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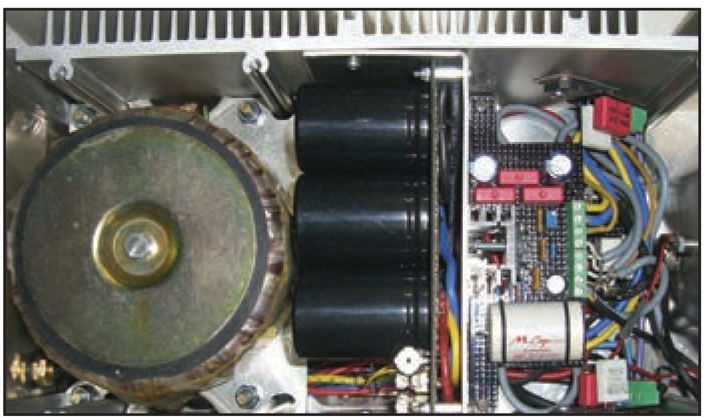

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Do You Want to Hear the Difference?

Why is it so easy to find an audio enthusiast claiming that his new headphones, DAC, preamplifier, or pair of speakers "completely changed his perception of the music" and "made it possible to hear things in a totally new way," and yet so many consumers cannot actually hear the difference when listening to high-resolution audio?

The difference in perceptions is precisely why I applaud initiatives such as the formal definition for High-Resolution Audio (HRA). HRA is defined as "lossless audio that is capable of reproducing the full range of sound from recordings that have been mastered from better-than-CD-quality music sources." The formal definition was finalized in June 2014 by The Digital Entertainment Group, in cooperation with the Consumer Electronics Association (CEA), The Recording Academy, Sony Music Entertainment, Universal Music Group, and Warner Music Group.

This represents a "promotional effort" by interested parties. But wasn't the same thing necessary when we transitioned to vinyl, reel-to-reel magnetic tape, cassette, 8-track, and compact disc (CD)? However, not all of them represented a quality increment in consumer perception. Some transitions were simply based on convenience (e.g., digital music distribution in compressed formats) and it will continue to happen as we pursue the convenience of wireless audio.

Appreciating that "perception" is "the way in which something is regarded, understood, or interpreted" (as defined by *Oxford Dictionaries*), it also directly relates to "one's willingness to actually think about, understand something or notice something easily using one of our senses" (*Merriam-Webster Dictionary*). Some people dedicate their lives to the continuous search for something inherently better, to innovation, and to accomplish real progress. The same motivation that lead brilliant minds to envision the MP3 compression algorithm—which by itself was a key enabler for the digital age—is also motivation for an audio engineer's continuous search for the best audio results—from recording to mastering and reproduction.

While attending two recent European technology trade shows, I heard countless debates regarding 4K images (4096 × 2160 pixels) and Ultra-High Definition (UHD) television (3840 × 2160 pixels) as being another industry effort to sell "technology for technology's sake," or selling consumers "a solution for a problem they don't realize they have." And yet, most industry specialists agree that innovation in image technology should combine higher resolution with higher frame rates, higher dynamic range, and so forth because the result would be "real benefits" for consumers. Even if today's consumers are watching (and supposedly satisfied with) highly compressed 720-pixel high-definition (HD) pictures that they think are "Full-HD" should we restrict innovation to the consumer's perception of real benefits? Should we debate technology and standards merits based on the end-consumer's ability to understand the problem/solution?

We already had the Super Audio CD (SACD) debacle, which was caused by the transition to file-based digital music distribution, and yet, that technology effort had longstanding merits, which were highly beneficial to consumers. The fact is, consumers don't need to listen to over-compressed music. Reaching an agreement on the description of "Master Quality Recordings" as sources to produce high-resolution files for distribution will help everyone understand that it's not about perception, it is about what we all—as consumers—deserve.

João Martins
Editor-in-Chief

The Team

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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.

Wireless Audio (Part 2)

Wireless Personal Audio Networks

In the first article in this series, we briefly introduced wireless audio, discussing how the technical evolution in wireless applications followed critical applications areas. In the second part, we discuss the underlying technologies currently implemented over Bluetooth and Wi-Fi.

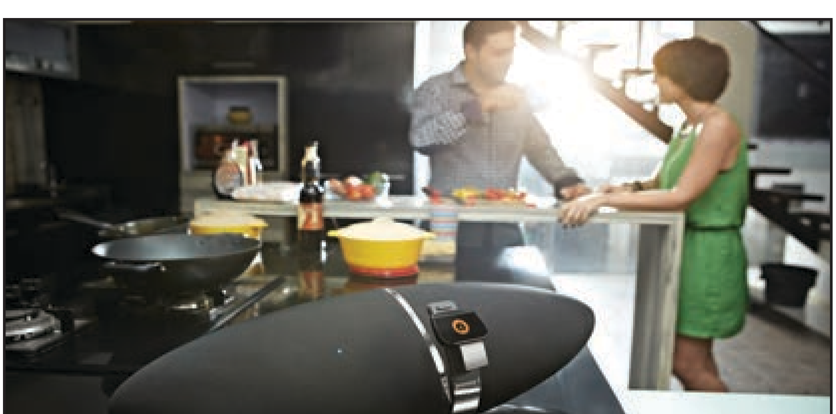
By
João Martins
(Editor-in-Chief)

What technologies and standards should we start with when we discuss wireless audio? As we've already mentioned, wireless audio transmission is in its infancy and Bluetooth is just the beginning.

Fundamental radio frequency (RF) technologies have been around for a few decades, but only now are they converging to actually enable superior applications on audio distribution. They are not only replacing cables but also offering superior multi-room coverage and even (convenient) multichannel by replacing simple RF transmission with digital data protocols and streaming via wireless data networks.

The entire wireless streaming and Bluetooth speaker craze effectively started a mere five years ago. The first Bluetooth radio module was introduced in 2000 and the first commercial wireless Bluetooth speakers were demonstrated in 2003. The first Bluetooth headphones (for music applications) followed in 2004, according to the Bluetooth Special Interest Group. The Bluetooth V. 4.0 specification arrived in 2011 but the global adoption of wireless speakers and particularly wireless headphones by consumers was mostly based on the Bluetooth 2.1+EDR (enhanced data rate) specifications, introduced in 2007.

Effectively, Bluetooth 4.1 chip solutions are only now starting to be industrially implemented in the new generation of wireless audio devices, both for wireless speakers and headphones. Among other things the Bluetooth V. 4.*x* SoC (System-on-Chip) platforms are enabling headsets, speakers, and soundbars to be controlled by a dedicated Bluetooth Smart remote control or mobile app, while concurrently streaming audio from a different device and network.



Wireless audio technology has quickly evolved over the last five years from music docks to wireless speakers being able to receive streaming music directly from the cloud. (Photo courtesy of Auris, www.theauris.com)

The Market

Bluetooth audio streaming (together with headphones) was responsible for sustaining continuous growth in the audio industry for the last four years. According to several market research studies, the global audio market is growing at almost 10% per year and is estimated to reach a value of $20 billion by 2018. The growth is fueled, in part, by accelerating technological developments in areas that include Bluetooth, Wi-Fi, Internet-compatible streaming and networking devices, and the growing popularity of digital music services such as iTunes, iTunes Radio (now reinforced with Beats Music), Spotify, Pandora, Deezer, and many others. Much in the same way digital downloads are giving way to streaming services, things are quickly changing, and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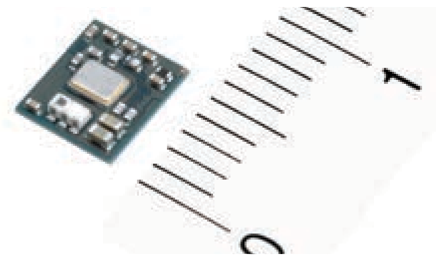
Bluetooth technology powered the global expansion of wireless audio to the consumer. The recent Bluetooth 4.0 low energy (LE) specification and Bluetooth Smart technology focus mainly at power-friendly applications with wearables and Internet of Things (IoT), enabling for better control and faster file transfers in packets, rather than live streaming.



