transducers firsthand. At the time, he represented Bodysonic in the US for commercial sound applications (e.g., amusement and theme parks, discothèque floors and other non-residential applications). One flagship project was for the Disney Epcot Center's GE Pavilion, where the ride seats had Bodysonic installed.

Bodysonic also had some success with automobile sound systems. It was installed in the 1984 Nissan 300ZX anniversary edition. Toyota also

worked with Fujitsu-10 on a couple of sports models using Bodysonic technology.

In the mid to late 1980s, Klasco worked with Sammi Sound on its Volcano shakers, which were also sold as the Phillips Resonators. They were popular with autosound aftermarket installers, which used them to supplement the subwoofers.

Around 1992, Klasco consulted for Aura Systems, which at the time was a high-tech firm focused on the US government's "Star Wars" projects. Aura Systems designed and built powerful push-pull actuators with unique magnetic topology for some very esoteric military applications. As the firm wanted to diversify to consumer products, Klasco suggested that a tactile bass device would be a great fit for its core competence in-linear actuators.

A speaker group was formed with a very talented team. Drew Kallen and his associates filed for bass shaker patents in 1993 and Aura Systems entered the shaker business, eventually building variants for other brands including Alpine and RBH (see **Photo 3**).

Another clever product from Aura Systems was the Interactor Vest, which users strapped on similar to a backpack. Designed for video games, it provided a huge impact. But, the product was too ahead of its time for the gaming products of the mid 1990s. Today, the Interactor would be ideal for connection to the bass effects channel (see **Photo 4**).

In 2001, Panasonic introduced tactile headphones. Marketed as "A Subwoofer for Your Head," the RP-HS900 Brain Shaker Extreme incorporated a shaker in the neckband (see **Photo 5**). Fast forward to 2014 and Vivitouch is bringing a new generation of 4-D Sound Haptic headphones to market. We will discuss these wearable devices in a future article.

Current Bass Shakers

In the current market there are a number of bass shaker products. Among them are a few of the more popular devices from ButtKicker, Clark Synthesis, and Earthquake.

ButtKicker—ButtKicker's tactile products are used in movie theaters, simulation rides, dance floors, 4-D theaters, planetariums, science centers, and amusement park attractions. The patented technology is promoted for home theaters as well as for unique applications at sporting events to broadcast along with the sound and video.

For sports-viewing applications, specially designed wireless sensors can be placed on the players. The impact they feel can become part of the broadcast, enabling fans at home to feel like they are part of the game. ButtKicker LIVE! brings



Photo 5: The Brain Shaker Extreme headphones have a virtual motion sound system (VMSS) control switch that lets you actually feel the bass. (Photo courtesy of Panasonic)



Photo 6: The Earthquake Quake 10.0 is a tactile transducer designed to reproduce the effects of the lowest frequencies in music and movies. (Photo courtesy of Earthquake)

“enabled sports” to the event (e.g., stadium, race track, or arena) as well as the home. By installing ButtKicker units in the seating (e.g., luxury boxes or a sponsor’s “best seats in the house”) fans can feel and become part of the action as it happens.

Clark Synthesis Transducers—Originally developed for military and commercial training applications, these tactile transducers can be used full range or for bass. Full-range applications include use with hot tubs and wall structures (for invisible speakers) and similar applications. The waterproof version can be used on pool walls.

Crowson Tech TES 100—The Shadow-8 Dual Motion Actuator is designed to bring tactile effect to all types of chairs and couches. The Shadow-8 actuators can be installed by a person with no mechanical skill in less than 15 min. No modifications to furniture or flooring are required. The system easily interfaces with standard audio equipment using conventional audio signals.

Earthquake of San Francisco—Earthquake’s products range from high performance to over the top. Specifically, Earthquake’s Quake 10.0 is an ideal transition to next month’s topic of motion simulators, which is one step beyond tactile shakers. The Quake 10.0 is sensitive, articulate, and renders even the most complex low-frequency material in every detail. It is so powerful it can even replace some of the traditional hydraulic platforms in theme parks (see **Photo 6**).

Beyond Tactile—Motion Simulation

Shakers go beyond subwoofers to provide the tactile sensation of structural vibration, but they cannot simulate the gravitational forces of vehicle or aircraft acceleration. The motion cues necessary to trick the mind into thinking it is immersed in the simulated environment and experiencing kinematic changes in position, velocity, and acceleration require additional degrees of drive forces. This

requires a motion simulator or motion platform to create the effect of being in a moving vehicle or roller coaster.

Next month, we will explore motion simulators for home use. Then, we examine “invisible speakers,” most of which use similar techniques as shakers but provide full-range sound reproduction by driving the walls with the audio signal. 

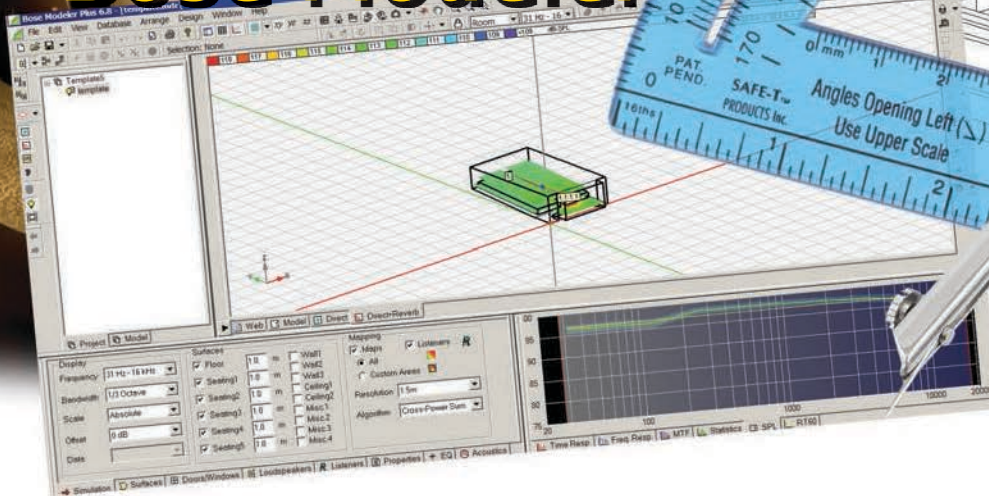
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Bose Modeler



Bose Modeler electroacoustic prediction software is one of the earliest ray-tracing computer acoustical modeling programs. It was first released for MacIntosh in 1986. The current Windows version is Modeler 6.8. It is designed to be used as an aid to scientific sound-system design and as a sales tool for sound contractors (especially in regard to its auralization capabilities). It is not intended to be a complete room-acoustics modeling program.

By
Richard Honeycutt
(United States)

Modeler's current version provides an implementation of what Bose calls the D²RASTIC procedure for designing a sound system:

- Calculate **D**irect coverage
- Calculate **D**irect + **R**everberant coverage
- Predict time arrivals—direct, first-order reflections, and second-order reflections (Δ)
- Predict speech intelligibility using the speech

transmission index (**STI**)

- Confirm the design by auralizing, using Bose Auditorioner

Upon opening Modeler, a screen pops up containing a grid on which you can build your model by clicking on corners (with the aid of X, Y, and Z coordinates) in the XY, YZ, and XZ planes. You can build the floor and extrude the walls. You can add material characteristics such as absorption and scattering in octave bands from 31 Hz to 16 kHz. Alternatively, you can import an EASE or DXF model if you already have one or prefer modeling in EASE or CAD. Successfully importing a DXF model requires that there be a materials layer in the DXF file.

Once you have built or imported the model, you will see the screen shown in **Photo 1**. The model shown in the photo is a small shoebox auditorium with a raked floor and a small stage. The box containing the numeral 1 represents a listener position; the wider box containing 1.1.1.1 represents the omnidirectional loudspeaker. Modeler also offers the ability to view the model in plan or to reverse the isometric view.

You can click on the Place Loudspeaker icon, and

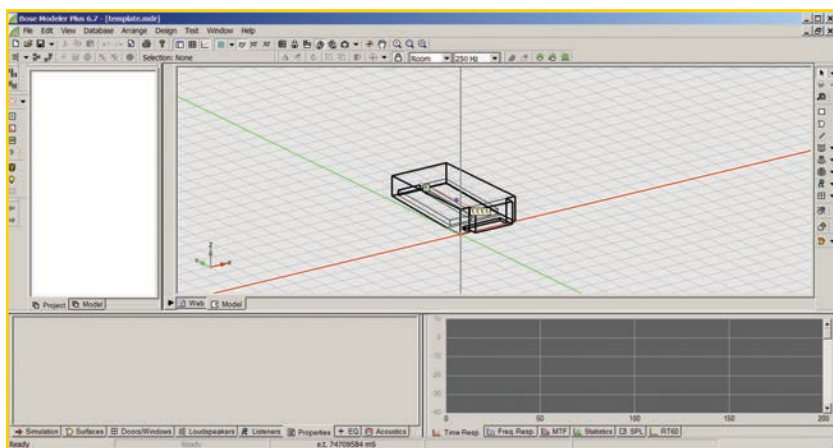


Photo 1: A screenshot similar to this one appears once you build or import a model.

then click on the model to place the loudspeaker. A dialog box will open for you to input the speaker's coordinates and select the speaker model. Clicking Database\Loudspeaker Database brings up the box shown in **Photo 2**.

Material Selection

Materials can be selected from the software's material database. You can also enter data for new materials. Next, you can click the RT60 button in the lower right. A graph similar to **Figure 1** will appear in the associated space on the screen.

Modeler includes a process to automatically make the predicted reverberation time (RT) match measured RT values by modifying material properties. This process can be useful, but must be intelligently used. It makes sense to modify absorption properties for which the values are not well-known (e.g., audiences and draperies), but the absorption of some other materials, such as painted concrete floors, are well known and should not be modified in an attempt to force predictions to match measurements.

In any real room, the effects of a material's increasing absorption in a certain octave band will depend on the material's location in the room. The Modeler procedure enables the user to choose which material to adjust and even which frequency for that material. Materials can be "locked" so their absorption will not be changed. The maximum permitted change in reflection strength is 3 dB every time the matching cycle is run.

If you have included speakers and seating areas, you can right click on the model and select "Direct (1 to 4 kHz)" from the D²RΔSTIc pop-up list. The speaker's direct-field coverage will then be displayed on the model (see **Figure 2**). Since this model uses an omnidirectional source, the direct coverage is symmetrical from side to side, and of course the sound pressure level (SPL) decreases with distance from the source.

In this model, the room has hard walls and ceiling so sound levels experienced by the audience will be significantly affected by the reverberant field also. Modeler also includes a "Direct + Reverberant (31 Hz to 16 kHz)" option in the D²RΔSTIc pop-out list. For this model, the results are shown in **Photo 3**.

Notice that there is a frequency response curve shown in **Photo 3**'s lower right corner. A similar curve is also included in the direct response display, although it was not included in **Figure 2**. Fans of omnidirectional speakers may notice that the direct:reverberant coverage is uniform within about 1 dB over most of the audience area. However, the omnidirectional speaker puts a hot spot near the

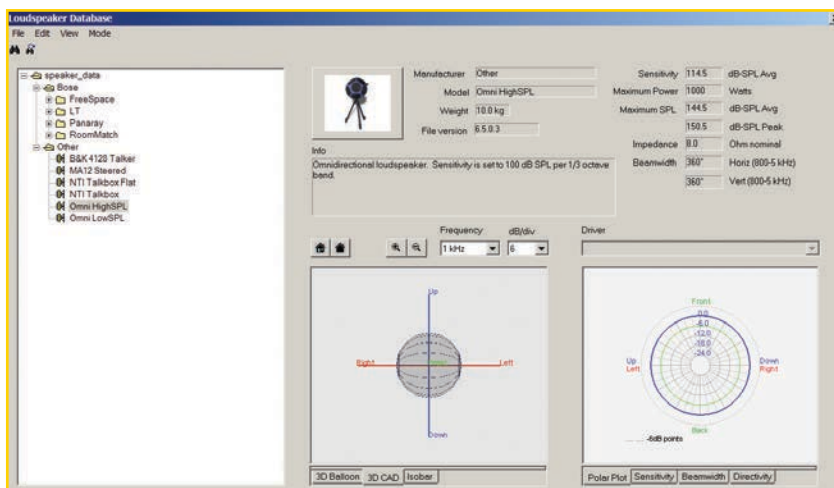


Photo 2: Modeler's loudspeaker database contains technical details for several loudspeakers.

front of the stage, which would make feedback squeal a problem. If SPL were the only consideration, a single loudspeaker with a wide horizontal coverage could be used in this room.

Sound-System Design Factors

Late-arriving sound (greater than 50 ms after the direct sound), noise, some types of frequency-response anomalies, and distortion created in the sound system are also important considerations in sound-system design. These can be quantified by speech intelligibility measurements.

Most modeling programs, Modeler included, assume that the sound system is correctly set up so distortion is not a factor. Modeler includes a method of incorporating EQ to eliminate frequency-response anomalies. Modeler and most other modeling programs permit the user to enter specific noise signatures (noise levels in various octave or one-third-octave bands) so the noise can be included in speech-intelligibility calculations. STI is the most nearly standard metric for speech intelligibility. Modeler includes its most comprehensive tool for examining late-arriving sound under the STI rubric.

To examine the late arrivals at a specific listening point, you can click the STI button on the left-hand side of the screen, click the Time Response tab near the bottom center of the screen, then choose a single frequency in the drop-down menu. The menu can be accessed via the Simulation tab at the lower left of the screen. **Photo 4** shows the resulting image.

In the graph at the lower right of the illustration, the red vertical line represents direct sound from the loudspeaker. If more than one loudspeaker were used, there would be multiple red lines, unless all

Figure 1: Modeler shows the model room's predicted reverberation time (RT).

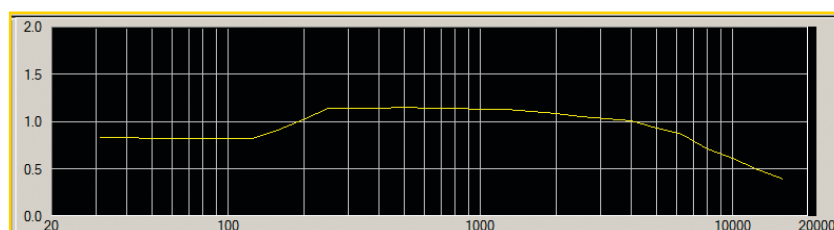
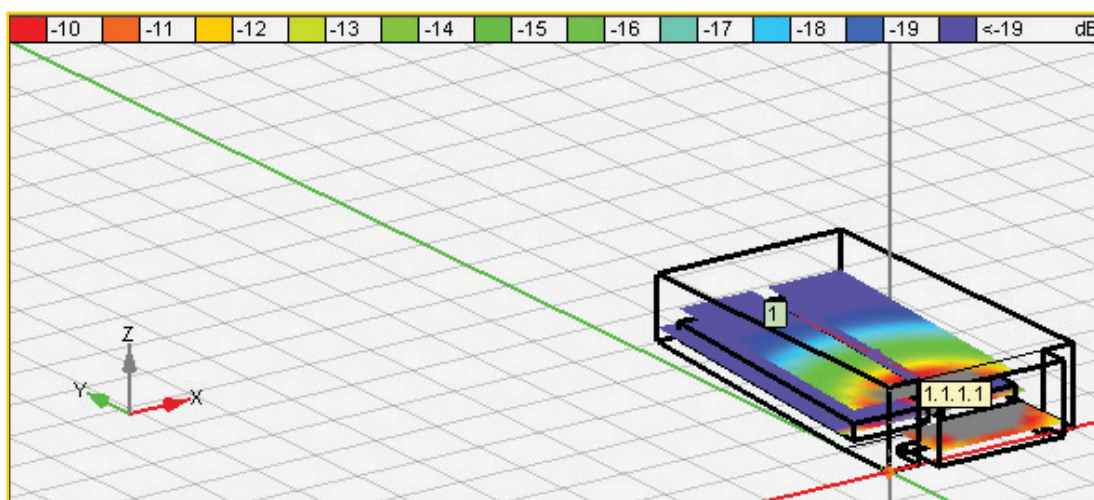


Figure 2: This is the direct-field coverage from an omnidirectional source in the small auditorium.



speakers were time-aligned. Green lines are first-order reflections (i.e., sound that has reflected off one surface). Blue lines are second-order reflections, which is as far as Modeler tracks reflections. The yellow line shows how reverberant energy builds and decays in the room, and is called the reverberant field envelope function (RFEF).