In 2014, TDK introduced the ultra-compact SESUB-PAN-T2541 Bluetooth low-energy module designed for Bluetooth Smart applications. Measuring just 4.6 mm × 5.6 mm with a slim insertion height of 1 mm, it is the world's smallest Bluetooth Smart module.

accelerating every year. Wireless audio is undoubtedly a part of that evolution.

Bluetooth wireless audio products currently account for 82% of wireless speakers and are expected to be a popular choice for the next few years. With the continuous expansion of Bluetooth Smart (V. 4.*x*) and the availability of development kits for the sophisticated implementations that Apple experimented with in its "i" devices, there is no question this technology will continue to attract consumers.

Speaker docks for audio playback were followed by soundbars and wireless connectivity. Consumers want their smartphones and tablets to be connected and have faster web connectivity, enjoying broadband streaming entertainment and cutting-edge music storage equipment. It is an evolution that some market analysts predict will accelerate to a 14% compound annual growth rate (CAGR) until 2018 (ABI Research).

Wireless speakers are currently growing at an annual rate of 181% while the "highly fragmented and increasingly competitive" headphone market is anticipated to continue to grow 5% CAGR until 2017 (Futuresource, 2014).

According to the latest studies from the Consumer Electronics Association (CEA), in the US, nearly one in five (16%) consumers own a portable wireless speaker and one in 10 consumers own a wireless multi-room audio (MRA) product. Nearly half of non-owners (44%) indicate interest in owning one. Additionally, 91% of wireless MRA product owners and 85% of portable wireless speaker owners would recommend their products to someone else.

Recent devices like this Nakamichi (WOW Technologies) Bluetooth Headphones, launched in 2013, still use Bluetooth 2.1 + EDR (enhanced data rate) with Advanced Audio Distribution Profile (A2DP), which is the most common version on the market.

Also according to CEA studies, primary listening locations for wireless audio solutions are the living/family room (47%) and bedroom (28%), but a quarter (25%) of consumers say their primary listening area is a different room in the home. Portable speakers are expected to grow from more than 47 million units in 2013 to nearly 61 million by 2018, according to ABI Research, reaching over half of the market by the end of 2018. It is a growth that can be expanded, depending on the popularity of new and existing digital music services.

Bluetooth Boom

Feature phones, smartphones, and tablets shipped more than 1.9 billion units combined in 2013, according to ABI Research. And, there are many opportunities for technology advancements in audio quality—from high-definition voice delivery to new smart audio accessories and products related to the new generation wearable devices. New generation mobile devices will rely on dedicated audio processors to address a broader range of usage environments (e.g., adapting in near real-time to current conditions).

Bluetooth was particularly successful in this area, compared even with higher quality Wi-Fi-based wireless implementations (e.g., AirPlay and Sonos).

Bluetooth's key weakness is that the technology was not originally designed for wireless audio applications and that its narrow bandwidth—Bluetooth Advanced Audio Distribution Profile (A2DP)—enables a maximum 768-kilobits per second (kbps) available throughput. The narrow bandwidth forces the use of compression, even if the original audio signal might already be highly compressed using MP3 or AAC algorithms. Bluetooth standard low complexity subband coding (SBC) codec supports up to 345 kbps for stereo audio and introduces delay to compress/decompress the files, resulting in further audio deterioration. It is a difficult problem for speaker manufacturers to control because it is dependent on other chip manufacturers and device certification, according to the original Bluetooth specifications.

Even taking into account Bluetooth support for optional codecs listed in the Bluetooth A2DP specification—including MPEG-1, MPEG-2, MPEG-4 AAC, aptX or even other non-A2DP codecs—higher quality audio can only be achieved within the specification constraints.

However, the Bluetooth specification can improve in this application area. SBC's transmission quality can also be significantly enhanced by selecting optimal encoding parameters compared to the default parameter sets given in the original standard. But for high-quality stereo audio implementations to be possible (e.g., using aptX streaming over the Bluetooth A2DP), the source device and a Bluetooth stereo speaker, a headset, or headphones need to incorporate the same technology to derive the sonic benefits of aptX audio coding.

The Bluetooth Special Interested Group recognized these limitations early on. Since the introduction of Bluetooth V. 3.0+High Speed (HS) in 2009, faster data transfers of up to 24 Mbps were envisaged using a Bluetooth link in parallel with 802.11 (Wi-Fi). Data rates of up to 54/108 Mbps are possible combining Bluetooth V. 4.0+HS with 802.11n and later specifications. Such solutions require specific SoC implementations, which only recently became available.

Texas Instruments's platforms (e.g., the WiLink 8 family) combine Wi-Fi and Bluetooth solutions with multiple certified modules, ensuring that wireless speakers and other connected audio devices will be able to receive an audio stream from practically any streaming device. The WiLink 8 solution's robust Wi-Fi/Bluetooth coexistence capabilities enable products to use Bluetooth's low energy as the interface for remote control while Wi-Fi functions as the channel for streaming audio.

Antenna diversity and 2.4-GHz and 5-GHz dual-band connectivity extend the WiLink 8 module's wireless communications range and enable it to maintain connectivity even in the most congested RF environments. Advanced Wi-Fi capabilities (e.g., Wi-Fi Direct) enhance the user experience. For example, a smartphone might be linked to a connected audio device such as a wireless speaker via Wi-Fi Direct while the speaker with the WiLink 8 solution maintains connectivity to the Internet. The smartphone streams music to the speaker, and the user is still able to browse the Internet or pull music streaming from the cloud, without connecting to the home access point or cellular network. Wi-Fi Direct also enables devices to directly connect without depending on existing WLAN infrastructure (routers) or Internet connections for simple wireless audio applications.

This convergence is why the market is exploring advanced Bluetooth implementations and turning its attention to developments on the Institute of Electrical and Electronics Engineers (IEEE) 802.11 standard for wireless local area networks (WLANs).

Going Wi-Fi

In sharp contrast with Bluetooth's limited specifications, Wi-Fi technology evolved faster, promising to combine quality and convenience in audio applications. Since consumers have discovered the convenience and versatility that wireless connected audio devices and systems can offer, it has become clear that a higher degree of interoperability among connectivity technologies would be required for enhanced user experience and improved sound quality.

There is no question that wireless networking technologies are going to extend many application categories, not only in home audio speakers but also in car or even semi-professional applications. It has already started to happen with musician wireless systems and Wi-Fi/Bluetooth or Wi-Fi and direct digital RF links being adopted in portable PA solutions and music controllers.