In this model, the latest second-order reflections are almost exactly 50 ms later than the direct sound, so they would not be heard as echoes. Higher-order reflections could be perceived as echoes, if they arrived more than 50 ms after the direct sound and had sufficient level. However, Modeler does not examine these late reflections, nor does it include an algorithm to predict the listener disturbance resulting from an echo at a given delay time and SPL.

Acoustic Imaging

Correct timing and SPL of late arrivals is also important for proper acoustic imaging: making the sound seem to come from the actual acoustic source

(e.g., the talker, singer, musician, etc.) rather than from one side or the other. The earliest-arriving sound's arrival direction is considered by the ear and brain to define the source's location. False directional cues are caused by strong reflections or secondary loudspeakers with incorrect delays.

Clicking F8 causes Modeler to present the STI map and data, per the 1988 standard. It also predicts per the 1998, the 2003, and the 2011 standards for both male and female voices.

For most of the room, STI is about 0.55, which is considered fair speech intelligibility. It is in the good range (STI = 0.6 to 0.75) in the front one-quarter of the auditorium. This distribution of STI values is likely caused by the use of an omnidirectional speaker, which provides a poor direct-to-reverberant sound ratio for most of the room.

The graph at the screen's lower right shows the modulation transfer function (MTF) in the octave bands. The bands are color-coded: red = 125 Hz, orange = 250 Hz, yellow = 500 Hz, green = 1 kHz, blue = 2 kHz, indigo = 4 kHz, and violet = 8 kHz.

The MTF is a representation of how much the signal is degraded by the sound system and room. It is plotted as the decimal fraction of modulation (information in the signal) that is preserved (vertical axis) vs the modulating frequency. Perfect conditions would be indicated by an MTF of 1 at all frequencies. (It would appear as a horizontal line across the top of the graph.) Reverberation causes a downward-sloping MTF (see **Photo 5**).

In this model, the room is not exactly reverberant, but the speaker does not have the directivity needed to put enough direct sound into the rear of the audience area. Background noise causes a decrease in MTF at all frequencies. Strong echoes cause a notch in the MTF. By examining the MTF, we can identify the cause(s) of poor speech intelligibility.

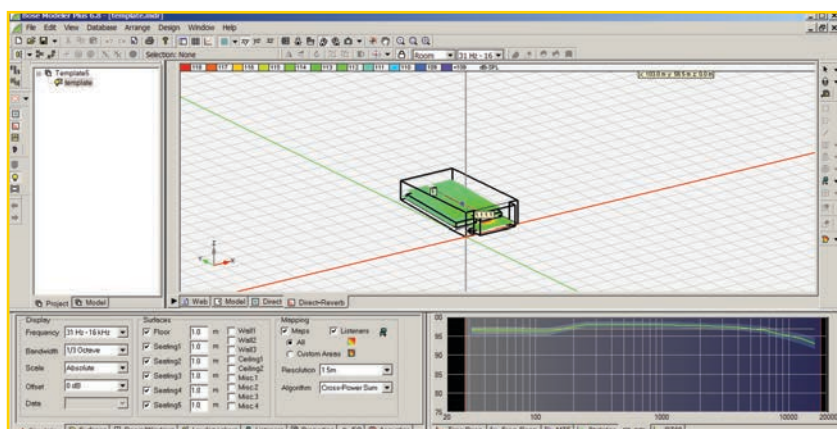


Photo 3: The direct + reverberant coverage is usually much more even than the direct coverage.

Auralization

The last step of the D²RΔSTIC procedure is to confirm the design by auralization. Auralization involves convolving some anechoically recorded sound with the room's impulse response. With music performed live in a venue, the music itself is free of echoes, and the process of playing it in the room acoustically convolves it with the room's acoustics. Auralization software performs the convolution mathematically, using the impulse response derived from ray tracing.

For the auralization experience to be valid, the listener must be in a controlled acoustical environment; otherwise, the listening room's acoustics will color the results to an unacceptable degree. One way to control the listening environment is to incorporate binaural processing into the wave file produced during auralization, and ensure that the listener uses excellent headphones with good rejection of external sound. Here's the rub: Few clients (with the exception of recording and broadcast studio owners) have—or even know how to identify and where to get—excellent headphones, even if they were willing to spend the necessary money for them.

Bose created a portable set of speakers with on-board electronics to provide an acoustically controlled environment for listening to auralizations. This system is called the Bose Auditioner (see **Photo 6**).

Modeler is designed so that it will only permit auralization if an Auditioner system is connected to the computer, thus providing a measure of quality control for the auralization experience.

Overall Function

As stated earlier, Bose's Modeler is a sophisticated software tool designed to facilitate scientific design of sound systems. It performs this function well, with a friendly user interface. The calculation time for each function is extremely short. Model input is straightforward once users become familiar with the method. The other option is to import a DXF or EASE model.

Some non-Bose speaker data in the CLF or EASE formats can be imported into Modeler. GLL and DLL data cannot be imported. A special key is required to permit importing non-Bose speaker data.

Modeler is not designed as a general-purpose acoustical modeling program and does not include functions such a program would have including the ability to view echograms and ray traces of reflections beyond second-order, EK grad echo analysis, calculation of strength (G), clarity (C80), definition (D50), mean free path analysis for analyzing reverberation, walkthrough auralization, or auralization

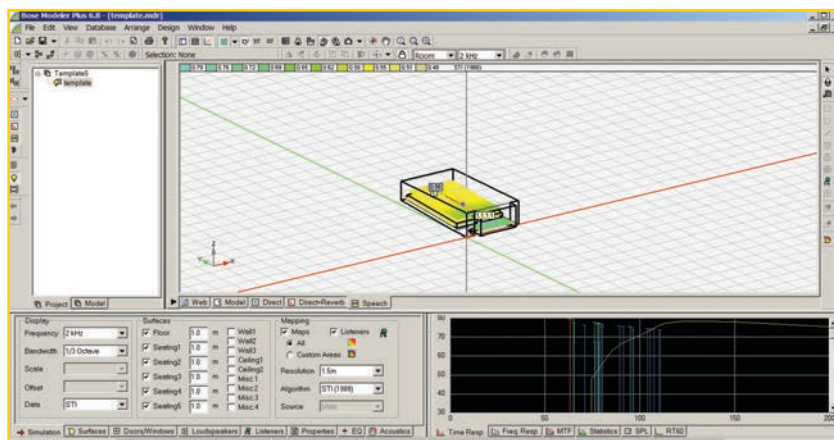


Photo 4: Modeler can enable users to examine late sound arrivals.

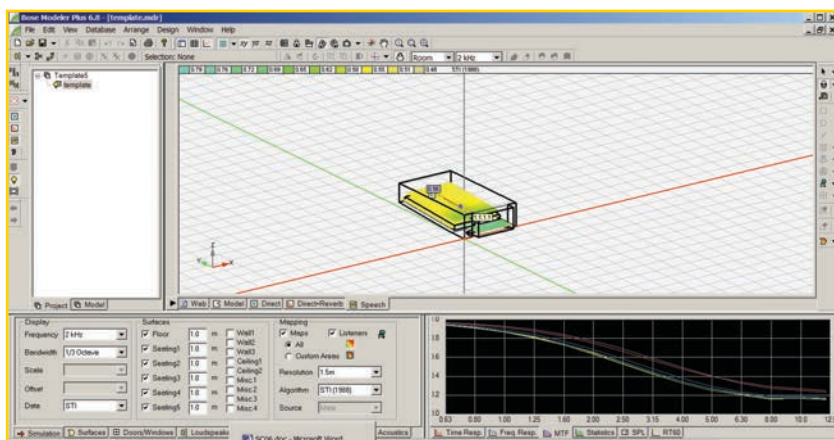


Photo 5: Modeler includes a significant amount of STI data in a single screen.



Photo 6: The Bose Auditioner is designed to provide a controlled environment for listening to auralizations. (Photo courtesy of Bose)

using headphones or external speakers.

A prospective user should carefully evaluate his/her needs before choosing Modeler or any other modeling program written by a loudspeaker manufacturer over a general-purpose modeling program. Modeler and software by other manufacturers such as Yamaha and Danley Sound Labs can be cost- and time-effective solutions for sound system designers who do not need the extended capabilities provided by the programs designed for more general applications. ❌

Recording Engineer Introduces Innovative Designs

An Interview with Craig Bernabeu of SBS Designs



By
Shannon Becker
(United States)

SHANNON BECKER: What compelled you to start your company, SBS Designs, in 2011?

CRAIG BERNABEU: I was not really seeing audio products designed for a variety of applications with different approaches to record or play back music that would suit my needs. So in late 2009/2010, the former head engineer from Summit Audio and I started designing visionary gear. SBS designs makes my vision of US-made high-end designs with a left-field approach available to users.

SHANNON: How did you choose the name SBS Designs?

CRAIG: I have a sound company called SBS that designs custom high-end analog systems for different venues. SBS Designs builds custom speaker cabinets, consoles, and desks for mixing consoles and outboard gear. I had originally planned to call the company SBS Electronics. Due to legal trademark issues I could not use the name so I came up with SBS Designs, which works perfectly. SBS is globally known for the systems I have designed in the past so I wanted to incorporate the name into my new business venture.

SHANNON: Tell us about your company's first product.

CRAIG: Our first product is the Iso-Q2, which is a three-band program equalizer with infinity cut, 15 dB of boost with five front controllers, three gain controls and two adjustable frequency potentiometers for the low and top end. Its I/O is balanced and unbalanced

with a selectable effects loop. The Iso-Q2 is 115 and 230 V ready. It also offers a rear-panel output gain trim for perfect gain matching or it can be used to reduce gain for heavy-handed end users.

The Iso-Q2 enables the end user to sweep the low frequencies from 50 to 400 Hz with a front panel potentiometer. The high frequency is sweepable with another front panel potentiometer from 3.15 to 12.6 kHz, and it offers a frequency response from as low as 8 Hz to 100 kHz. It's primarily used in recording studios, and mastering houses; as live PA for mix engineers, electronic dance music (EDM) performers, and DJs performing in clubs or at festivals; or for sound reinforcement.

This Iso-Q2 can also be used as a performance processor to cut and boost on the fly, cutting out and isolating target frequencies from a song to get an audience excited in a live venue to take the energy to the next level. In the studio, the Iso-Q2 is intended as a program equalizer to sweeten up the source or equalize rough frequencies that need to be smoothed out.

SHANNON: What other products have you developed?

CRAIG: The SP-1 is the company's first tube processor. The SP-1 is a unique design, offering features and technology that has not been previously developed. It is designed to warm up and improve the dynamic range. The SP-1 is a handmade hi-fi processor that will let you record and add dimension to music in ways never achieved.

Craig Bernabeu used his audio knowledge to start his own professional audio company, SBS Designs.



a)

With the SP-1, I wanted to put a twist on vacuum tube processing. From the original concept, it took more than five years to get it perfect. We designed a few different concept prototypes and then put the prototypes through years of testing in different applications including live, studio and even home hi-fi applications.

There were a few things with the prototype that I did not like, so I started redesigning it and our engineering experts said I was crazy, and that no company would put this kind of R&D into one product. We spent thousands of hours on the SP-1.

That's what is different about SBS Designs. The product needs to be right and very unique so it really stands out. So we went back to the drawing board and made multiple changes until it was perfect. Now, the SP-1 is my vision of what it should be.

What is really unique about the SP-1 is its reverse phono curve option, which is a first of its kind of feature with full control with the front panel controls. The SP-1 offers a phono stage to use with turntables for end users who want to get the most from vinyl to sample.

I wanted to include a feature that enables users to put phono curves on line sources to attain digital or analog line sources as close as possible to the sound of turntables. It is really amazing how good you can get line digital sources to sound.

SHANNON: Can you share some of the other challenges involved with the designs?

CRAIG: One major challenge is coming up with unique designs to put a twist on audio by offering features not previously done or different to the last design I have already brought to the market. But the biggest challenge with this philosophy is offering features for the end user but not at the expense of the sound quality.

To ensure this process properly works, I design my products two or three years before they are available to the public. Then I can really test them in a variety of applications so I know they work how I intend them to and sound the way I expect. This is very tricky to do, especially when I need SBS Designs to have perfect synergy with other high-end brands.

I have never approached any SBS Designs product with the idea that it will be like anything else on the market. I have always wondered what I could



b)

The Iso-Q2 is a solid-state three-band isolator/three-band program equalizer with five usable front panel controls (a). The Iso-Q2 can be used in an effects loop insert, mixbus, or main output of any console (b).

do differently than others to get the most out of the source and give users innovative options that are fun to use. I want to bring the fun factor back to recording or playing back music.

SHANNON: What makes your amplifiers unique?

CRAIG: The S-series amplifiers, which range from the S1 to the S6, are uniquely designed with a high-end old-school approach. Attention to detail is our first priority. Everything from the board layout and circuit design to all the components on the board, right down to the heatsink and manufacturing, are made to our specifications so the amplifiers sound and reproduce amazingly.

Our head of engineering and I are sticklers about this. If it doesn't sound and function a certain way, it does not get released no matter how long it takes to get done. I don't operate with time restrictions just to rush it and get it out the door. If it's not right, it does not leave the facility.

So I think what makes us different is that we ensure our products are the way we want them to be throughout the entire process. This attention to detail is understood day one.

This especially applies with our ratings. We don't play the ratings game just to show impressive numbers and have our products fall short of their specifications or go with the 1/8 duty cycle rating like most amplifiers do. Our S1 and S2 amplifiers offer 50% true duty cycle full output loads at full frequency from 10 Hz to 100 kHz and the same goes for the way we specify our larger S3, S4, S5, and S6 amplifiers.

We also decided to offer two high-current amplifiers. The S5 is a pure Class-AB, 2-Ω stable amplifier. The S6 is a Class-AB +B that is 2 Ω stable and designed to work flawlessly with 2-Ω load-driven hard and passive speakers with difficult chokes. In these applications, the S5 and the S6 would work perfectly and sound amazing.

SHANNON: How has the audio community reacted to your products? Is the audio market a difficult one to enter?

CRAIG: The community has been very receptive to SBS Designs. I have received several compliments and our products have been selling globally. It seems the audio community is really enjoying the twist I have been putting on audio by adding different features to get the most from a mix or a sound system and providing really high-quality sound, which allows the end product to be improved whether it is live or studio recordings.

It's been great so far with a lot of successful engineers and producers getting behind SBS Designs with great results on a production or in a live application.

The S5 is used in several mastering studios and receiving praise from engineers such as Nick Moon at Tone Proper Mastering in Oregon. Moon is using the S5 on the road to mix and he recently used the S5 to mix Liv Warfield when she performed on the *Late Night with Jimmy Fallon* show (<http://sbsdesigns.com/liv-warfield-mixed-on-the-jimmy-fallon-show-s5-amp>). So the S-series amplifiers are really getting incredible praise and demand is growing as SBS Designs's reputations spreads.

The audio market, as with any market, can be difficult when you offer a left-field approach so it can take a while to make an impact. But, if you're determined, you can make it happen, and it will.

SHANNON: Tell us a little about your background. What did you do prior to starting SBS Designs?

CRAIG: I went to school in the 1980s in New York City and got a degree in recording. I worked in a few studios then I started focusing on high-end analog

pro sound system designs. They sold globally via word of mouth, and my sound system designs won or were nominated for best sound system times from 2003 to 2011.

I also built studio reference systems and worked closely with many major manufacturers and their engineers to change their products to structure them to meet my needs.

After several years, I noticed some of the great audio components be replaced by cheap products. I decided to design my own high-end products and I came up with a concept for an entire line.

I approached Summit Audio with my concept and they were interested. The first product, a tube processor that took two years to develop, was released in 2002. Next, we developed the amplifiers and then a solid-state equalizer. I also became an investor in Summit Audio in 2004.

After working with Summit Audio, Summit's former head engineer and I started developing an entire new line. That's when I started SBS Designs. SBS Designs are now sold in retail outlets in the US and a few countries around the world. It is growing fast and amazing things are happening for SBS Designs in 2014.

SHANNON: What's next for your company?

CRAIG: I have been working on my 2015 products for the last year and a half. My background is speaker design, and I have completed a full monitor line that I would eventually like to release.