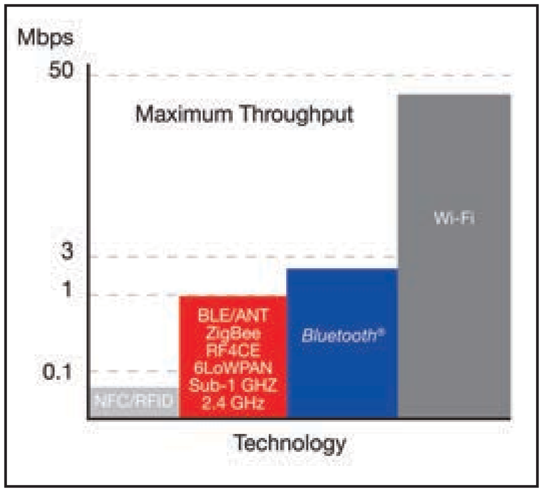
With the spectrum for Wi-Fi being reinforced around the world (1, 5, and 60 GHz in addition to the initial 2.4 GHz), the ubiquity of WLAN solutions for many applications, including streaming audio will continue to drive double-digit sales growth rates, as projected by the Wi-Fi Alliance. The importance of Wi-Fi in the audio industry is set to continue fostering growth in many new product areas and applications over the next years. The emergence of multi-gigabit point-to-point wireless will soon enable high-quality multichannel audio streaming in tandem with uncompressed HD video and more.

With industry anticipation of WiGig 60-GHz technology, the Wi-Fi Alliance predicts an

Proving how far wireless audio has evolved, even considering technology limitations, the Harman Kardon BT has been considered one of the best-sounding Bluetooth headphones available.

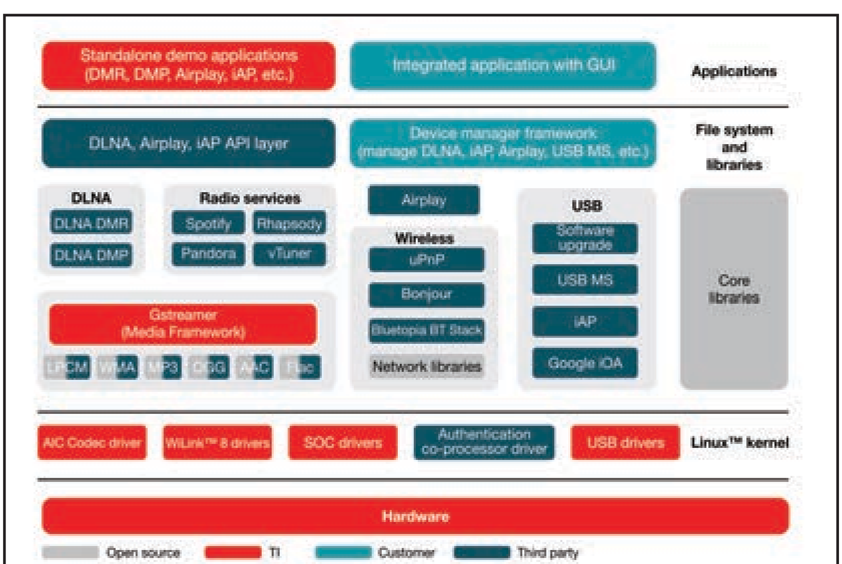
Marshall Headphones, the Swedish company that licensed the famous British brand for consumer products, introduced the Stanmore Bluetooth speaker/guitar-amplifier concept in 2013. Not the classic Bluetooth speaker, it uses Bluetooth V. 4.0+EDR technology and the aptX codec.

The graph shows sustained data rates on wireless connectivity technologies. (Information courtesy of Texas Instruments, Inc.)



explosion in wireless connectivity applications, including integration with existing Wi-Fi solutions. Wireless speakers based on Wi-Fi solutions offer the convenience of portability and MRA as already demonstrated by brands such as Sonos—the pioneer in Wireless MRA solutions—and dozens of other brands launching new multi-room products in 2014.

The industry is clearly embracing the benefits of WLAN and network data protocols, now indispensable in digital wireless audio applications. Bluetooth remains the sensible option mainly for interface and control as well as noncritical audio applications in short-range portable/battery powered applications including headphones, while higher quality, MRA and direct Internet streaming will clearly evolve with Wi-Fi.



Complete solutions for next-generation wireless-connected audio require many underlying software modules. These include all of the software stacks for complete Wi-Fi and Bluetooth connectivity, operating systems support, a considerable amount of firmware, as well as low-level drivers. (Information courtesy of Texas Instruments, Inc.)

According to the Wi-Fi Alliance, Wi-Fi connectivity increases the purchase likelihood for smart home devices and Wi-Fi is a key feature. Connected applications (e.g., home security, smart energy, and in-vehicle infotainment) will complement existing home networks, with smart devices interactively operating. It remains to be seen what impact controlling personal health devices, appliances, thermostats, or lighting systems on the same network will have toward audio streaming, but Wi-Fi is clearly evolving and has the potential to converge all those applications.

Wi-Fi can elegantly handle intermittent transmission of small amounts of data to high-definition multimedia, enabling a range of connectivity for different uses. Nevertheless, that's precisely where the case for combining different technologies (e.g., Bluetooth and Wi-Fi) makes sense.

In its 15 years of existence (celebrated in 2014), Wi-Fi evolved from the original 11-Mbps data rate offered in the first Wi-Fi certified products (1999), to today's 1 Gbps, already tested in advanced devices. Device-to-device connectivity was introduced with Wi-Fi Direct in 2011. It basically provided an alternative to Bluetooth in most applications, while still offering the potential to support typical Wi-Fi speeds of up to 250 Mbps.

The Wi-Fi Alliance roadmap features several key elements that will support the proliferation of several new segments and applications. New frequency bands and WiGig in 60-GHz interoperable products supporting multiple-gigabit data rates for room-range connectivity are scheduled to appear in 2016. The recently ratified 802.11ad standard enables multi-gigabit networking, data syncing, and video and audio streaming, while maintaining its wireless bus extension docking capabilities. The Wi-Fi Alliance has also begun to define certification programs based on 802.11ah and 802.11af for operation below 1 GHz to support longer-range, very-low-power connectivity (alternatives to Bluetooth), which will all impact wireless audio applications in a crucial way.

New wireless audio systems are already streaming high-quality audio with the most recent Wi-Fi dual-band technology, 802.11ac, operating in both the 2.4- and 5-GHz bands. Companies such as Broadcom, Intel, Marvell, Mediatek, Qualcomm, and Realtek are supplying complete development platforms, integrated solutions, and reference designs. For instance, Qualcomm's VIVE solutions use the new 802.11ac standard to power wireless MRA, 4K video streaming and online games. Qualcomm's VIVE also offers StreamBoost, a product of Qualcomm Atheros, to intelligently concurrently manage all multiple applications, while consuming

New generation solutions such as the SmartSpeaker (A1) are totally wireless portable systems based on Wi-Fi, supporting Apple's AirPlay and even Wi-Fi Direct mode.

six times less power than previous Wi-Fi technologies.

Wireless audio applications obviously benefit from the 5-GHz spectrum and wider channels. With more products crowding the wireless spectrum, the 5-GHz band enables faster transmission with less interference and superior channel agility to maintain maximum speeds between 802.11ac devices while using narrower channels for legacy 802.11n devices.

Qualcomm also announced that its subsidiary, Qualcomm Atheros, and Wilocity, a developer of 60-GHz multi-gigabit wireless chipsets, have launched the industry's first tri-band reference design that combines 802.11ac and 802.11ad wireless capabilities on a single module. Based on Qualcomm VIVE 802.11ac Wi-Fi and Wilocity's 802.11ad WiGig wireless technologies, this reference design delivers tri-band Wi-Fi, enabling consumers to connect to 60-GHz-enabled devices at multi-gigabit speeds, while maintaining home coverage with 2.4-GHz/5-GHz Wi-Fi.