For 2014, SBS Designs has two new designs—the SP-1 PRO and the SX-3. The SP-1 PRO is our second vacuum tube processor. It will feature seven front-panel usable controls, designed to really let the end user improve the dynamic range of any full-range source.

Similar to the SP-1, a carefully designed expansion process will bring out exciting detail that is so often masked by recordings that are compressed and engineered to be mono compatible. The SP-1PRO will also bring back incredible detail to digitally stored/processed recordings that were subject to data reduction processing.

SP-1 PRO also has a sub bass expander and an extended range of top treble expander. These expanders are top quality fully analog that don't contain any noise or distortion generators. They add dynamic range by controlling the target frequencies with photoresistors.

When bringing a second vacuum tube processor to market, I had to be creative and provide features not previously used to help the end user receive more from the vacuum tube technology. (More



The SP-1 tube device has two front-panel controls (a). The SP-1 offers a line level to phono level playback with a reverse phono curve to a line level source (b).

information about the tube technology is available at www.sbsdesigns.com.)

In late 2014, we will release the SX-3, a two-way, three-way dual mono stereo analog crossover or a four-way mono. As with my other designs, I put a spin on it that you just don't see on other crossovers, especially in the analog domain.

SHANNON: Do you have any advice for *audioXpress* readers who want to build their own sound systems?

CRAIG: Learn the importance of the physics of a cabinet design for a loudspeaker. Do not think you can build a speaker just any way and make corrections in the processing. This is a huge misconception in proper loudspeaker design.

Try to learn how to design a speaker so it can naturally reproduce music with as little processing as possible. Overprocessing seems like the norm these days, but the best, most amazing sound is the least processed with as little crossover points possible. A good speaker system effortlessly works



The SP-1 PRO includes seven usable front panel controls (a) and it can be used in any fully balanced or unbalanced circuit without any drive loss (b).

with the least amount of gear in the signal chain and provides great results.

Before beginning a loudspeaker design, truly understand an amplifier's specifications not just the watts per channel. Learn the Thiele-Small (T-S) parameters of a loudspeaker. Knowledge of room acoustics is also ultra important so there are no weak links in the chain. That way the end result will deliver as promised and maybe even more than promised. 

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You Can DIY!

LuminAria

A SIT Preamplifier

The annual Burning Amp Festival (BAF) in San Francisco, CA, is the premier event for DIY audio enthusiasts. Each year the BAF provides handcrafted equipment displays, listening rooms, brilliant talks by audio cognoscenti, raffles, giveaways, sausage, and beer. Last summer I resolved to do whatever it took to attend, and since many of my audio friends and heroes would be there, I didn't want to go without a DIY project to display. So, I had to come up with something small enough to fit in my suitcase and light enough to schlep from my hotel to the festival, which was held at Fort Mason on the bay.

BURNING AMP

By
Michael Rothacher
(United States)

Paradoxically, in audio, invention is oftentimes the mother of necessity so I decided I needed a preamplifier. All I had to do was design and build a BAF-worthy, good-sounding, unusual preamplifier with a spiffy set of performance specifications and have it show-ready in about two months. Nothing inspires me quite like an unreasonable objective and self-imposed pressure. This is the story of that project, which I named the LuminAria, warts and all.

Design Goals

Starting a design project is easy, but finishing one generally requires some specific goals so we'll begin there. DIY preamplifier projects are a bit scarcer than amplifier projects; perhaps this is because preamplifiers can be a little more difficult to build, demanding more in terms of switches, volume controls, wiring, metalwork, and so forth. So, when I decided to do a preamplifier project, I knew I wanted something simple, not only to satisfy my minimalist taste, but also to ensure that it may actually get built.

I also decided this would be an active preamplifier with voltage gain. Passive and buffered-passive preamplifiers are popular among audiophiles because they handle input switching and volume control without adding much coloration to the music signal. As long as your amplifiers have enough gain and cooperative input circuits, they do a really great job.

However, I like to experiment with simple little one-stage amplifiers that often require some special attention at the input in terms of voltage and source impedance (see my article, "L'Amp: A

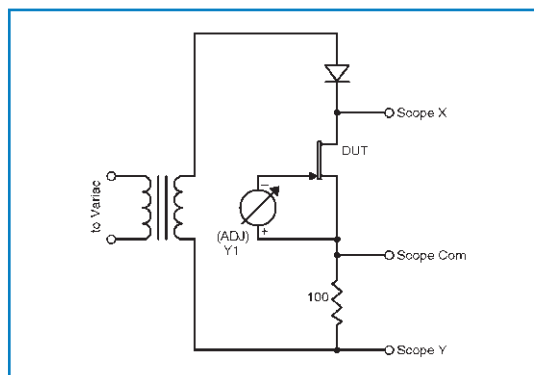


Figure 1: The makeshift curve tracer enables us to get a closer look at the 2SK82.

Simple SIT Amp," *diyAudio*, 2011). I wanted a versatile preamplifier that would address these issues and work well in a variety of situations.

How much voltage gain and output do we want? I thought 12 dB or more would be nice, and it would be good to provide more than just a couple of volts in case we want to drive a power follower such as a Firstwatt F4 or some other design; let's say 20 Vpk or more.

High input impedance would work well with a variety of music sources, and low output impedance would be good in case the amplifier has low input impedance and/or high input capacitance, which is sometimes the case with minimalist amplifiers. And, naturally, there should be really low distortion with little or no feedback.

To summarize: a simple, unusual preamplifier with ample gain, a lot of voltage swing, high input impedance, low output impedance, low distortion and no feedback, which fits conveniently in a suitcase would be preferable.

There's Plenty of Room at the Bottom

I also decided it might be fun to explore uncharted waters and experiment with some exotic devices. Thanks to Nelson Pass, I'm hooked on static induction transistors (SITs). Their characteristics are great for building simple, low-distortion gain blocks with little or no feedback. Having built a number of single-stage power amplifiers with SITs, I wanted to see if these devices could work their magic in a preamplifier.

SIT characteristic curves are very triode-like and linear so they seemed like a good fit for a best-of-both-worlds preamplifier design. Of course, a SIT whose characteristics are optimized for the voltages and currents found in a preamplifier circuit would be optimal, but when I searched for a suitable small part I quickly discovered the choices are really limited. A few small SITs have been made (e.g., the 2SK63 or the 2SK79) but I wasn't confident that enough of them would be available to support this DIY project, so I chose the more readily obtainable Sony 2SK82.

Now, you may be thinking that's a power part. It doesn't have enough gain. What about capacitance? Insert your favorite protest here. But, sometimes we have to work with what we've got. Working through the various problems and opportunities can be part of the fun.

First, observe that the current-voltage characteristics (i.e., I-V curves) in datasheets don't always tell the whole story, and the ranges of current and voltage shown are chosen for specific applications. Sometimes though, it is useful to "zoom out" or "zoom in" on a particular part of the graph and see how things look on a different scale. This is easily accomplished with a curve tracer if you have

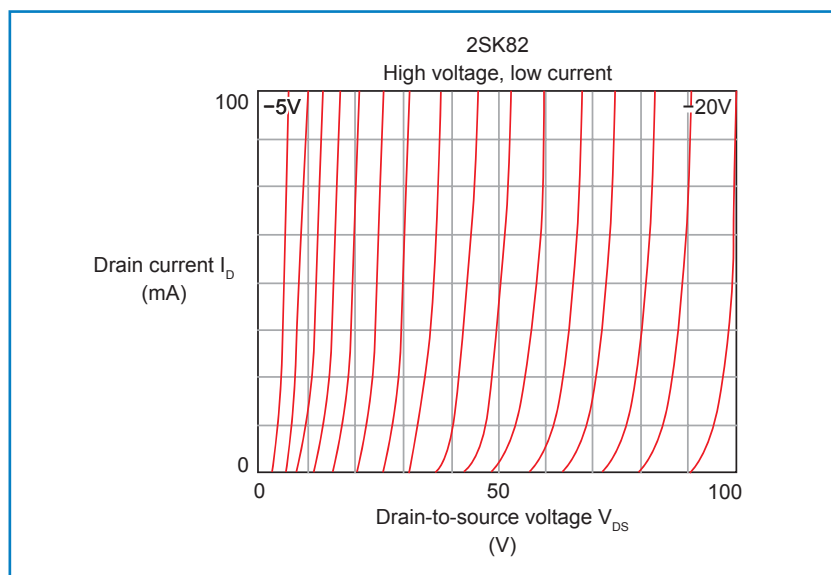


Figure 2: The 2SK82's I-V curves are shown at high voltage and low current.

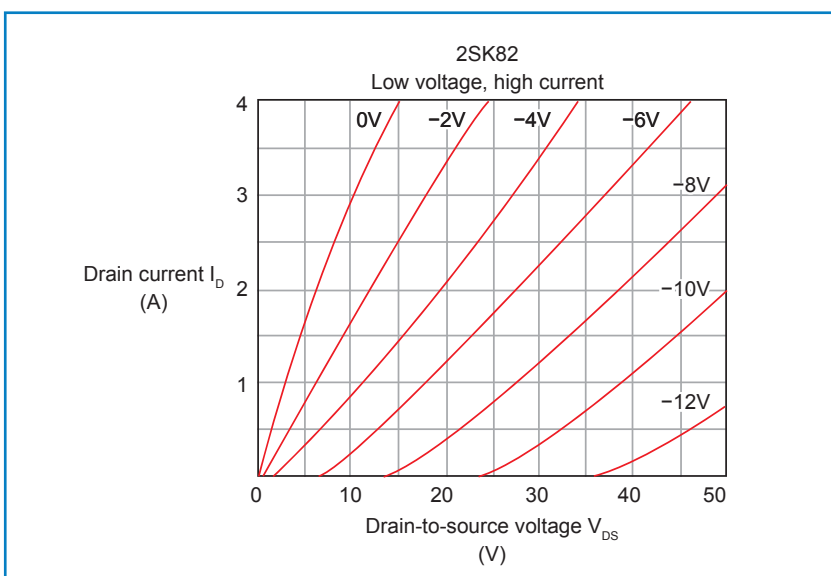


Figure 3: The 2SK82's I-V curves are shown at low voltage and high current.

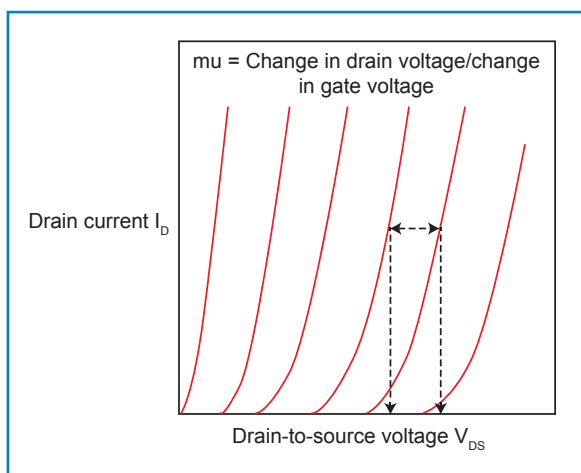
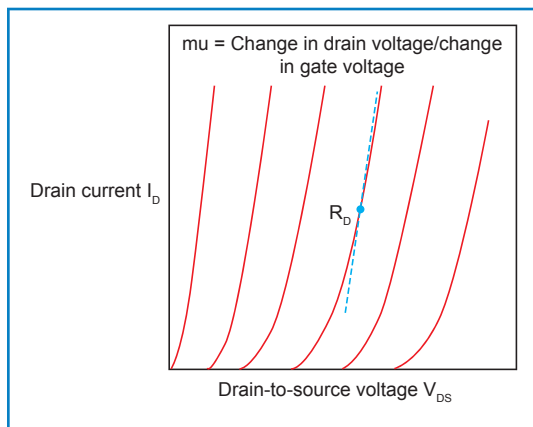


Figure 4: The amplification factor (μ) is determined by examining the change in drain voltage divided by the change in gate voltage when the drain current is held constant.

Figure 5: The drain resistance shows the change in drain voltage divided by the change in drain current with the gate-to-source voltage held constant.



one. Of course, curve tracers can be expensive, especially the ones that measure high voltage and current.

To get a closer look at the 2SK82, I used the circuit in **Figure 1** to generate the set of curves in **Figure 2**. This is an economical and lab-expedient way to make I-V curves. However, it is more labor-intensive than using a curve tracer and you have to be very careful not to exceed the device's limits. I detailed the specifics of creating a plot similar to this in my L'Amp design. Because the voltage and current scale is different, these curves look more like a triode's familiar curves compared to those in **Figure 3**, which shows the same device at power amp voltage and current levels.

SIT Characteristics

The rescaled I-V curves show how the device behaves within the desired operating window. To determine the device's amplification factor (μ), examine the change in drain voltage divided by the change in gate voltage when the drain current is held constant (see **Figure 4**). It can be easily found from the 2SK82's characteristic curves. **Figure 2** shows that μ increases as the drain voltage increases.

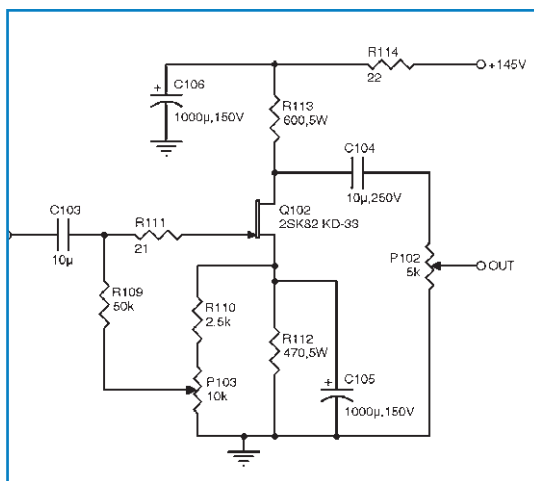


Figure 6: The circuit is a simple common source amplifier.

At around 80 to 90 V and 50 to 60 mA, the μ is approximately 8. This would be difficult to surmise from the curves shown in **Figure 3**, which suggest a μ of 4 or 5, since the horizontal axis only goes to 50 V. A higher amplification factor is necessary for this project, and this opportunity may have escaped my attention if I hadn't rescaled the curves and taken another look.

Another important characteristic is drain resistance, which is the change in drain voltage divided by the change in drain current with the gate-to-source voltage held constant (see **Figure 5**). It is the slope of the line tangent to the curve at the point of interest (e.g., the operating point). The SIT's drain resistance makes it possible to design low-output impedance gain stages without global negative feedback.

Figure 2 also shows that the drain resistance at around 80 to 90 V and 50 to 60 mA is approximately 75 Ω . This is much lower than the plate resistance encountered in a small triode vacuum tube.

SITs, like other transistors and vacuum tubes, exhibit capacitance between their terminals, which if sufficiently high, will negatively impact high-frequency performance.

In my L'Amp design, a SIT power amplifier utilizes the 2SK82, and the effective input capacitance is approximately 2,500 pF. High capacitance isn't uncommon with big-die power devices, and using one in a preamplifier is a worthwhile challenge. The good news is that the 2SK82's input capacitance decreases as the voltage increases across it. Still, at 80 to 90 V and 50 to 60 mA, the capacitance will be about 1,500 pF. This will need to be addressed if the music source's output impedance is too high.

The Basic Circuit

Figure 6 shows a simple common source amplifier. The source pin is tied to ground and the output is taken from the SIT's drain. The common source configuration provides both voltage and current gain. If a music signal is introduced between the SIT's gate and source, the AC voltage varies the AC current from drain to source. It is then converted to voltage across the drain resistor, R113, which provides an amplified copy of the music signal at the output node, high-pass filtered by C104 to remove DC.

To bias this circuit properly, a negative DC voltage must be supplied between the SIT's gate and source to set the flow of current from drain to source. As this bias voltage is made more negative, and the SIT will conduct less until finally current no longer flows, pinching off the SIT.

This circuit uses adjustable self-bias. Self-bias

employs a resistor attached from source to ground, which raises the SIT's source by the voltage dropped across this resistor. If the gate is held at some lower voltage or ground, it will become more negative than the source. Here, the bias voltage is developed across R112, and P103's wiper provides an adjustable tap to control the gate-to-source voltage, which in turn controls the DC bias current through the SIT.

The R112's value is chosen by examining the SIT's IV curves and determining the gate-to-source voltage required to set the required idle current and drain voltage. **Figure 2** shows an operating point of 90 V at 50 mA will required around -19 V from gate to source. We'll want to drop a little more than this (e.g., 24 V) across the source resistor to enable some adjustment and variation among parts. Use Ohm's law to find the resistor value (470 Ω was the closest available value):

$$R = \frac{V}{I} = \frac{24 \text{ V}}{0.05 \text{ A}} = 480 \Omega$$

There's also a large capacitor wired across the source resistor. To understand its purpose first look at the voltage gain without C105:

$$A = \frac{\mu R_D}{R_D + r_d + R_S (\mu + 1)} = \frac{8 \times 600}{600 + 75 + 470 (8 + 1)} = 0.98$$

Where mu is the amplification factor in the operating region, r_d is the drain resistance, R_D is the drain resistor value, and R_S is the source resistor value. Clearly, -0.18 dB doesn't meet our design goal. The gain is drastically reduced because R_S is large compared to R_D , and provides a lot of degenerative feedback. And that's where C105 comes in. By placing an appropriately sized capacitor in parallel with the

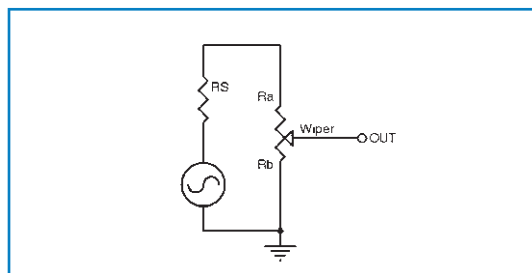


Figure 7: The output impedance is altered by the position of the volume control.

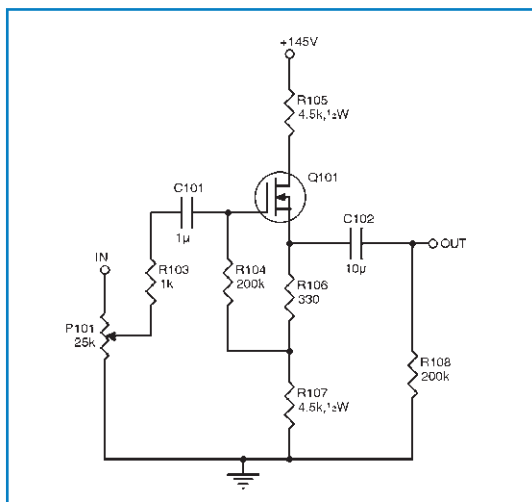


Figure 8: An optional buffer at the input drives the SIT, which provides the voltage gain and low output impedance.

source resistor, R_S is bypassed at audio frequencies and the calculation is reduced to:

$$A = \frac{\mu R_D}{R_D + r_d} = \frac{8 \times 600}{600 + 75} = 7.11$$

Now, the input signal is amplified 7.11 times or 17 dB. Some or all of the source resistance can be bypassed depending on how much gain is traded for feedback.

In this case, there isn't a lot of gain to start with, and the design goal calls for little or no feedback, so just bypass the whole resistance. To calculate

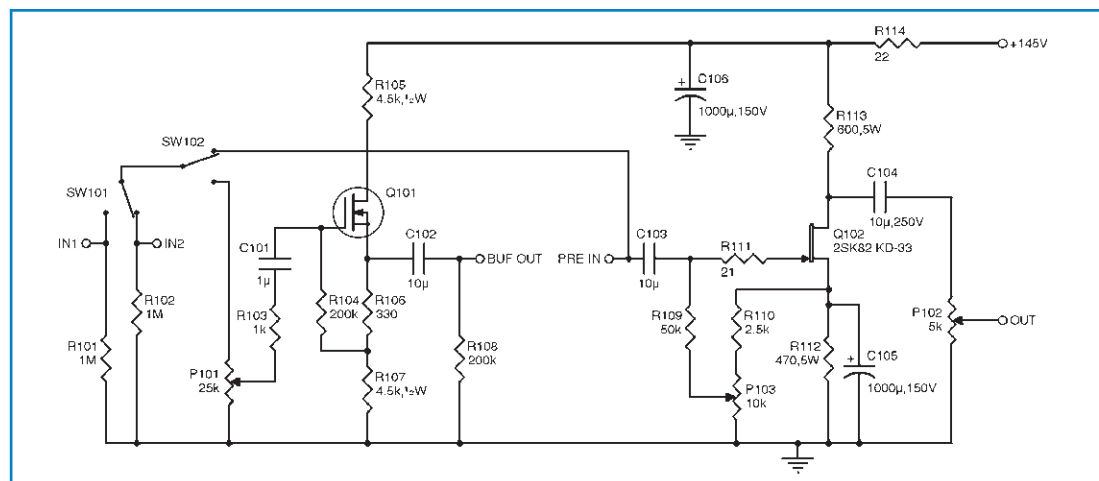


Figure 9: The complete schematic shows the buffer and gain sections with input selection and volume controls.

the value for the bypass capacitor, use the following formula:

$$C = \frac{1}{2\pi FR}$$

I used 2 Hz for the cutoff frequency. F and R are computed as follows:

$$R = r_s \parallel R_s = (71 \parallel 470) = 62 \Omega$$

where:

$$r_s = \frac{R_{LOAD} + r_D}{\mu + 1} = \frac{566 + 75}{8 + 1} = 71 \Omega$$

$$R_{LOAD} = \text{Drain Load} \parallel \text{Driven Stage Load} = (600 \parallel 10,000) = 566 \Omega$$

(I used 10,000 Ω for the driven stage load, assuming that to be the lower limit of what we're likely to encounter.) So:

$$C = \frac{1}{2\pi FR} = \frac{1}{6.28 \times 2 \times 62} = 0.00128 \text{ F}$$

That's 1,280 μF . I settled on 1,000 μF , which was a convenient on-hand value for a 2.6-Hz cutoff.

This circuit's output impedance is the drain load resistor in parallel with the resistance looking into the drain, which is why it is useful to have a low drain resistance.

$$R_D \parallel r_D = 600 \parallel 75 = 67 \Omega$$

R113 is an important resistor, as it will affect the stage's gain, operating point, distortion, and output impedance. The load line chosen for this stage depends on the design goals. If maximum gain and

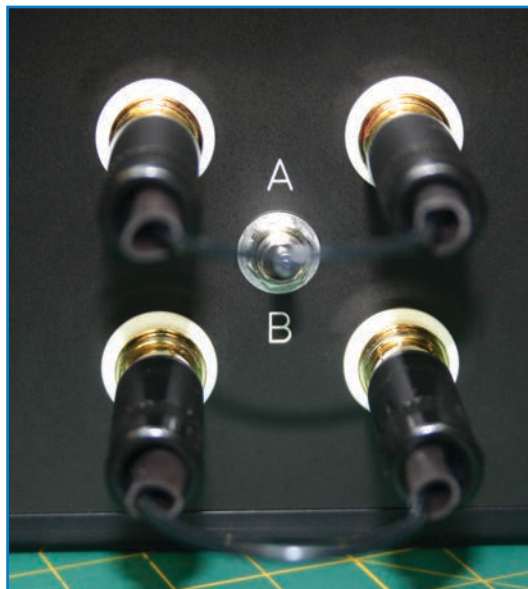


Photo 1: External jumpers enable the preamplifier sections to be used alone or in cascade.

Parts List

(Resistors are 25 W unless marked otherwise)

Preamplifier (One channel)	
Parts	Values
R101, R102	1 M Ω
R103	1 k Ω
R104	200 k Ω
R105, R107	4.5 k Ω , 0.5 W
R106	330 Ω (for 7 to 8mA)
R108	200 k Ω
R109	50 k Ω
R110	2.5 k Ω
R111	21 Ω
R112	470 Ω , 5 W
R113	600 Ω , 5 W
R114	22 Ω
C101	1 μF , 100 V
C102	10 μF , 100 V
C103	10 μF , 100 V
C104	10 μF , 250 V
C105	1,000 μF , 35 V
C106	1,000 μF , 150 V
P101	25 k Ω , Log or Lin
P102	5 k Ω , Log or Lin
P103	10 k Ω , 0.5 W
Q101	DN2540
Q102	2SK82 KD-33

Power Supply (One channel)	
Parts	Values
R101	2.5 k Ω
R102	221 Ω
R103	82k Ω , 0.5 W
C101	1,000 μF , 200 V
C102, C103	10 μF , 200 V
D101-D110	12 V, 0.5-W Zener
D111, D112	15 V, 0.5-W Zener
D113	20 V, 0.5-W Zener
D114	Green LED
BR101	Diode bridge
Transformer	2 $\times$ 115-to-120-V Secondaries 80+ VA

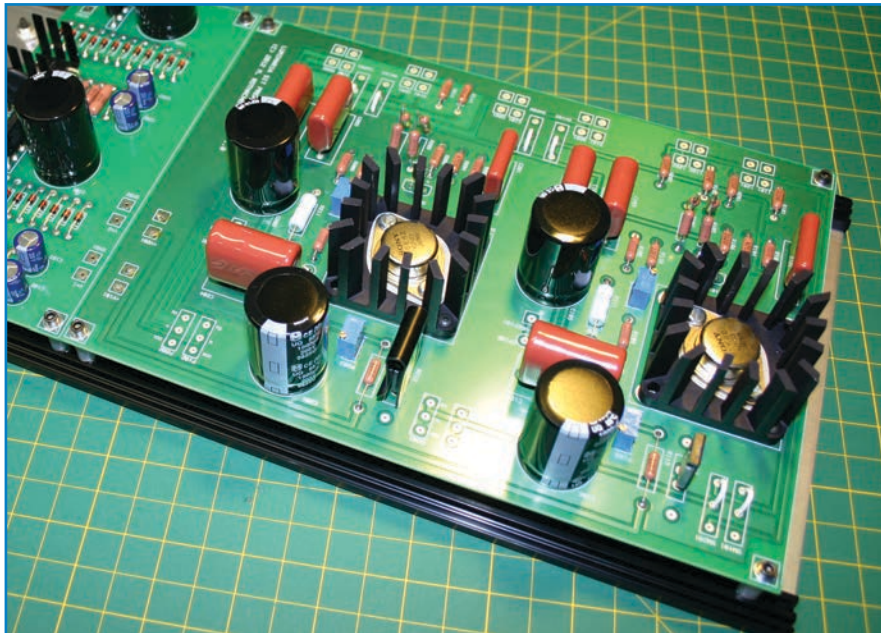


Photo 2: The prototype was built on a PCB.

voltage swing are desired, the load line is chosen to meet those criteria.

In this particular case, R113 was chosen for adequate gain with minimal distortion around 1 V. I arrived at the final value, 600 Ω , using a variable power resistor, which I tweaked while watching the distortion analyzer. The distortion is minimized or otherwise fine-tuned by load line cancellation, where gain variations due to drain current and drain voltage partially cancel each other, often to very good effect.

Ins and Outs

Figure 6 shows the basic preamplifier, which has the volume control potentiometer at its output rather than its input. This enables the preamplifier to look directly into the music source's output impedance, which typically is low. If the volume control potentiometer is instead placed at the input of the circuit, an undesirable high-frequency rolloff may be introduced. This would occur due to the RC filter formed by the SIT's rather high input capacitance and the volume control's output impedance.

A volume-control potentiometer's output impedance depends on the output impedance of the circuit driving it, the potentiometer's value, and the wiper's position (see **Figure 7**).

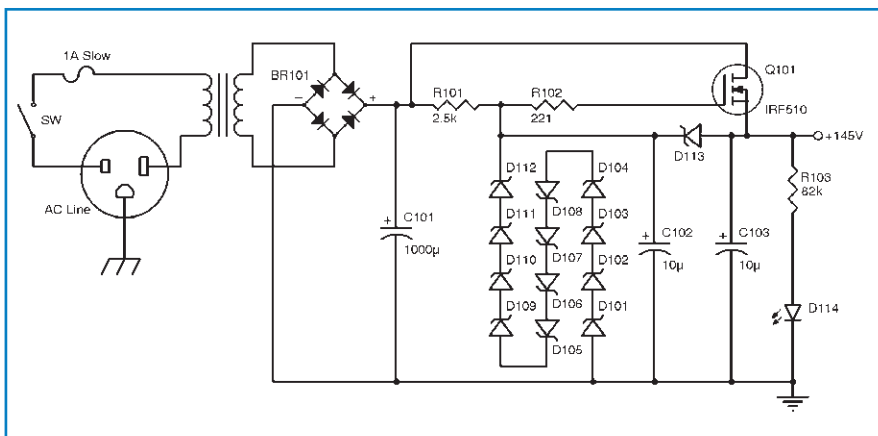


Figure 10: This schematic shows one channel of the power supply.

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$$R_{OUT} = (R_{SOURCE} + R_A) \parallel R_B$$

For example, a 25-k Ω linear volume potentiometer with its wiper at the midpoint, driven by a 100- Ω source, will have a 6,275- Ω output impedance. Assuming R_{SOURCE} is low, the output impedance will always be highest around the resistance midpoint of the potentiometer. With the potentiometer rotated fully counter-clockwise, the output impedance will be equal to R_{SOURCE} , and, when fully clockwise the wiper is grounded and the output impedance is zero.

If this 25-k Ω potentiometer were connected to the basic preamplifier circuit's input, the high-

frequency response will be -3 dB at less than 17 kHz when the volume knob is at 12:00, which doesn't meet the requirements. A lower value volume potentiometer could be used, but the source component may not drive the low impedance. However, with a 5-k Ω volume potentiometer at the output, the output impedance will vary from 0 to 1,500 Ω . At the 9:00 position, a typical (loud) setting, the output impedance will be around 560 Ω if a logarithmic potentiometer is used.

If the power amplifier doesn't object to this source impedance range, and the music source can handle the 1,500-pf input capacitance, this basic preamplifier will work. However, something a little more versatile may be preferable for use with a broader range of amplifiers.

Adding an Input Buffer

Most tube and transistor preamplifiers utilize a voltage gain stage coupled to an output buffer stage, which lowers the output impedance to effectively drive the power amplifier. LuminAria flips this around and offers an optional buffer at the input driving the exotic gain device, which provides both the voltage gain and low output impedance (see **Figure 8**).

A buffer stage offers high input impedance and low output impedance. This enables us to add a high-resistance volume control potentiometer at the preamplifier's input while driving the SIT without adversely affecting the high-frequency response.

The circuit shown in **Figure 9** is a simple source follower using the Supertex DN2540 depletion MOSFET. The DN2540 doesn't make many appearances in linear amplifiers, but I wanted to give it a try in this 100% negative feedback buffer. This circuit's voltage gain is slightly less than unity (around 0.98):

$$A = \frac{\mu}{\mu + 1} \times \frac{R_S}{(R_S + r_D)} = \frac{4,500}{(4,500 + 75)} = 0.98$$

This circuit is self-biased by R106. The gate resistor R104 is bootstrapped, which causes the buffer's input impedance to be significantly increase:

$$r_{INPUT} = \frac{R_G}{1 - A \times \left(\frac{R_L}{R_L + R_K} \right)} = \frac{200,000}{1 - 0.98 \times 0.93} = 2.2 \text{ M}\Omega$$

The input capacitance is approximately:

$$C_{INPUT} = C_{DG} + (1 - A) \times C_{GS} = 100 \text{ pF} + (1 - 0.98) \times 100 \text{ pF} = 102 \text{ pF}$$

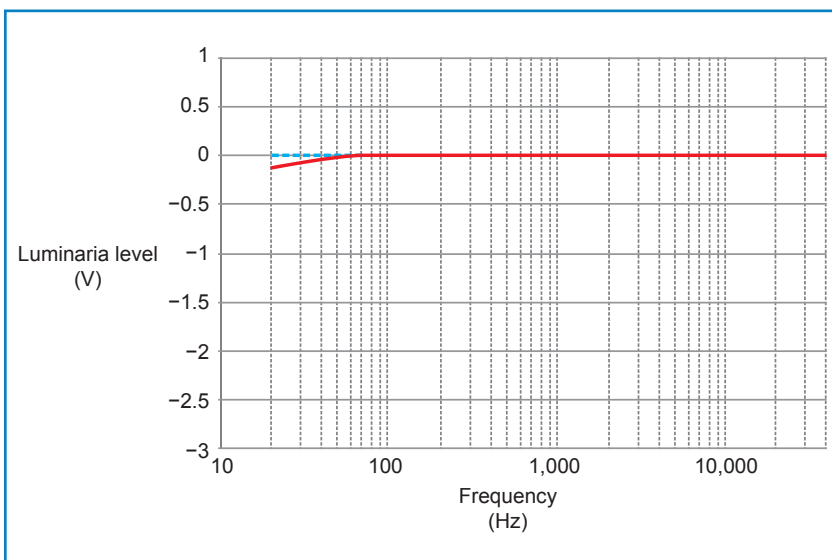


Figure 11: LuminAria's frequency response is shown.