Resources

ABI Research, www.abiresearch.com.

Bluetooth Special Interest Group (SIG), Inc., "Bluetooth Advanced Audio Distribution (A2DP) Profile Specification," 2007, www.Bluetooth.org.

——, "History of the Bluetooth Special Interest Group," www.bluetooth.com/Pages/History-of-Bluetooth.aspx.

Future-Source Consulting, www.futuresource-consulting.com.

P. Khara and A. Nishat, "Complete Solutions for Next-Generation Wireless-Connected Audio," Texas Instruments White-Paper, www.ti.com.

Qualcomm Technologies, Inc., www.qualcomm.com.

SmartSpeaker, www.thesmartspeaker.com.

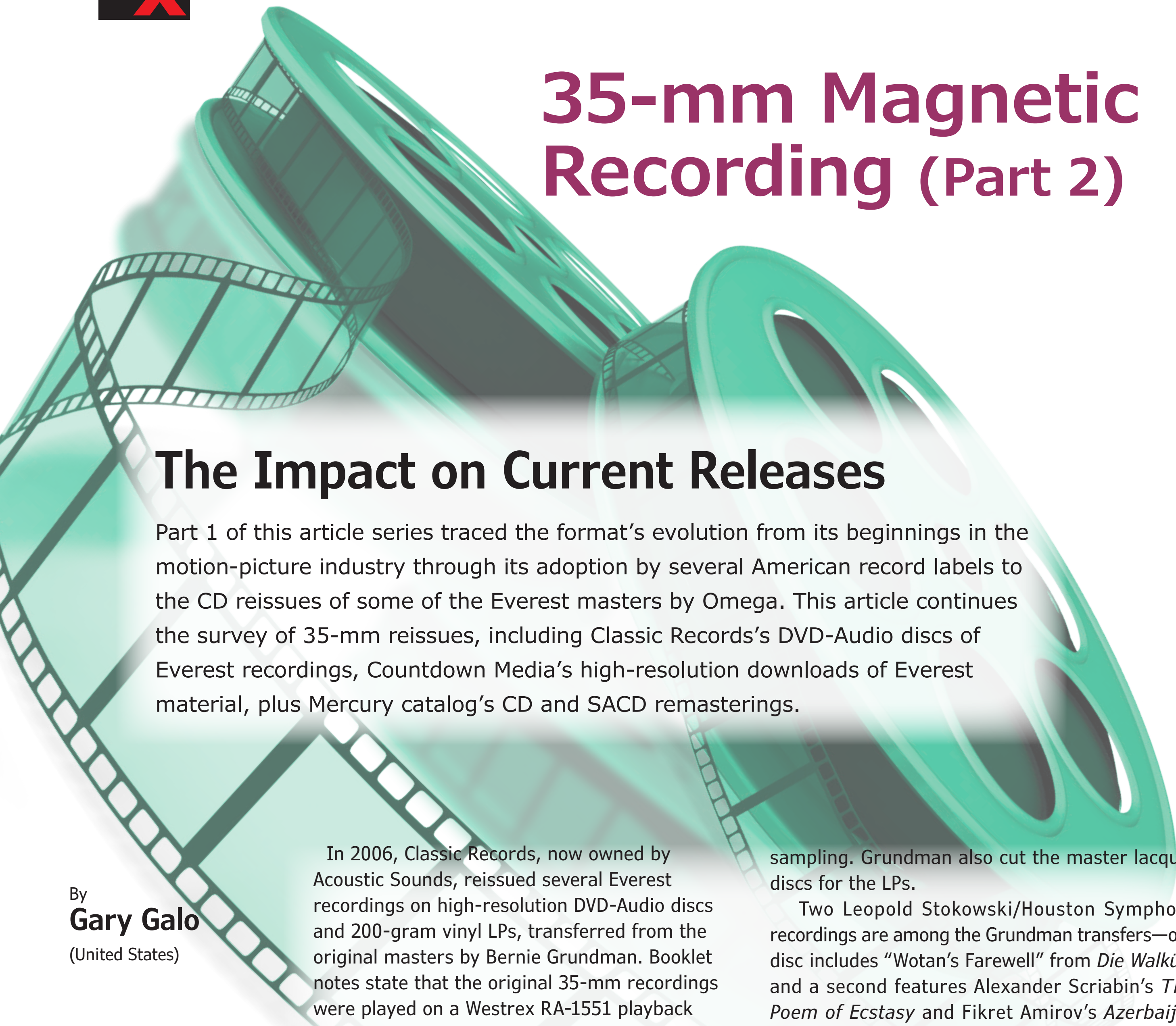
Texas Instruments, Inc., "Texas Instruments Wireless Connectivity Guide," www.ti.com/wirelessconnectivity.

Wakefield Research, "Wi-Fi Alliance Consumer Electronics Show (CES) 2014 Survey," www.wakefieldresearch.com.

Wi-Fi Alliance, "Discover Wi-Fi: Connect Your Life," White Paper, www.wi-fi.org/connectyourlife.

——, "Discover Wi-Fi: 15 Years of Wi-Fi," White Paper, www.15yearsofwi-fi.com.

Wireless Speaker and Audio Association, "Wireless Speaker and Audio (WiSA) Association White Paper," www.wisaassociation.org.



35-mm Magnetic Recording (Part 2)

The Impact on Current Releases

Part 1 of this article series traced the format's evolution from its beginnings in the motion-picture industry through its adoption by several American record labels to the CD reissues of some of the Everest masters by Omega. This article continues the survey of 35-mm reissues, including Classic Records's DVD-Audio discs of Everest recordings, Countdown Media's high-resolution downloads of Everest material, plus Mercury catalog's CD and SACD remasterings.

By
Gary Galo
(United States)

In 2006, Classic Records, now owned by Acoustic Sounds, reissued several Everest recordings on high-resolution DVD-Audio discs and 200-gram vinyl LPs, transferred from the original masters by Bernie Grundman. Booklet notes state that the original 35-mm recordings were played on a Westrex RA-1551 playback machine "meticulously" restored by Len Horowitz.

Horowitz built special playback electronics—claimed to be vastly superior to any others used in these machines—and fitted the modified RA-1551 with custom three-track playback heads. The transfers were done with Horowitz operating the 35-mm machine and Grundman creating the 192-kHz/24-bit digital master. He used a proprietary ADC designed by Kevin Halverson of Muse Electronics. (Scott Sedillo and Beno May are also given credit in the transfer process.)

The DVDs are double-sided with a 192-kHz DVD-Audio program on one side and a 96-kHz program on the other. The latter is compatible with DVD-Video players. Each two-disc package also includes a CD that was made from a separately captured 44.1-kHz/16-bit master to avoid non-integer down sampling. Grundman also cut the master lacquer discs for the LPs.

Two Leopold Stokowski/Houston Symphony recordings are among the Grundman transfers—one disc includes "Wotan's Farewell" from *Die Walküre* and a second features Alexander Scriabin's *The Poem of Ecstasy* and Fikret Amirov's *Azerbaijan Mugam* (see **Photo 1**). The 192-kHz DVD-Audio programs sound quite spectacular with exceptional detail and resolution and excellent soundstage information. Although the CDs do not sonically match the DVD-Audio discs, they are very well done. My only quibble is that Classic Records' transfers are very "up-front," perhaps even a bit aggressive in their sonic presentation. Comparing "Wotan's Farewell" to its counterpart in the Omega series is interesting. Classic Records' transfer offers a much closer sonic perspective, similar to what a conductor hears on the podium. The Omega transfer puts the listener a few rows back in the audience. They almost sound like recordings made with two different microphone placements, which they are not. The Omega transfer cannot match the large soundstage and precision of localization heard on either the DVD or the CD in the Classic Records package, but they remain musically satisfying, and some listeners may prefer their more "laid-back" sonic character. I haven't heard the Classic Records LPs.