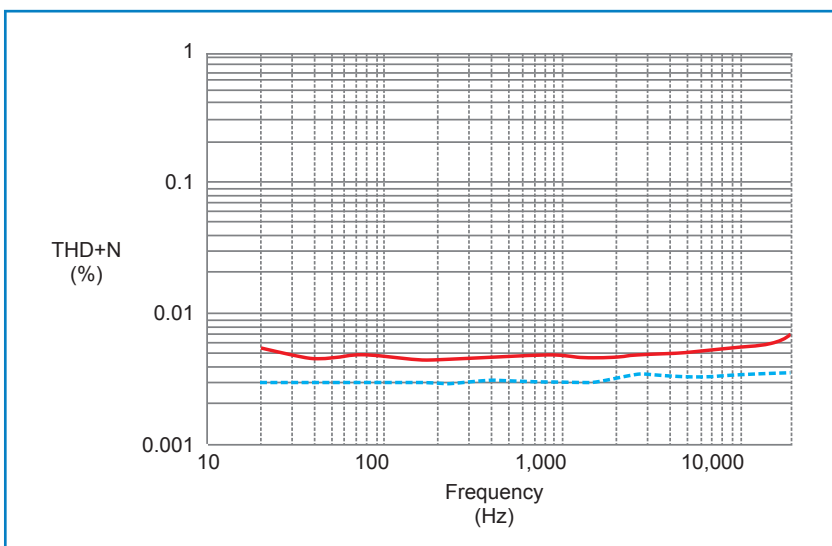
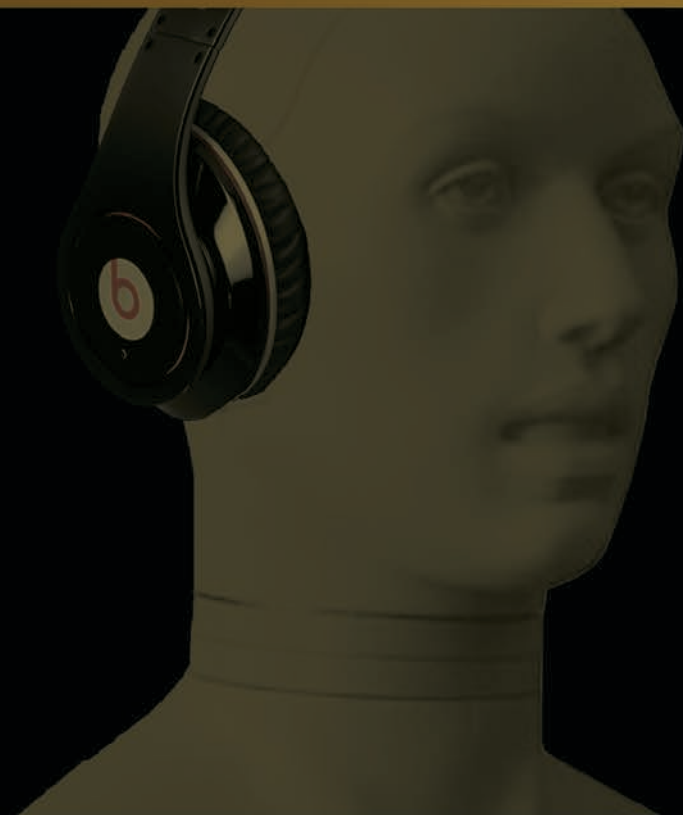


Figure 12: The distortion vs the frequency response is shown at 1 V.

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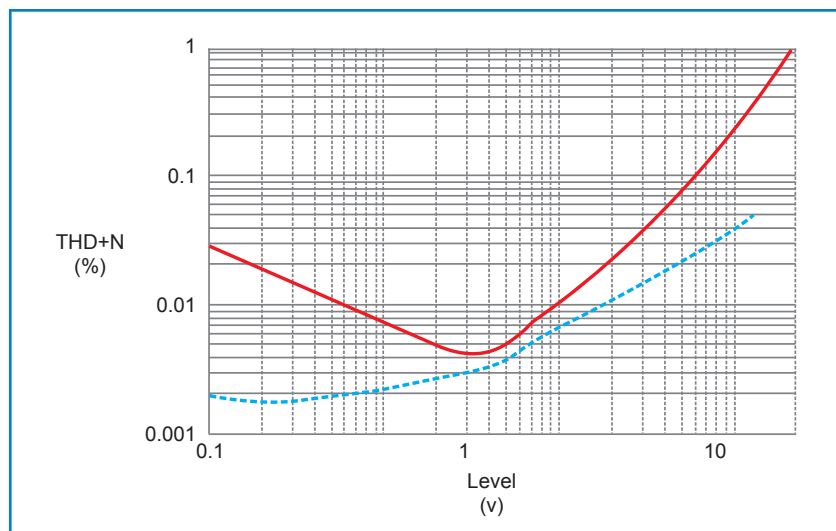


Figure 13: Harmonic distortion and noise are around 0.004% at 1 V.

The output impedance is approximately equal to the MOSFET's inverse transconductance, or more specifically:

$$\frac{R_s}{G_m R_s + 1} = \frac{4,500}{0.03 \times 4,500 + 1} = 33 \Omega$$

The high supply voltage enables us to use a higher value for R107, which reduces distortion in addition to raising the input impedance. The result is a pretty simple high-swing buffer with 0.003% THD+N at 1 V, and less than 0.03% THD+N at more than 10 V.

Build Notes

Figure 9 shows the complete schematic for one channel as built. There are just a couple of new parts here. The SW101 is for source selection and the R114 and the C106 provide some additional power supply filtration. The buffer and output sections are not internally wired together. This makes it possible to use the two stages separately, or in cascade by removing or inserting a jumper between the buffer's output RCA and the SIT stage's input RCA (see **Photo 1**). SW102 connects the appropriate stage to the input selection depending on whether or not the jumpers are present.

For the 2SK82, I used grade KD-33, which indicates a certain range of pinch-off voltage. Other grades will have different pinch off voltages and may or may not work properly in the circuit as shown. The SIT must be attached to a suitable heatsink as it will dissipate 5 W or so.

I built my prototype on a PCB (see **Photo 2**). This is not strictly necessary and you could certainly build

this circuit point to point. I used linear-type volume controls from PEC, though log potentiometers would provide more fine control. You could use a higher value for P101 if you want to; 50 kΩ would not be unreasonable.

Figure 10 shows one channel of the regulated power supply I used. An isolation-type transformer with 115-to-120-V secondaries rated at 80 VA or more can be utilized to provide the unregulated voltage (around 160 V), which should simplify the parts search.

R101 feeds current through Zener diodes D101–112, which provide a 150-V reference to an N-channel MOSFET voltage follower. The reference voltage minus Q101's gate-to-source voltage appears at the regulator's output filtered by C101–103. D113 prevents Q101's gate-to-source voltage from exceeding the device rating. R102 prevents parasitic oscillation of the MOSFET. R103 and D114 provide the glowing green assurance that the preamplifier is powered on.

Only a couple adjustments are needed. R106 is shown as 330 Ω, but it should be sized to give 7 to 8 mA through the MOSFET, or the MOSFET can be selected to provide 7 to 8 mA with a 330-Ω resistor. Once built, you can adjust the SIT's idle current for 50 to 60 mA with P103. Ideally, use a distortion analyzer or fast Fourier transform (FFT) analysis to reach the final operating point.

Usage and Performance

The LuminAria can be configured a few different ways. When you don't need gain, configure the preamplifier as a buffer with input selection and volume control. For use as a buffer, no jumpers are installed, the output is taken from the "Buffer Out" RCAs, SW102 is switched to position B, and volume is controlled with P101.

If your music source has an output impedance of a few hundred ohms or less and can source a couple of milliamps, try listening to the SIT by itself. In this configuration, no jumpers are installed, SW102 is switched to the A position, output is taken from the preamplifier output RCAs, and volume is controlled with P102. As previously mentioned, the amplifier's frequency response may be negatively affected by the position of P102 if the input capacitance is very high.

The most versatile configuration cascades both amplifier sections, exploiting their benefits. In the cascade configuration, the jumpers are installed, output is taken from the preamplifier outputs, and SW102 is set to position B. Volume is controlled with P101. Set P102 to the fully clockwise position where the output impedance will be 67 Ω, though

About the Author

Michael Rothacher is an executive for a US manufacturing firm by day, and a mad scientist by night. His interests include audio circuit experimentation, circuit-bending, computer programming, and writing.

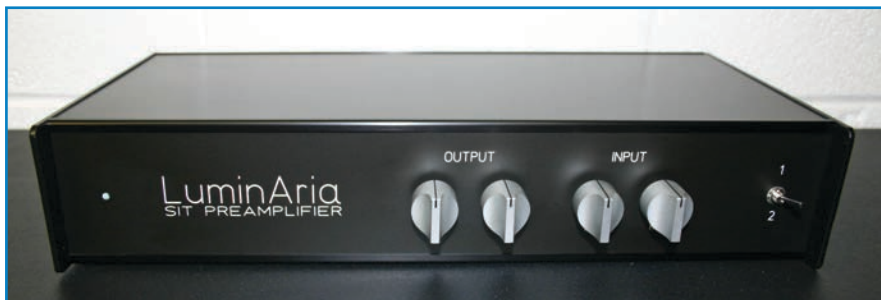


Photo 3: The LuminAria has a simple control interface.

you can experiment with its setting.

Figure 11 shows the LuminAria's frequency response. The dashed line is the buffer section, while the solid line represents both stages in cascade. We're down 0.15 dB at 20 Hz and flat beyond 40 kHz with a 600-Ω source.

Figure 12 shows THD+N vs frequency response at 1 V. Again, the dashed line is the buffer section, while the solid line represents both stages in cascade. Both sections exhibit distortion performance that's fairly flat across the audio spectrum.

Figure 13 shows THD+N vs output level. Harmonic distortion and noise are around 0.004% at 1 V and remain below 0.01% between 0.4 and 2 V where most of the action will take place.

Performance

The LuminAria was a really fun project (see **Photo 3**). Ultimately, it did make the journey to San Francisco where it was displayed with equipment from audio greats

such as Nelson Pass, Wayne Colburn, and many of my DIY friends.

Perhaps you're wondering how it sounds? Well, I will tell you that it has stayed in my equipment rack for six or seven months as of this writing, and I've never been tempted to put my commercial preamplifier back into service.

The LuminAria is a very simple design, but I wouldn't call it a beginner project. The SITs are expensive and ought to be gain matched for best results. It utilizes a high-voltage power supply and the final adjustment requires a distortion or spectrum analyzer.

If you need an unusual preamplifier with a lot of output, low distortion, and low output impedance, perhaps you'll build this one. If you do, I hope you'll agree that it's a neat gadget. There are many ways to make a preamplifier that sounds good and measures well, but this one's unique. Now go build something. 

Resources

N. Pass, "SIT Introduction," *First Watt*, 2011, www.firstwatt.com.

M. Rothacher, "L'Amp: A Simple SIT Amp," *diyAudio*, 2011, www.diyaudio.com.

Sources

2SK82 KD-33 Static induction transistors
Acronman | www.circuitdiy.com

DN2540 Depletion MOSFET
Supertex | www.supertex.com



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An Alternative to Linear Regulators

Equivalent Power-Line Ripple Rejection with Less Power Dissipation

By
Bill Reeve
(United States)

Hum is usually caused by 60- and 120-Hz ripple on the amplifier's unregulated DC power supply and that amplifier's imperfect ability to reject power supply ripple. No one wants to listen to an audio amplifier that hums.

The desire to eliminate power supply ripple tempts amplifier designers to regulate their power supplies, and negative feedback linear regulators do effectively reduce hum extremely. However, even for moderately powerful amps, linear regulators dissipate a prohibitively large amount of power. **Figure 1** shows a typical unregulated power supply.

Any unregulated power supply has output impedance. So as load current increases, an unregulated power supply's output voltage drops. A linear regulator that follows this unregulated supply must have a set point at least its headroom requirement below the lowest voltage the unregulated supply will drop to under its heaviest expected load.

For example, the output voltage of an unregulated supply that sources 60 V under a light load typically drops below 50 V under a 2-A load. A linear regulator

designed for this system may need to have an output voltage of approximately 48 V to avoid going out of regulation. Under a 1-A load, the power dissipated by the linear regulator would be approximately 7 W (i.e., $55 \text{ V} - 48 \text{ V} \times 1 \text{ A}$).

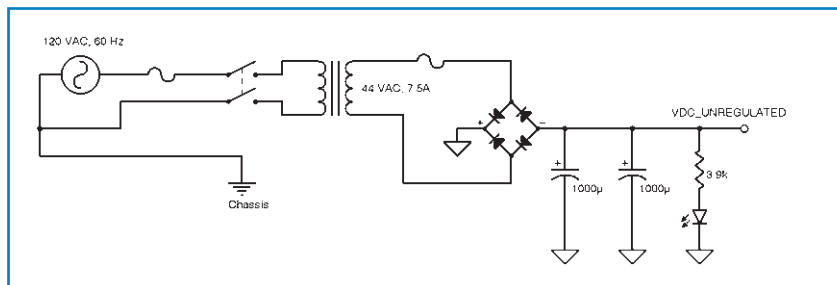
As an alternative, a source follower-based (SFB) low-pass filter provides good power-line ripple rejection while dissipating much less power—no more than 2 W (see **Figure 2**).

Figure 3 shows a power dissipation comparison between a linear regulator set to 48 V and an SFB filter using a logic-level MOSFET with a 2-V gate-source threshold. The comparison assumes that the power source is the unregulated power supply shown in **Figure 1**. This supply has a no-load output voltage of about 60 V and a 5- Ω output impedance. Note that the SFB filter's power dissipation linearly increases with the load current. (It is simply the pass transistor's gate-source threshold voltage times the load current.) The linear regulator's power dissipation peaks at medium loads.

How It Works

The SFB filter's transistor acts as a source follower. Its output (source) voltage is following a two-pole, low-pass filtered version of its input (drain) voltage, so the majority of the ripple on the SFB filter's input

Figure 1: A conventional, unregulated DC power supply outputs about 2 V_{pp} voltage ripple at 120 Hz on top of about 55 VDC under an amp of load.



is rejected at its output. Unlike a negative-feedback linear regulator, an SFB filter does not hold its output voltage constant. The SFB filter's output (source) voltage floats approximately the MOSFET's gate-source threshold voltage below its average input (drain) voltage—minimizing power dissipation.

MOSFET power transistors with low (logic level) gate-source threshold voltages are key to the SFB filter's low-power dissipation. Maximum gate-source threshold voltages for logic-level MOSFETs range from 1 to 2 V. Typical power MOSFETs (e.g., the IRF540N) have a threshold voltage of approximately 4 V, so power dissipation is four times as high.

A bipolar Darlington transistor can be used to build an equivalent filter (see **Figure 4**). It has almost identical ripple rejection, but higher power dissipation than the logic-level MOSFET-based filter. Its output voltage floats approximately 3 V below its average input voltage.

Design Considerations

Stability and Transistor Protection—The MOSFET's 1-k Ω gate resistor ensures stability under the large capacitive load the filter is driving (see **Figure 2**). The 10-V Zener diode protects the transistor from excessive gate-source voltage when the system is powered off under load. Of course, be sure to mount the transistor on an appropriately sized heatsink.

Filter Resistor Selection—Since the two-pole RC filter's attenuation increases at 40 dB per frequency decade, and we want to have at least 50-dB attenuation at 120 Hz, we would like to have a pole frequency at or below 1 Hz. The filter capacitors can easily be much larger than the 1,000- μ F values shown in the reference design.

Since this MOSFET transistor is a voltage-controlled device, large resistor values do not affect the transistor's performance. However, bipolar transistors are current-driven devices, and the filter resistors limit the transistor's base current. The SFB filter shown in **Figure 4** uses a Darlington transistor because its high current gain (Beta) enables it to pass large load currents with limited base current. If we had used a single bipolar power transistor, we would have had to use much smaller resistors—limiting the filter's ripple rejection performance.