Downloads from Countdown

Last year, Countdown Media, which now owns the Everest catalog, began remastering a sizeable portion of the Everest 35-mm recordings from the original films. These transfers are available as on-demand CD-Rs from Amazon and as high-resolution pulse-code modulation (PCM) downloads from HDTracks and iTunes. (The downloads include booklet graphics and notes.)

Countdown Media has used Grundman's masters for the material he transferred for Classic Records, but with "additional restoration" by Lutz Rippe. The remaining material was newly-transferred by Rippe using an Albrecht MB51 35-mm magnetic film player. The MB51 is a capstan-based machine, which avoids the problems encountered when playing shrunken films on a sprocket-driven playback system. The MB51 was introduced in 1980 and the manufacturer developed special playback heads in 2006 to help the machine cope with the vinegar syndrome problem. (For more information on Albrecht, visit www.mwa-nova.com/Company/Company.html).

I heard several of the Countdown remasterings as 96-kHz/24-bit downloads (some are also available at 192 kHz). Sonically, I'm not entirely satisfied with them. Grundman's Stokowski transfers of the Richard Wagner and Scriabin works appear to have been processed with digital hiss reduction, but to my ears it's been overdone. There are subtle artifacts from the digital noise reduction (DNR), which is always a danger if the restoration engineer tries to remove too much hiss. These restorations also lack the pristine clarity and detail of the Classic Records DVDs—a veil has been placed between the recordings and the listener. Some of the transfers also suffer from occasional pitch instability.

The choral sound on Omega's CD of the Lili Boulanger compositions is smooth and warm. Although Countdown Media's remastering has a larger soundstage, which I have come to expect from high-resolution transfers, there's a thin layer of grit in the choral sound that is absent on the Omega CD (see **Photo 2**).

Countdown Media also transferred Stokowski's *Parsifal* excerpts. *Parsifal* is a recording sadly overlooked by Classic Records, but with sonic characteristics similar to their other transfers. In 2008, the British Criterion Music Co. capitalized on the Everest legend with a series of budget-priced CD reissues. The sound quality does no justice to the originals. Based on the sound, I'm almost certain

Photo 1: Two of Leopold Stokowski's albums with the Houston Symphony were remastered for Classic Records by Bernie Grundman and issued on high-resolution DVD-Audio discs and 200-gram LPs. This one is devoted to music by Alexander Scriabin and Fikret Amirov.

that these CDs have the same 0.25" tape copies in their ancestry as those used for the Everest reissues that appeared many years ago on the Priceless and Bescol labels. On Amazon, the label is identified as "Everest UK"so buyer beware.

Mercury Reissues

The sound quality of the Mercury recordings engineered by C.R. Fine is consistently excellent. This applies to both his 35-mm work and the myriad stereo recordings he made on 0.5", three-track tapes. (His mono recordings are also among the best ever made.) As I previously mentioned, Fine used three-spaced omnidirectional microphones for the Mercury classical stereo recordings. During the period he was making 35-mm recordings, he generally used Schoeps M201 microphones (sold in

Photo 2: Countdown Media has remastered a large portion of the Everest 35-mm catalog, including this 1959 recording of five works by Lili Boulanger, conducted by Igor Markevitch. An Albrecht MB-51 capstan-based player was used for many of these transfers to overcome the difficulties of playing films suffering from shrinkage on sprocket-driven equipment.

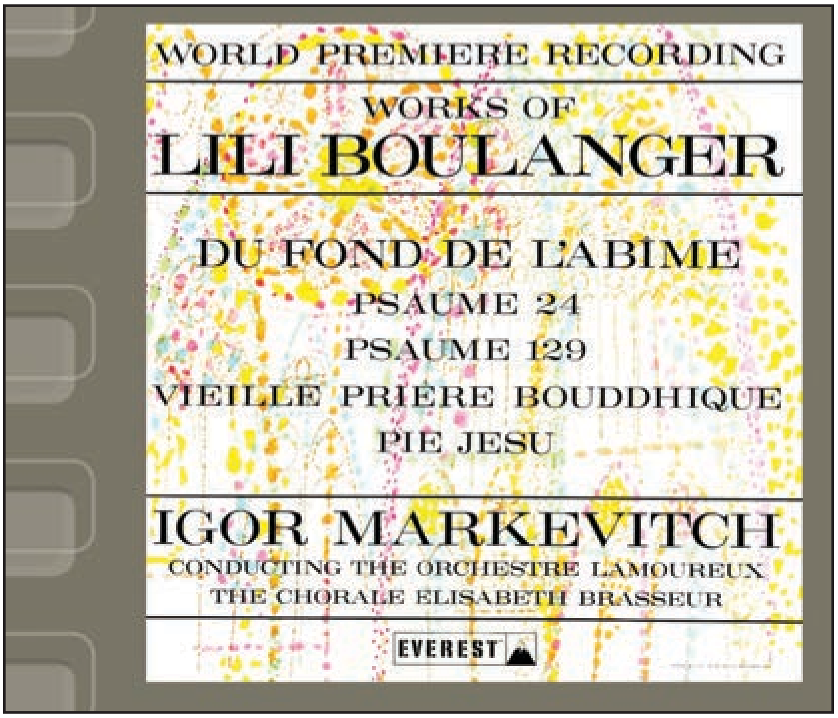


Photo 3: Several of Mercury's outstanding 35-mm recording were remastered for CD in the early 1990s, and again for SACD in 2005. This CD of Bartók's *Concerto for Orchestra* conducted by Antal Dorati combines a definitive performance with reference-quality sound.

About the Author

Gary Galo retired in June 2014 after 38 years as an Audio Engineer at the Crane School of Music, SUNY, in Potsdam, NY. He now works as a volunteer, transferring vintage analog recordings in the Crane archive to digital format. He is the author of more than 280 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*.

the US as the Telefunken M201).

Fortunately, a sizable portion of the Mercury catalog was reissued on CD some 20 years ago. They contain sonically impressive transfers produced by Wilma Cozart Fine. A few items were also transferred to Super Audio CD (SACD) in the mid-2000s at the Emil Berliner Studios in Hanover, Germany, where the masters for all of the recordings under the Universal Music umbrella—Deutsche-Grammophon, Philips, Mercury, Westminster, British Decca, American Decca, et al.—are now held.

Wisely, Mercury did not try to fill its catalog by duplicating the same standard repertoire covered by other record companies. Instead, Mercury utilized the artists at its disposal in the repertoire in which they specialized. They were most fortunate to have had the services of the Hungarian-born conductor Antal Dorati for several outstanding 35-mm recordings.

Dorati was highly regarded in 20th-century repertoire and in the orchestral music of his countryman Bela Bartók, he had few rivals. His performance of Bartók's *Concerto for Orchestra* with the London Symphony Orchestra, recorded in Wembley Town Hall in 1962, remains one of the finest ever recorded (see **Photo 3**). Dorati also excelled at the music of the Second Viennese School, where he found an ideal balance between the intellectual and emotional demands of this repertoire. For anyone interested in this music, Dorati's Mercury CD of Arnold Schoenberg's *Five Pieces for Orchestra*, Op. 16, Anton Webern's *Five Pieces for Orchestra*, Op. 10, plus Alban Berg's *Three Pieces for Orchestra*, Op. 6 and *Lulu Suite*, is essential. Despite being made more than 40 years ago, both recordings offer reference-quality sound that has held up well.