Using resistors that are too large (10 k Ω or greater) with a Darlington transistor will choke the filter's ability to supply load current. Obviously, small-valued resistors (2 k Ω or less) will make the filter less effective.

Output Capacitors—As with a linear regulator, the SFB filter's output impedance increases with frequency, so the output capacitors shown in **Figure 2** and **Figure 4** are necessary to supply high-frequency currents to the load.

Ripple Rejection

The SFB filter does not maintain its ripple rejection performance at higher frequencies.

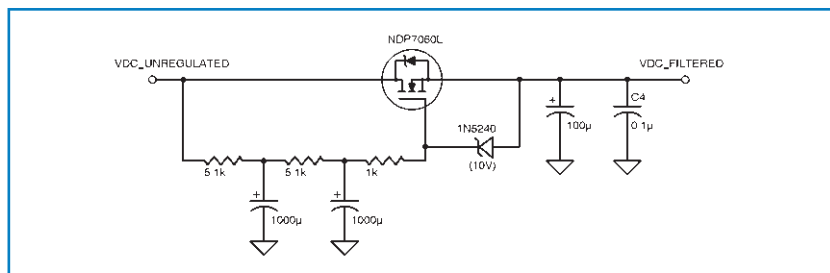


Figure 2: When combined with the unregulated power supply shown in Figure 1, the SFB low-pass filter system can easily provide 1 A at 55 V (55 W) to an amplifier output stage with a greater than 40 dB of ripple rejection (a factor of 100) at 60 and 120 Hz while the filter itself dissipates less than 2 W.

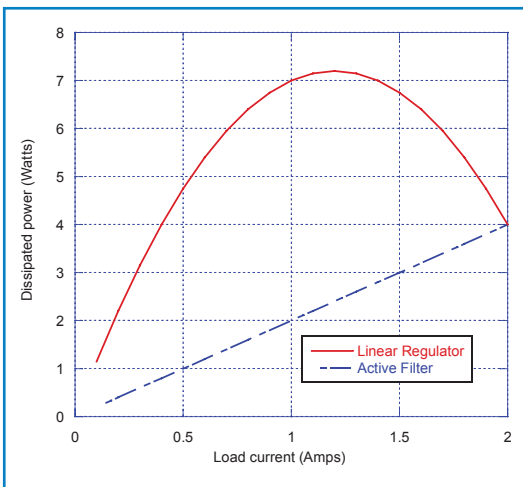


Figure 3: A power dissipation comparison between a linear regulator and an SFB filter is shown.



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Figure 4: This SFB low-pass filter is designed around a bipolar Darlington transistor.

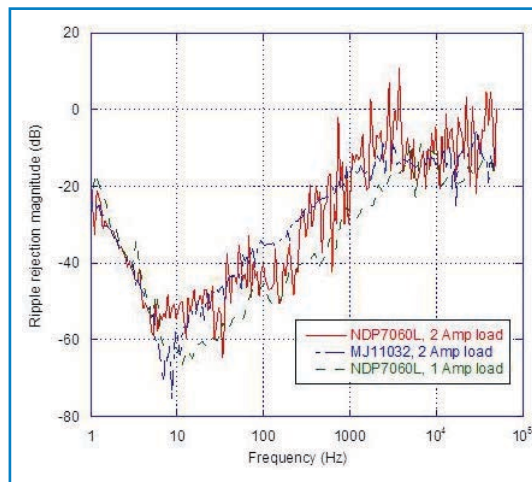
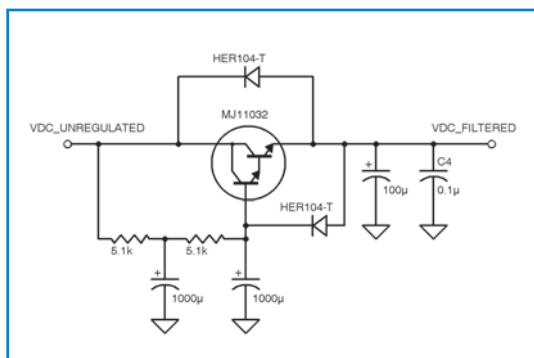


Figure 5: Ripple rejection, measured using the circuits shown in Figure 1, Figure 2, and Figure 4 shows the NDP7060L MOSFET at 1 A of load current (55 W to the load), 2 A of load current (80 W to the load), and the MJ11032 Darlington also 2 A (80 W to its load).

About the Author

Bill Reeve is a director at Lockheed Martin Corp. in Palo Alto, CA. He has graduate engineering degrees from the Colorado School of Mines, The University of California, Berkeley, as well as a Masters in Electrical Engineering from Santa Clara University.

Parasitic capacitance in the filter's high-power transistor limits the filter's frequency response. Output-voltage-to-input-voltage transfer function magnitude measurements show the SFB filter only tracks the two-pole low-pass filter feeding its base up to about 10 Hz (see **Figure 5**). Above that frequency, the transistor's frequency limitations lower the ripple rejection.

An Interesting Compromise

The SFB filter is an interesting compromise between an unregulated and a regulated supply for

audio amplifiers. It provides good power-line ripple rejection with less power dissipation than a traditional linear regulator.

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MUGEN HYBRID AMP



The Mugen Hybrid Amplifier's Driver Update

Photo 1: The Mugen Hybrid Amplifier got its name from the Japanese word "mugen," meaning infinite or endless.

My Mugen Hybrid Amplifier project published in *audioXpress* October 2013 relies on common, widely available tubes—the ECC83 and the ECC88—for the driver section. In this article, I am providing an alternative that will enable the distortion to become less than 0.1%, up to 80 W, without overall feedback.

By
Wim de Haan
(The Netherlands)

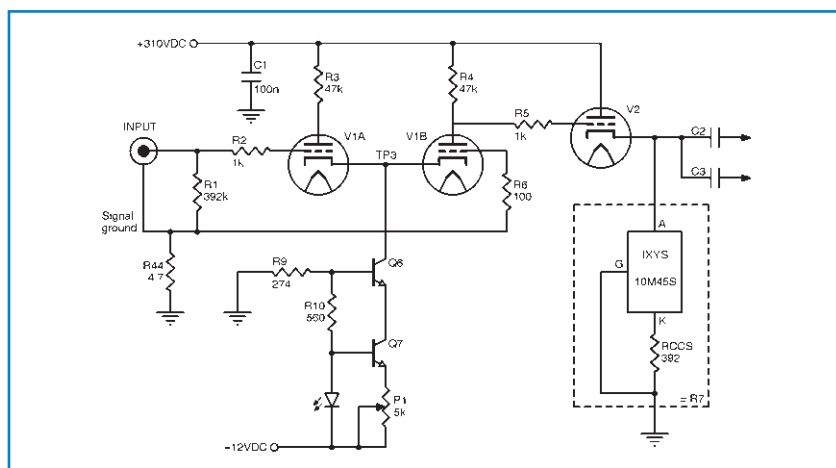


Figure 1: The design details for the updated Mugen Hybrid Amplifier are shown.



Photo 2: The amplifier design uses a combination of transistors and valves.

This can be achieved using IXYS's constant-current source (CCS) circuitry. I did not recommend this circuitry information in my original article because it was not widely available (even though the original prototype was tested that way). Many audio enthusiasts have discovered this CCS, IXYS's IXCP-10M45S, which can be used as a valid alternative to cathode resistor R7. The CCS can also be used with the existing circuitry, PCBs, and power supplies.

I built my Mugen amplifier in 2007 and originally published the design in *Elektron* (October 2007). The Mugen Hybrid Amplifier design replaces the output valves and transformer with a solid-state circuit using modern transistors, which can directly drive a loudspeaker (see **Photo 1**). However, the original prototype used well-priced uncommon and unknown tubes that had great specifications. The prototype used a 5965 tube for the long-tail driver and a 6BQ7A cathode follower.

The 5965 is well suitable for audio purposes. I used as long-tail driver (tube V1 shown in **Figure 1**). It provides enough gain (μ of 47), relative low output impedance, and low distortion. The 5965 is also pin-compatible with the ECC83 family. For optimal use the anode resistor (R3 and R4) is changed to 47K and the anode current is set to 3.5 mA. I used P1 to set the test point TP3 to 3 V. As a long-tail driver, the 5965 does not provide enough gain to be used with overall feedback. This is fine because I don't think overall feedback is necessary using these configurations (see **Photo 2**).

The 6BQ7A has good specifications and is an excellent choice as cathode follower (see **Photo 3**).

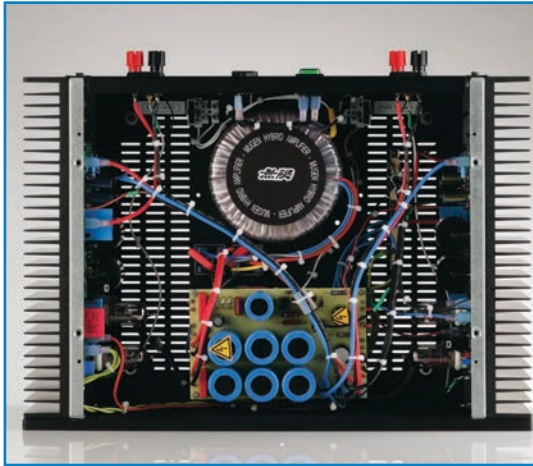


Photo 3: Once assembled, the amplifier's hybrid design can be fully appreciated.

It is also pin compatible with the ECC88 and it can be used for V2 without any problems. In this design, the 6BQ7A only uses one half of the V2 (see **Figure 1**).

The original design had another (major) feature. Instead of a resistor in the cathode follower section (R7), I used the IXCP-10M45S as a current source (see **Photo 4**). I did not use the IXCP-10M45S regulator in the design discussed in the *audioXpress* article because it was difficult to find and would make it challenging to build the amplifier. However, if you use the IXCP-10M45S as the current source, the amplifier's output power measures less than 0.1% distortion, and this would withstand up to 80 W without overall feedback. The IXYS IXCP-10M45S is often available on eBay.

In **Table 1** and **Table 2**, I have provided various configurations to show the distortion measurements. I used an experimental setup with a National VP 7701A distortion analyzer and a Philips PM5109S RC generator for the measurements. (I obtained the measurements with a 5-Ω load.) Note: No overall feedback was applied. The circuitry's part numbering conforms to the original article. 

Resources

Elektor, PCB layout, www.elektor.com/magazines/2007/october/mugen-a-hybrid-audio-amplifier.254273.lynxk.

W. de Haan, "Mugen—A Hybrid Amplifier," *audioXpress*, October 2013.

Sources

IXCP-10M45S CCS Regulator

IXYS Corp. | www.ixys.com

5965 and 6BQ7A Vacuum tubes

Available from BTB Audio (www.btb-elecktronik.de), eBay (www.ebay.com), Tube Depot (www.tubedepot.com), and The Tube Store (www.thetubestore.com)



Photo 4: The IXYS-10M45S current-source regulator can also be used in the Mugen amplifier.

About the Author

Wim de Haan is a medical engineer with a degree in electronics. He started building amplifiers because store-bought equipment did not meet his demands or his wallet. He likes classical piano music as well as Keith Jarrett and Michel Petrucciani. He is the webmaster for the websites of pianists Mikhail Voskresensky, Evelina Vorontsova, and Russian composer and pianist Nikolai Kapustin.

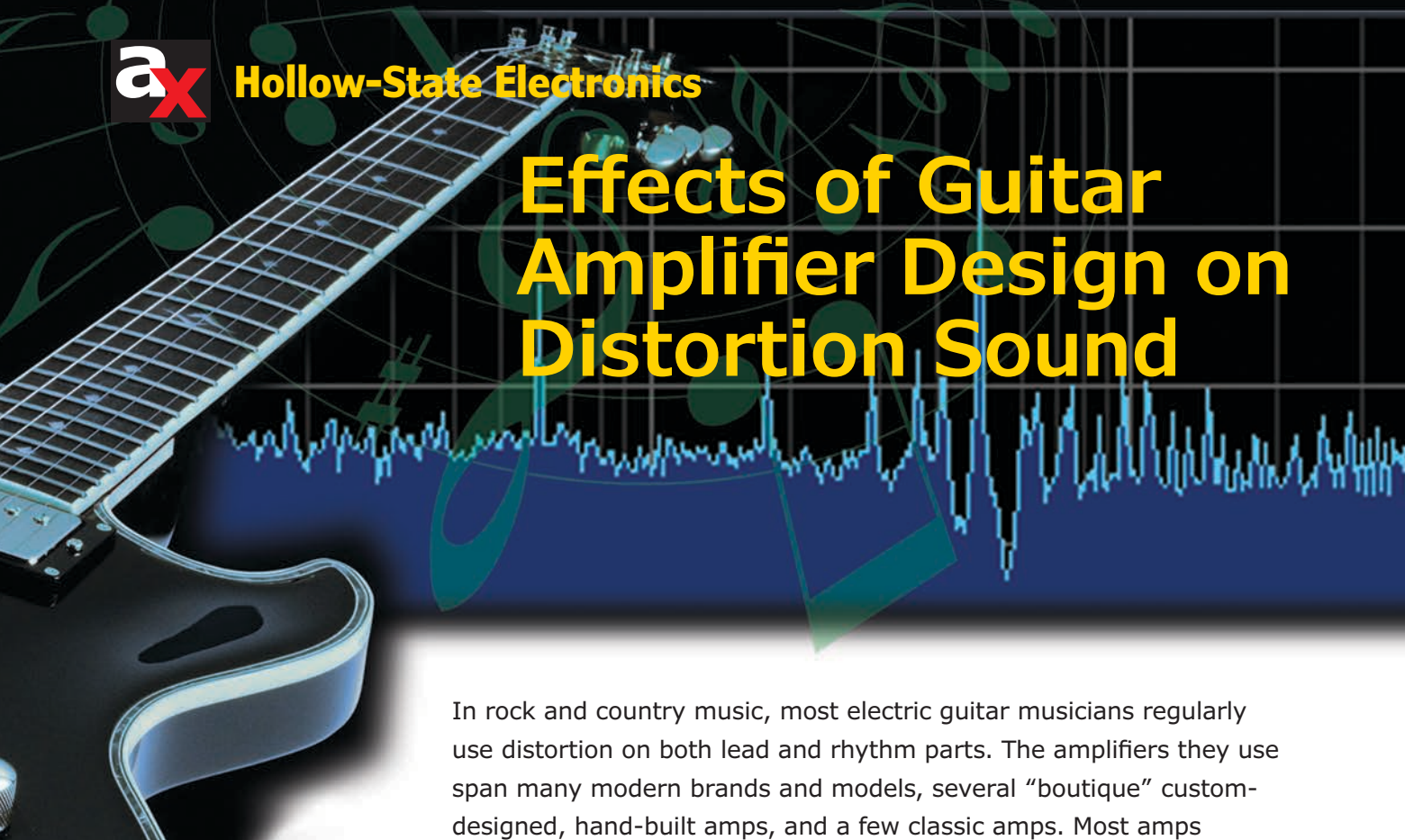
V1 = 5965 with R3 and R4 = 47 kΩ/2 W		V2 with R7 = 22 kΩ/3 W		
U _v = 310 V	U _{aV1a,b} = 145 V I _{V1a,b} = 3.5 mA TP3 = 3 V	U _{kV2} = 148 V I _{V2} = 6.7 mA	Gain = 22 dB	
		Configuration 1	Configuration 2	Configuration 3
Output voltage	Output power	Distortion V2 6N1P	Distortion V2 6BQ7a	Distortion V2 6CG7/6FQ7
2.3 V _{EFF}	1 W	0.072%	0.065%	0.052%
5 V _{EFF}	5 W	0.074%	0.055%	0.058%
10 V _{EFF}	20 W	0.045%	0.075%	0.125%
15 V _{EFF}	45 W	0.08%	0.17%	0.26%
20 V _{EFF}	80 W	0.28%	0.36%	0.45%
22.3 V _{EFF}	100 W	0.64%	0.55%	0.65%

Table 1: Configuration 1 shows the distortion results using a 6N1P preamplifier vacuum tube. Configuration 2 shows the distortion results using the 6BQ7 cathode follower. Configuration 3 shows the distortion results using 6CG7/6FQ7 preamplifier vacuum tubes.

V1 = 5965 with R3 and R4 = 47 kΩ/2 W		V2 with R7 = 22 kΩ/3 W		
U _v = 310 V	U _{aV1a,b} = 145 V I _{V1a,b} = 3.5 mA TP3 = 3 V	U _{kV2} = 148 V I _{V2} = 6.7 mA	Gain = 22 dB	
Output voltage	Output power	Distortion V2 6N1P	Distortion V2 6BQ7a	Distortion V2 6CG7/6FQ7
2.3 V _{EFF}	1 W	0.08 %	0.08%	0.07%
5 V _{EFF}	5 W	0.1 %	0.85%	0.075%
10 V _{EFF}	20 W	0.09 %	0.65%	0.06%
15 V _{EFF}	45 W	0.082 %	0.05%	0.04%
20 V _{EFF}	80 W	0.013 %	0.09%	0.08%
22.3 V _{EFF}	100 W	0.3 %	0.22 %	0.26%
IXYS* = IXCP-10M45S (using a TO-220 heatsink) using Rset IXYS = 392E				

Table 2: The Mugen Hybrid Amplifier's distortion levels are shown when using the 5965 tube vs the IXCP-10M45S.

Effects of Guitar Amplifier Design on Distortion Sound



By
**Richard
Honeycutt**
(United States)

In rock and country music, most electric guitar musicians regularly use distortion on both lead and rhythm parts. The amplifiers they use span many modern brands and models, several “boutique” custom-designed, hand-built amps, and a few classic amps. Most amps preferred by professional guitarists are hollow-state models. Each guitarist will choose a particular amp based partially upon preferences for certain features, but mainly upon preferences for a certain sound.



Photo 1: The Fender Pro Junior is a 15-W tube amplifier with a traditional design.

As we have discussed in recent Hollow-State Electronics articles, the sound depends upon several design features, among which is the distortion signature. An entire industry has grown up around digitally based “plug-ins” for performance and recording equipment, seeking to emulate the sound—including the distortion signature—of specific highly-regarded amplifiers. The various plug-ins achieve varying degrees of success.

In this article, we will examine one particular amplifier’s design and see at what stages most of the distortion occurs. We will also discuss the distortion’s frequency composition.

Several amplifiers could have been chosen for this analysis: a classic concert amplifier from a major manufacturer, an amplifier incorporating a master volume control to permit overdriving the preamplifier while playing at reasonable levels, an amplifier with switchable high- and low-gain input channels, an amplifier with one of the many popular modifications, or even a boutique amplifier. We chose an amplifier that represents an example of good generic traditional tube-amplifier design: The Fender Pro Junior (see **Photo 1**). This small amplifier has a straightforward design incorporating one input, a volume and a tone control, and a single 10” speaker. It is advertised at 15-W output, using a pair of EL84 (6BQ5) power pentode output tubes.



Figure 1: The 35-mV sine wave was fed to the preamplifier.

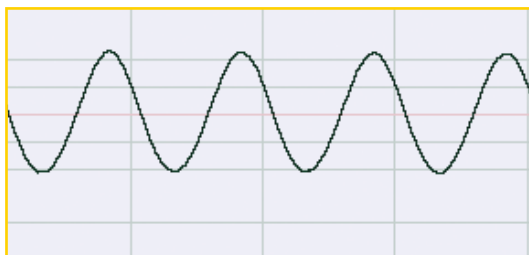


Figure 2: The output wave at the preamplifier triode's plate is practically perfect.

Preamplifier

The preamplifier stage consists of half of a 12AX7A dual triode in a cathode-biased configuration fed from a 232-V B+ supply. The plate resistor is 100 k Ω . The cathode resistor is 1.5 k Ω and is bypassed. The stage gain is 57. When the preamplifier is driven by the 63-mV, 500-Hz sine wave its output has about 25% total harmonic distortion (see **Figure 1**). The input wave's distortion was practically unmeasurable. The output at the preamplifier's plate is still quite clean (see **Figure 2**). Frequency analysis of the preamplifier's output wave shows a few power-line-related spikes plus a bit of second harmonic (see **Figure 3**).

If the preamplifier were overdriven enough that the preamplifier tube distorted significantly, the volume control would have to be set very low. Otherwise, output-stage distortion would be extreme. We will examine this later.

Tone Controls

The Pro Junior has a volume control and a single tone control. When these are set at 50%, the 500-Hz wave fed into the driver triode is about 43 dB below the preamplifier's output voltage. Since the controls are passive, they do not generate any nonlinear distortion. However, they can modify the distortion signature by attenuating the harmonics more or less than the fundamental. **Figure 4** and **Figure 5** show the waveform after the volume and tone controls and the frequency analysis of that waveform.

The main difference we see when we compare **Figure 3** and **Figure 5** is that the wave after the controls has more power-line-related noise picked up in the recording equipment due to the lab environment in which the very-low-level signal was recorded.

Driver

The driver stage consists of the other half of the 12AX7A whose first half served as the preamplifier triode. The design is the same as that of the preamplifier, except that the cathode resistor is not bypassed, giving a lower gain of about 30. When driven hard with a high volume setting and a high tone-control setting, the unbypassed cathode

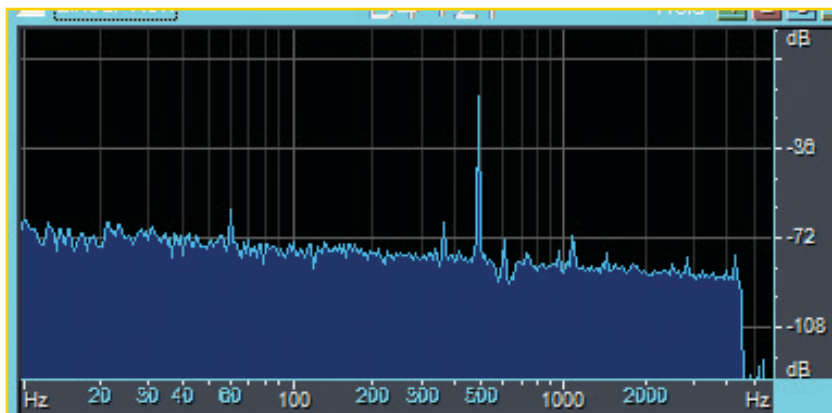


Figure 3: The frequency analysis of the preamplifier's output wave shows very little distortion.

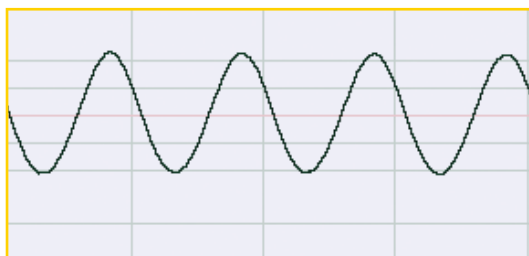


Figure 4: After the volume and tone controls, the waveform is still almost perfect.

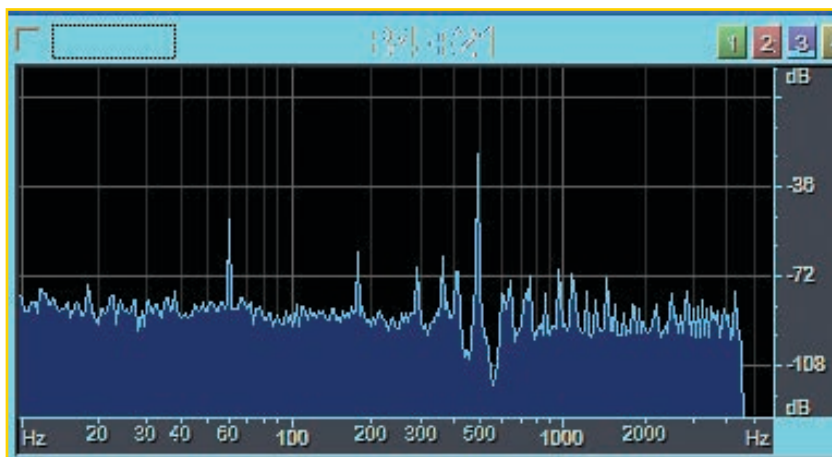


Figure 5: The frequency analysis of the wave after the volume and tone controls still shows little distortion.

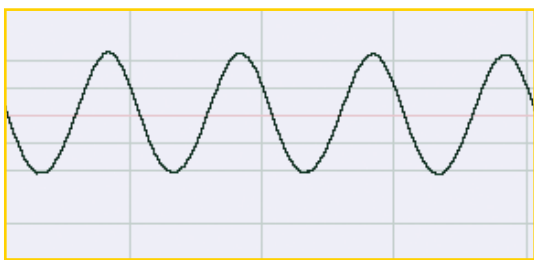


Figure 6: The driver stage's output wave is still relatively distortion free.

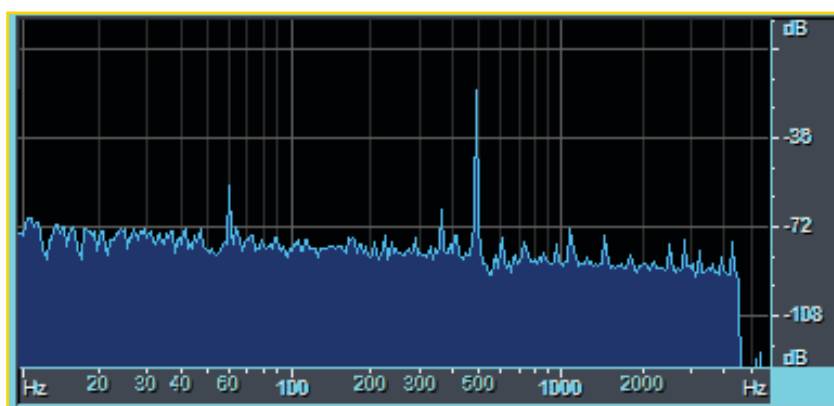


Figure 7: The output wave from the driver stage is quite clean.

resistor would cause the driver to produce less distortion than if it were bypassed. As expected, the output waveform and frequency analysis in **Figure 6** and **Figure 7** show little distortion.

Phase Splitter

The phase splitter is a long-tailed pair using both sections of a 12AX7A dual triode. In some tube guitar amplifiers, the phase splitter is the stage responsible for most of the distortion. In the Pro Junior, the output waveform still looks clean (see **Figure 8**).

However, the frequency analysis reveals some distortion, amounting to about 4% THD (see **Figure 9**). In amplifiers designed for minimum distortion (e.g., for home music systems), the phase-splitter stage is quite challenging to design for low distortion. The output wave from the phase-splitter is taken from plate to plate, not from one plate to ground. This cancels even-order harmonics produced in the phase splitter.

To accurately show the waveform, it was necessary to photograph the trace from a dual-trace oscilloscope, with channel 1 fed from the top tri-

ode's plate and channel 2 fed from the bottom triode's plate. Then, I inverted trace #2 and added, producing the same results as when the output stage feeds the output transformer. Because it was taken from an oscilloscope trace rather than from the screen of a digital recorder, **Figure 8** has a different appearance than the preceding waveform illustrations.

Push-Pull Output Stage

The Pro Junior uses a pair of EL84 (6BQ6) tubes connected in push-pull for the output stage. Tube manuals rate the maximum power output of an EL84 push-pull pair at 17 W with a 300-V B+ supply. Overall negative feedback is used from the output transformer secondary to the phase splitter's control-grid circuit. If the controls are set at 50%, most of the distortion will be generated in the output stage.

Figure 10 shows the output waveform at the plate of one of the EL84s. As with **Figure 8**, this oscilloscope trace is measured from the output

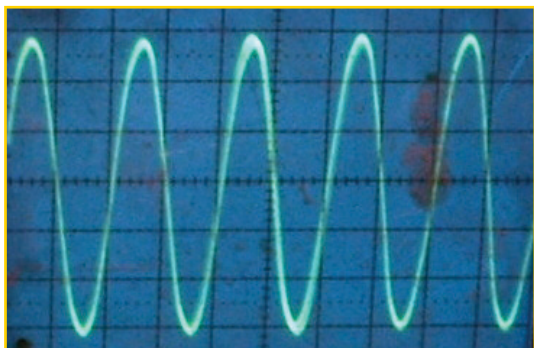


Figure 8: The phase-splitter stage's output wave still looks relatively distortion free.

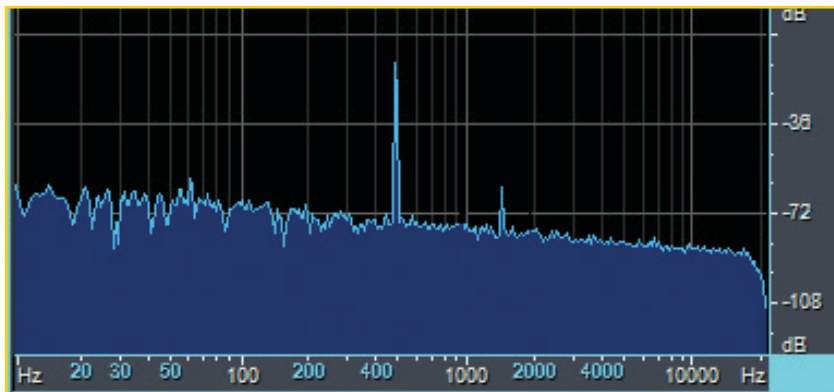


Figure 9: The phase-splitter adds some third-harmonic distortion to the waveform.

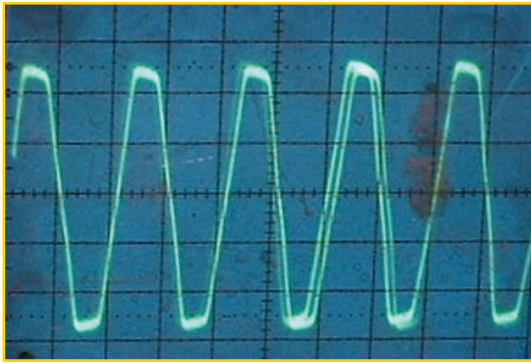


Figure 10: The output from the push-pull stage shows noticeable distortion.

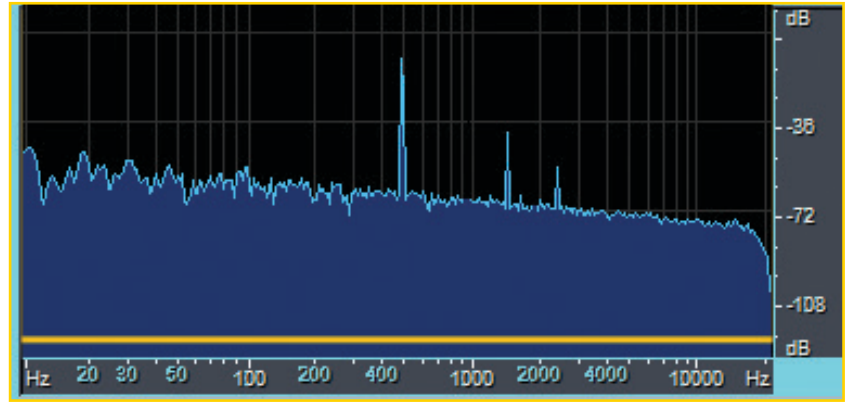


Figure 11: The harmonic levels are higher at the output of the push-pull stage than at the phase splitter's output.

stage's plate to plate. **Figure 11** shows the frequency analysis. Notice the higher levels of third and fifth harmonics (1,500 and 2,500 Hz), amounting to about 25% THD.

Output Transformer

Many articles and blogs have been written about the effect of output transformers on distortion. The

physics involved is somewhat complex, and the fine points of transformer design affect the results. However, it is interesting to compare the output waveform at the input to the transformer with the waveform at its output.

It turns out that the output wave from the transformer is almost indistinguishable from

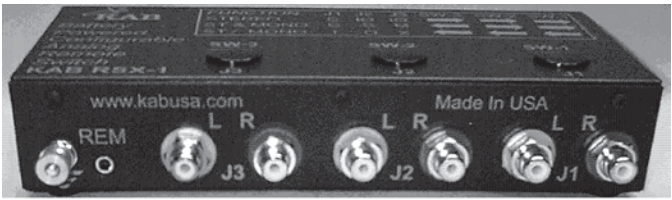


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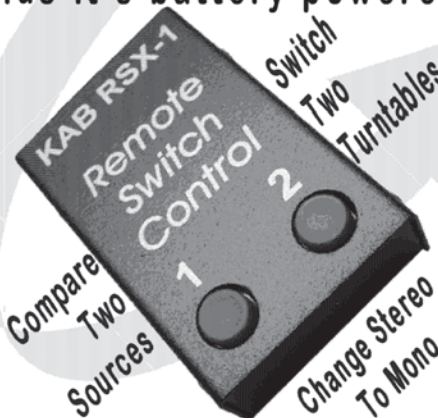
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Figure 12: When the preamplifier is overdriven, it produces this output waveform.