Mercury SACDs

In 2005, the Universal Music Group reissued a handful of Mercury recordings on SACD, including a few made with 35-mm equipment. For these transfers, the 35-mm films were played on a Sondor Oma-s Chace Magnetic Film Recorder/Reproducer which, according to the SACD booklet notes, was specially constructed to overcome any problems of acetate-based magnetic films affected by vinegar syndrome. This machine has a closed-loop, sprocket-driven transport, but the sprockets have been reduced in size to accommodate films with up to 4% shrinkage. A special three-section roller keeps even badly-wrinkled films in contact with the three-channel magnetic playback head. The manufacturer notes that as long as the magnetic coating has not separated from the base, "perfect copies of the original will be possible."

The Society of Motion Picture and Television Engineers (SMPTE) equalization was used and checked against a reference tape provided by Bob Eberenz, Fine's Associate Engineer at Mercury (see **Photo 4**). The transfer was made at 192-kHz/24-bit PCM and converted to Direct-Stream Digital (DSD) for SACD mastering.

Wilma Cozart Fine's transfers for the CD series were referenced to determine the correct three-channel-to-stereo mix. Her original transfers were also used for the CD layers of the SACDs so listeners can compare for themselves. (The SACDs also have a multichannel program that enables the listener to hear a three-channel playback.) Particularly outstanding in this series are Antonín Dvořák's *Cello Concerto* in B-minor with soloist Janos Starker, and Pyotr Ilyich Tchaikovsky's complete *Nutcracker* ballet,

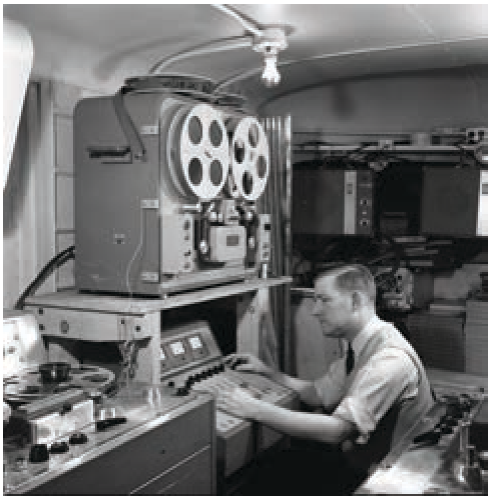


Photo 4: Bob Eberenz adjusts the faders at a Detroit Symphony session in 1962 with the truck outfitted for 35-mm magnetic film recording. (Photo courtesy of Tom Fine/www.preservationsound.com)

Resources

G. Galo, "35-mm Magnetic Recording (Part 1): The High-End Format of the 1960s," *audioXpress*, October 2014.

——, "High-Definition Digital," *audioXpress*, February 2014.

——, "The Legendary Rachmaninoff Symphonic Dances," *audioXpress*, December 2012.

L. Henrichsen and D. Patmore, "The Ascent and Descent of Everest," *Classic Record Collector*, Spring 2007.

——, "Assault on the Summit," *Classic Record Collector*, Autumn 2007.

MWA Albrecht GmbH, www.mwa-nova.com.

Sondor, www.sondor.ch.

Source

Oma-s Chace Magnetic Film Recorder/Reproducer
Sondor | www.sondor.ch/oma_e.html

both with Dorati conducting the London Symphony Orchestra.

J. Gordon Holt, founder of *Stereophile* magazine, once wrote: "The better the sound, the worse the performance." Happily, that does not apply to Mercury. It is a label that consistently offered excellent performances captured in reference-quality sound. Some of the reissues mentioned in this article are out of print. However, a few dealers still stock them, and used copies can be found on the Internet.

Author's Note: I am grateful to Tom Fine for allowing me to draw heavily on his presentation for this article, for supplying many of the photographs, and for his helpful suggestions during the preparation of this article. I would also like to thank Lonn Henrichsen for answering several pertinent questions. Finally, for more detailed information on the history of Everest Records, I recommend the two-part series by Henrichsen and David Patmore.

Recommended Recordings

Everest/Countdown

L. Boulanger, *Du fond de l'abîme* (*Psaume 130*), *Psaume 24*, *Psaume 129*, *Vielle Prière Bouddhique*, *Pie Jesu*, Chorale Elisabeth Brasseur and Orchestre Lamoureux conducted by Igor Markevitch. SDBR 3059. (High-resolution download from HDTracks and iTunes).

A. Scriabin, *The Poem of Ecstasy*, Op. 54 and F. Amirov, *Azerbaijan Mugam*, Houston Symphony Orchestra conducted by Leopold Stokowski. SDBR 3032. (CD-R from Amazon; high-resolution download from HDTracks and iTunes).

R. Wagner, *Die Walküre,* "Wotan's Farewell" and "Magic Fire Music," Houston Symphony Orchestra conducted by Leopold Stokowski. SDBR 3070. (CD-R from Amazon; high-resolution download from HDTracks and iTunes).

R. Wagner, *Parsifal,* "Good Friday Spell" and Act III Symphonic Synthesis, Houston Symphony Orchestra conducted by Leopold Stokowski. SDBR 3031. (CD-R from Amazon; high-resolution download from HDTracks and iTunes).

Mercury Living Presence

B. Bartok, *Concerto for Orchestra*, London Symphony Orchestra conducted by Antal Dorati. CD 432 017-2 (Amazon, ArkivMusic or eBay).

A. Dvořák, Concerto in B-minor for Cello and Orchestra, Op. 104; M. Bruch, *Kol Nidrei*, Op. 47, Janos Starker, cello; London Symphony Orchestra conducted by Antal Dorati. SACD 475 6608 or CD 432 001-2 (Amazon, ArkivMusic, or Elusive Disc).

A. Schoenberg, *Five Pieces for Orchestra*, Op. 16; A. Webern, *Five Pieces for Orchestra*, Op. 10; A. Berg, *Three Pieces for Orchestra*, Op. 6 and *Lulu Suite* (with Helga Pilarczyk, soprano). London Symphony Orchestra conducted by Antal Dorati. CD 432 006-2 (ArkivMusic).

P. Tchaikovsky, *The Nutcracker* (Complete ballet). London Symphony Orchestra conducted by Antal Dorati. SACD 475 6623 or CD 432 750-2 (Amazon, ArkivMusic, or Elusive Disc).

Nordic Audio Tales (Part 3)

The Resilience of Scan-Speak

The third article in this series on Nordic audio pioneers explores Scan-Speak's audio history. Get a glimpse of the company's early innovations, ownership changes, and its unique ability to offer new technologies in an ever-changing market.

By
Mike Klasco
(United States)

Founded in 1970, Scan-Speak is still based in its original town of Videbæk, Denmark (see **Photo 1**). Located in the heart of the Jutland Peninsula, Scan-Speak's acoustic engineers adhere to the legendary principle "Never Compromise" in its pursuit of sound perfection. For more than 40 years, Scan-Speak has focused on premium-grade high-definition transducers—with several innovations and patents ranging from ultra-low distortion magnet systems to all aspects of optimized moving components (see **Photo 2**).

Photo 1: Scan-Speak's headquarters are located in Vadebaek, Denmark. (Photo courtesy of Scan-Speak)

Ejvind Skaaning, a noted Nordic speaker engineer, founded Scan-Speak in February 1970. His primary purpose for founding Scan-Speak was to manufacture drivers for the Scandyna A-25 (i.e., the Dynaco A-25). Scan-Speak's first product was the D2008 tweeter, developed for a new line of drivers offered from 1971–1974. By 1972, Scan-Speak employed about 40 employees (about the same staffing it has today). And, the D2008 tweeter was endlessly copied for several decades.

During 1973, Scan-Speak's Ragnar Lian invented and patented the Symmetric Drive (SD) motor, which consists of a copper cap placed on the pole piece to minimize eddy currents. The design's new and patentable aspect was that the copper is located and dimensioned so that the inductance variation of the driver coil position becomes symmetric around the rest position.

Late in 1973, Skaaning made an investment deal regarding Scan-Speak with David Hafler. Hafler had been Dynaco's director and made his fortune when he sold the company to Tyco in 1969. In 1974, Skaaning pulled out of Scan-Speak financially but he continued to work as the company's laboratory manager. (He later started what became Morel in Israel). Soon after, Scan-Speak ownership shifted to Fonofilm Industry—the parent company for Ortofon, which was owned by Hafler and Zoran Chanin.

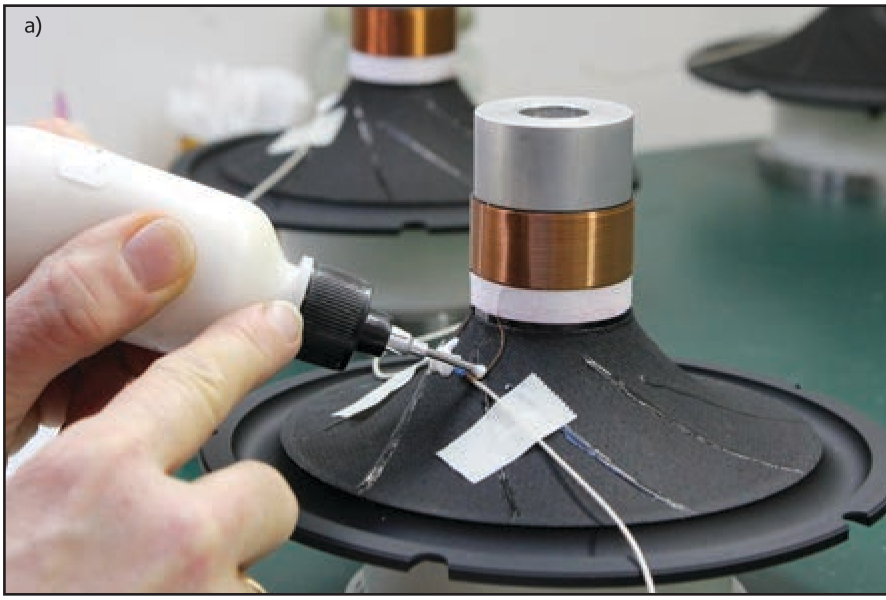


Photo 2: The voice coil lead out wire bonding process is shown up close (a) and from a distance (b). Scan-Speak's cone splicing and damping processes are effective. (Photos courtesy of Scan-Speak)

I remember ADCOM, a local audio sales agency in the 1970s that represented KLH, Ortofon, and a few other brands. Ortofon was originally marketed in the US by Electro-Sonic Labs. The company was (and is) famous for its moving coil phono cartridges. Later, ADCOM became an audio brand that offered preamplifiers and power amplifiers in the 1980s and 1990s. But back in the mid 1970s, Ortofon started to sell finished speakers under the Ortofon brand in Europe.

HARMAN purchased Ortofon and the ownership change did not work out well for Scan-Speak. The company was liquidated, which resulted in its tooling and production gear being sold to various companies. It is interesting to note that Niels Jespersen was the export manager for Ortofon at the time. Today, he is on the board of directors at Scan-Speak.

Ortofon was then split into two companies, (Ortofon Manufacturing and ADCOM). Ortofon Manufacturing became affiliated with Harman-Kardon, which was founded by Sidney Harman and Bernard Kardon. In the 1970s, Sidney Harman accepted an appointment in the Jimmy Carter administration as undersecretary of the US Department of Commerce. When Harman took office in 1976, he sold his company to Beatrice Foods to avoid a conflict of interest. Beatrice Foods then sold many portions of the company, including the original Harman-Kardon division. By 1980, only 60% of the original company remained. After Harman left government in 1978,

Photo 3: In 1988, Scan-Speak launched a loudspeaker line made with Kevlar cones. The 18W 8424G00 is shown in top (a) and side (b) views. (Photos courtesy of Scan-Speak)

Photo 4: The Revelator 15W 8530K00 is legendary. (Photo courtesy of Scan-Speak)

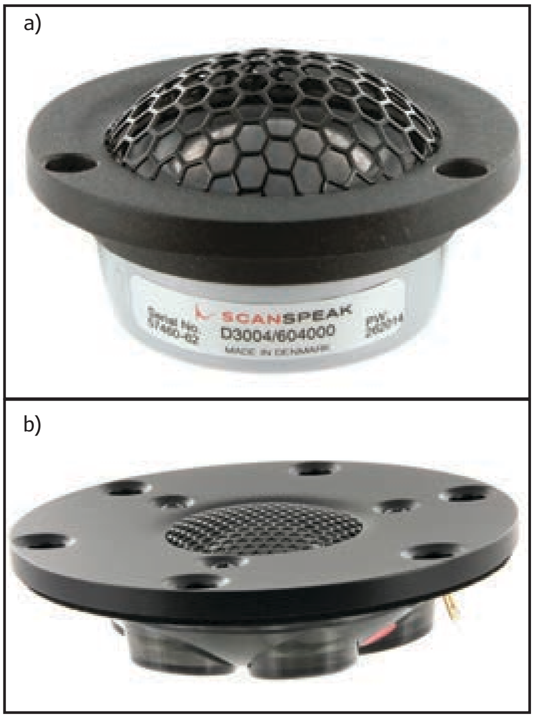


Photo 5: Scan-Speak's D3004/664000 beryllium dome wideband tweeter (a) contains a unique magnetic structure (b). (Photo courtesy of Scan-Speak)

Photo 6: In 2012, Scan-Speak launched its 13" subwoofer, the 32W/4878T00. (Photo courtesy of Scan-Speak)

he created HARMAN International Industries and reacquired several businesses that he had sold to Beatrice Foods.

In 1977, Dantax, a high-end Danish box speaker brand, purchased Scan-Speak or at least what was left of it, including some of the production equipment, patents, and rights to the Scan-Speak brand name. These remnants were moved to the Dantax location in Pandrup, Denmark. At this time, Dantax had 45 employees and the Scan-Speak purchase greatly expanded its organization.

Oskar Wrønding was a key person in re-establishing Scan-Speak and became the head of product design, development, and production. He was later promoted to technical manager. Wrønding eventually established a boutique high-end speaker company named Hiquphon, which produces tweeters that are inspired by the Scan-Speak D2008/D2010. As for Dantax, today it has evolved into an international company, offering a range of Danish designed and developed hi-fi and home cinema loudspeakers, tabletop radios, and hi-fi products under the Scansonic brand.