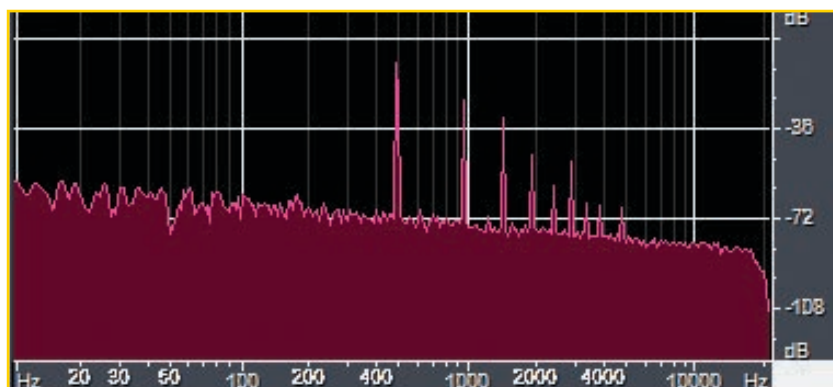


Figure 13: Overdriving the triode preamp stage produces odd- and even-order harmonics.

that at the EL84 plate. Frequency analysis shows at most a 1-dB increase in third-harmonic level.

Even these results must be viewed with caution, though, since the gain of the output stage depends upon the load impedance provided by the output transformer, so some nonlinearity in transformer

behavior could be reflected in the output stage's output waveform.

Overdriving the Input

The Pro Junior can produce a different distortion signature if the amplifier input is driven with

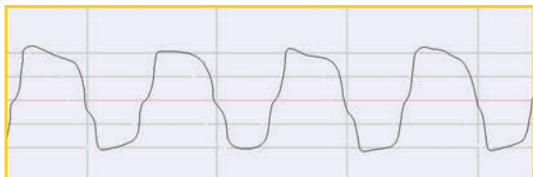


Figure 14: With the volume control at 3 and a 3.2-V input, the output looks like this.

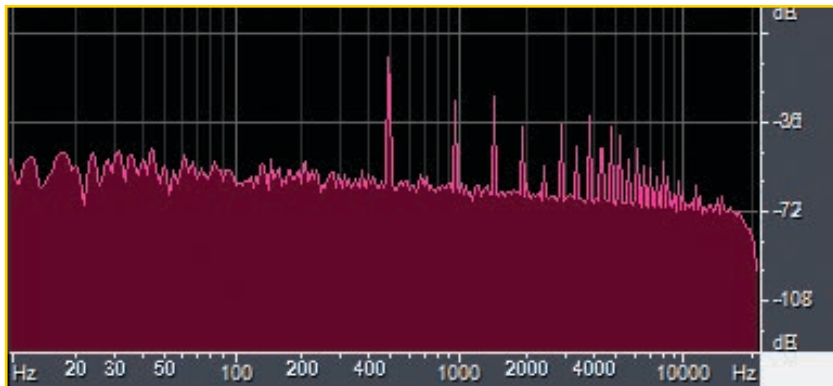


Figure 15: The frequency analysis shows many added harmonics.

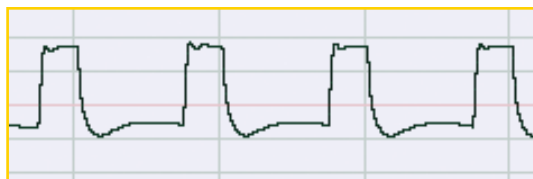


Figure 16: The GBX output contains a much higher proportion of upper harmonics than does the output from a tube amp.

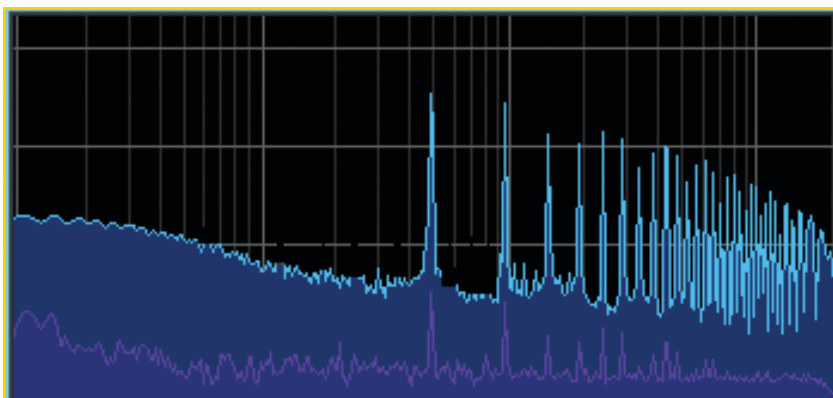


Figure 17: Rather than a square wave, the GBX produced a rectangular wave when overdriven.

a higher voltage and the volume control is turned down. If we apply a 3.2-V, 500-Hz wave to the amplifier input and turn the volume control down to 1.5, we can noticeably overdrive the preamplifier stage without causing much phase-splitter and output-stage distortion. Of course, a guitar will not put out 3.2 V, so an outboard preamplifier would have to be used to get this much input signal.

Figure 12 and **Figure 13** show the waveform and frequency analysis for the signal at the plate of the preamplifier tube. Notice the presence of both odd- and even-order harmonics.

If we look at the amplifier's output, we see an inverted version of the same wave, with almost no difference from the waveform at the preamplifier output. If we turn the volume control up to 3, the phase splitter and output stage add more distortion (see **Figure 14** and **Figure 15**). We can clearly see clipping on both the positive and the negative peaks, and also crossover distortion. The THD is now about 40%.

Thus, by driving the input hard and keeping the volume control setting low, we get a different distortion signature from the Pro Junior than if we use higher volume settings. However, an outboard preamplifier is needed to drive this amplifier hard enough to obtain the "gentler" preamplifier-stage distortion.

A Solid-State Comparison

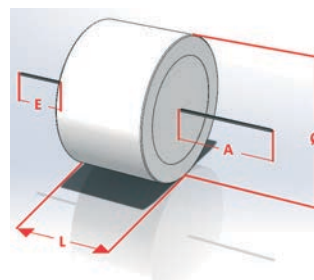
By the 1970s, most manufacturers of solid-state guitar amplifiers had developed ways to make the distortion of their amplifiers sound more agreeable. The early solid-state amplifiers clipped the sine waves into square waves when they were overdriven. One of the better manufacturers, GBX of Canada, produced an amplifier whose Effects input produced a much less buzzy distortion than did the earlier solid-state amplifiers. **Figure 16** and **Figure 17** show the output of one of the GBX amplifiers with its effects input overdriven.

As is usual for solid-state amplifiers—even the better ones—the GBX produces a much higher percentage of higher harmonics than does a tube amplifier. This fact is responsible for most of the difference in the distortion sound of the two amplifier types.

Although we have seen that the Fender Pro Junior produces most of its distortion in the output stage, with some contribution from the phase splitter, we have also seen that this amplifier could produce a different character of distortion if the input were overdriven by the use of an outboard preamplifier. Then the character of distortion would change at higher volume settings, as the output stage would contribute more of the distortion sound. Also, if the preamplifier is overdriven, the character of distortion can be varied by the tone control, which affects preamplifier distortion but not phase-splitter or output-stage distortion.

Test results included in this hollow-state electronics article only pertain to the amplifier tested. Other amplifier designs, the use of separate high-gain channels, and different kinds of tone controls, and so forth would have yielded much different results. 

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Member Profile

Member Name:

Jack Philpot

Location:

South Holland, IL

Occupation:

Jack is retired from AT&T. At one time, he was a frequent contributor to *Audio Amateur*. Jack's articles appeared in several *Audio Amateur* issues including February 1978, February 1979, February 1980, January 1983, February 1983, April 1983, February 1986, February 1988, April 1989. He also wrote an article for *Speaker Builder* that ran in January 1982.

Member Status:

Jack said he started subscribing to *audioXpress* when he discovered that it was a continuation of the *Audio Amateur* publication. He said he was hoping to find back issues that may contain an upgrade article on the Adcom GFP-565 preamplifier.

Audio Interests:

Jack is very interested in home audio, both new and vintage.



Most Recent Purchase:

He recently added a Cambridge DAC for for listening to high-resolution digital music sources and an Ortofon 2M Blue, designed for high-performance reproduction of analog records, to his home audio system.



Current Audio Projects:

Jack converted two dual-mono Adcom GFA-535 II power amplifiers to mono blocks with dual-power supplies. He used an extra pair of outputs from spare modules and added them to each driver board similar to the GFA-545 II. He built the unit so that the output terminals are bridged for bi-wiring. 

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1

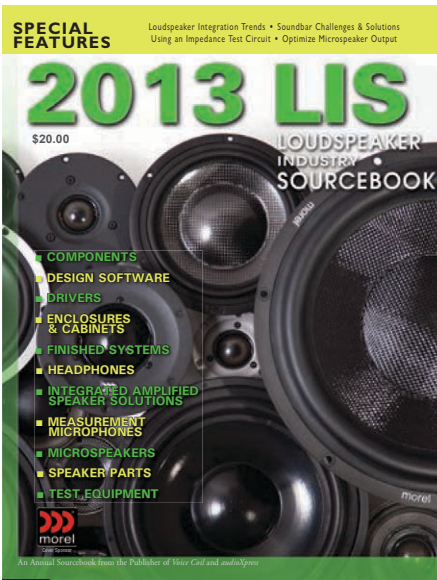
1 LINEAR AUDIO SERIES (VOLUMES 0-6)

Linear Audio is a series of bookzines full of unique content you won't find anywhere else. Each book offers everything from tutorials to circuit and system design, to test reports and book reviews. Why wait? Read, learn, and do it yourself! See the website for details on all seven editions.

2 LOUDSPEAKER INDUSTRY SOURCEBOOK 2013

The *2013 Loudspeaker Industry Sourcebook* is the most comprehensive collection of listings on loudspeaker materials ever assembled. Whether you're looking for design software, drivers, enclosures, manufacturing equipment, microspeakers, microphones, test and measurement equipment, or voice coils, this book has you covered. The new *2014 Loudspeaker Industry Sourcebook* is coming soon!

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2

3 FUNDAMENTAL AMPLIFIER TECHNIQUES

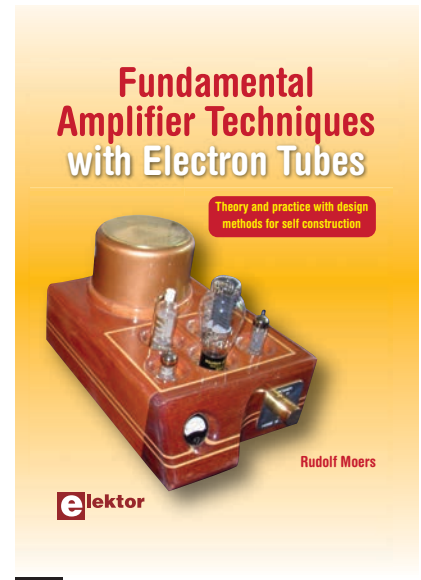
This book provides the reader with useful knowledge about electron tube technology in the application of audio amplifiers. Serving as a unique reference source, it provides insight as you learn critical techniques. Then you can build the circuit yourself. This is a must-have resource!

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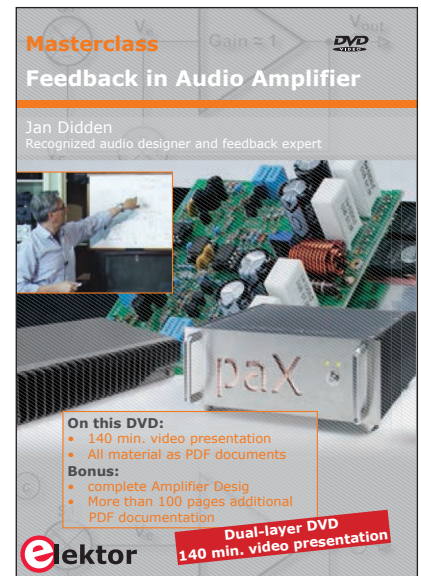
4 MASTERCLASS FEEDBACK IN AUDIO AMPLIFIERS DVD

The *Masterclass: Feedback in Audio Amplifiers* DVD provides a clear overview of the benefits that can be obtained from feedback and error correction. It also details their limitations and disadvantages. It is recommended to all audio designers and serious audio hobbyists!

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3



4

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April 5–10, 2014

Las Vegas Convention Center, Las Vegas, NV, US

www.nabshow.com

NAB Show 2014



The National Association of Broadcasters (NAB) Show is one of those not-to-be-missed opportunities where you can find many of the most important audio companies involved in the fields of recording, studio, post-production, film sound, consumer distribution technologies, acoustics, and so forth, in one location. The NAB Show is typically one of the most vibrant, exciting, and diversified exhibitions for the professional audio industry. Plus, it's certainly the most important annual event for the radio industry. With digital radio and new distribution models on mobile networks and online streaming on the horizon, radio certainly will be one of the hot topics for 2014. Most of the audio companies will be in the Central Hall, adjacent to registration and the main LVCC entrance.

April 26–29, 2014

Estrel Hotel and Convention Center, Berlin, Germany

www.aes.org

136th International AES Convention



The third Audio Engineering Society (AES) International Convention will include a full technical program of papers, posters, e-briefs, tutorials, and workshops covering a comprehensive range of audio-related topics, and feature top industry professionals. The 136th AES Convention, co-chaired by Sascha Spors and Umberto Zanghieri, will bring audio engineers from around the world to share the latest knowledge in audio research, development, and applications. Additionally, the Berlin convention will strongly focus on sound for picture, recording, and broadcast aspects of audio engineering, in addition to the popular off-site technical tours and social events.

Following the success of the Project Studio Expo (PSE) at AES conventions in San Francisco, CA, and New York, NY, the PSE will make its European debut at the 136th AES Convention. A special technology showcase will also provide participating companies with a chance to directly interface with interested end users and customers.

May 13–15, 2014

Expo Center Norte Pavilhão Amarelo, Vila Guilherme, São Paulo, Brazil

www.aesbrasilexpo.com.br

18th AES Brazil Expo



The Audio Engineering Society (AES) show in Brazil is unlike any other AES convention held in the US or elsewhere. In fact, even though the conference sessions follow the traditional association model—almost 100% in Portuguese—the exhibition floor is predominantly dedicated to sound reinforcement and PA systems due to the importance of that professional audio segment in the Brazilian market, which is one of the world's largest music markets. The event, held in the megalopolis of São Paulo—Brazil's business capital—is an important annual gathering for the domestic pro audio industry, including speaker component and amplification manufacturers. In fact, AES Brazil is the main pro audio event in South America and provides an opportunity to find local partners and learn more about Brazil.

May 26–29, 2014

China International Expo Center, Beijing, China

www.palmexpo.net

PALM Expo China 2014



China's most established international exhibition on Pro Audio, Light, Music & Technology (PALM) will celebrate its 23rd presentation at the new China International Exhibition Center, in Beijing. For the last 22 years, PALM Expo China has brought together many industry professionals, creating an excellent platform for business in the region. Promoted by Informa Exhibitions, the PALM Expo 2013 attracted more than 1,100 exhibitors and more than 50,000 visitors. Many of the visitors attended the event because of the conferences, seminars, and workshops. The PALM Expo 2014 is expected to reinforce the growth from previous years, showcasing the latest line of products and services from live sound, professional recording, sound reinforcement, disco/theater lighting, special effects, and visual and sound communications.

June 14–20, 2014

Las Vegas Convention Center, Las Vegas, NV, US

www.infocommshow.org

InfoComm 2014



The audiovisual communications marketplace, as the InfoComm association defines this major event, increasingly represents the intersection of AV and IT, technology and architecture, business and entertainment, government and hospitality, healthcare and education, live events, and installed systems. InfoComm is undoubtedly the most important professional AV show on the American continent. For 2014, more than 950 exhibitors from 300-plus audio companies and 35,000-plus attendees including more than 110 countries are expected. InfoComm 2014 will provide a unique opportunity to see the latest technology, learn new skills, and grow our professional network.

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