In 1978, Scan-Speak engineering developments continued with its Flow Resistance device, an acoustic "valve" deadening internal organ pipe resonances in bass reflex ports (i.e., aperiodic damping). In 1988, Scan-Speak launched a loudspeaker line with Kevlar cones which included its 13M, 18W, and 21W models (see **Photo 3**).

In 1989, Vifa purchased Scan-Speak. And, its production setup, documentation, and administration was gradually transformed to "Vifa style." In 1991, Scan-Speak's production was moved to Videbæk, Denmark, just a few blocks from Vifa. Technical developments continued with the second-generation Symmetric Drive, SD-1. Patented by inventor Lars Goller, it consisted of three rings of copper placed at specific positions in the magnet system for eddy current nulling. In 1995, Scan-Speak launched its epic "Revelator" product line with the D2905/9900 28-mm diameter tweeter with waveguide faceplate (see **Photo 4**).

In 2005, Tymphany acquired DST, which included Vifa, Peerless, and the Scan-Speak group. In 2008, the Scan-Speak Illuminator line was developed with a patented magnetic system and products shipped the following year.

In 2008, the world economy dipped and various Tymphany groups underwent transitions. Scan-Speak separated from Tymphany Denmark, it remained in Videbæk. In 2009, there was a management buyout of Scan-Speak. Tymphany ownership changed in a management buyout and the Peerless and the Vifa brands moved to Asia.

During the spring of 2009, Scan-Speak launched its Discovery line. This line represented Scan-Speak's entry line, but it was offered at a more affordable price. The Discovery line still utilizes the well-known patented waveguide and NRCS cone. The Discovery line broadens the market and customer potential and is, as all Scan-Speak products, still made in Denmark.

In 2010, Scan-Speak launch the D3004/664000 beryllium tweeter (see **Photo 5**). In 2012, Scan-Speak showcased a completely new 13" subwoofer, the 32W/4878T00 (see **Photo 6**), and an extremely high-efficiency beryllium tweeter, the D2908/714000, as part of its Revelator line. In 2013. Scan-Speak continued the line with a 2.5" full-range driver that is ideal for compact high-definition soundbars. While a high-definition soundbar might seem to be an oxymoron, I had a chance to listen to a prototype when I visited Scan-Speak. I can tell you the clarity of movie dialog is quite apparent and non-fatiguing.

At the Munich High-End Show last May 2014, Scan-Speak demonstrated a prototype of a new small bass driver (see **Photo 7**). At the same time, Scan-Speak privately previewed two new compact beryllium tweeters that should be formally announced soon.

Photo 7: At the last Munich High End Show, Scan-Speak demonstrated its new 12WU small bass driver. (Photo courtesy of Scan-Speak)

In April 2014, Scan-Speak announced it was acquired by Eastern Asia Technology, a Taiwan public company that is an offshoot of an original group of companies that were formed in 1971. Generally known as Eastech, the company is an OEM/ODM speaker systems and earphones manufacturer that possesses a comprehensive acoustic, structural, and driver development engineering team.

With Eastech's backing, the future is bright for Scan-Speak. However, it is still too early to speculate as to what may be next for Scan-Speak and the products it will offer.

Resources

Lansing Heritage, www.audioheritage.org and http://test.audioheritage.org.

Genelec M Series

Acoustic Design and Sustainability

For the past 35 years, Genelec has been known for its ability to introduce new and innovative technology. Genelec's new energy-efficient M Series takes a novel approach to monitor design and manufacturing as part of a wider ongoing sustainability program. M Series research and development work is focused on advanced sustainability without sacrificing accuracy and reliability.

By
Christophe Anet
(France)

Genelec, a manufacturer of active loudspeaker systems, is based in Iisalmi, Finland. The company designs and produces products for professional studio recording, mixing and mastering applications as well as broadcast and post production facilities. Genelec's M Series Natural Composite Enclosures (NCEs) employ a new environmentally friendly material manufactured in Finland from wood fibers and other recyclable materials. The M030 and the M040 bi-amplified active monitors also share a new technology platform featuring green power-saving electronics, Intelligent Signal Sensing (ISS), power management, automatic mains voltage selection, linear and clean Class-D power amplifiers, as well as easy-to-use room compensation adjustments (see **Photo 1**).

The monitors also have Genelec's enclosure design heritage, featuring rounded edges and curved front and sides to minimize cabinet edge diffraction for precise imaging. The new vibration damping enclosure yields large internal volume and high mechanical strength. The integrated advanced Directivity Control Waveguide (DCW) enables neutral reproduction, particularly in acoustically compromised spaces. And, its novel Laminar Integrated Port (LIP) bass reflex system provides accurate low-frequency response and faithful tonal reproduction characteristics.

Ecologically Sustainable and Recyclable

Since its founding, Genelec's design philosophy has been based on sustainability and environmental values. Conservation of natural resources and efficient use of material and energy in all levels

Photo 1: The Genelec M030 features a Natural Composite Enclosure (NCE) in patterned finish.

of manufacturing, shipping, and during the product lifetime is essential to the brand. Genelec's low life cycle carbon footprint products are manufactured under the same roof, in Finland, which lead to environmentally efficient solutions.

For example, Genelec's factory is using renewable energy for its heating. The recycled aluminum used in the die-cast enclosure manufacturing saves 95% of energy compared to using virgin aluminum. The wood fibers used in the M Series enclosure material is sourced from sustainably managed forests. The M Series packing is made of recyclable cardboard. Shock absorber linings inside the packing are made of recycled paper pulp.

Conserving natural resources extends beyond the use of recyclable natural materials. From start to finish, the design principle for the M Series has been to make it ecologically sustainable. Products have been designed to be durable and have a long lifetime. Materials are selected so that they can be reused, either in their original purpose or as new types of products. This is the best form of sustainability.

Innovative Natural Composite Enclosure

Early in the company's history, extensive research was made to find new enclosure materials. In 1983, Genelec

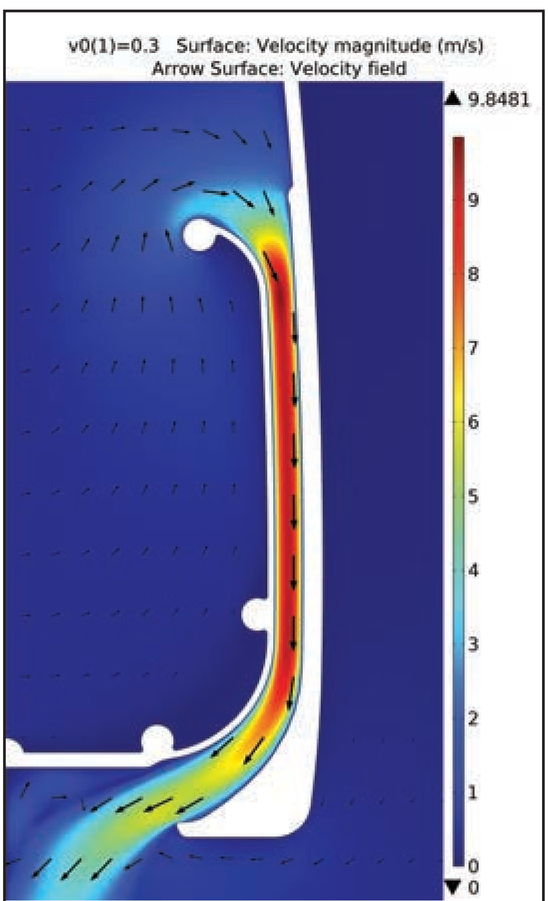


Figure 1: The cross-section plot of the flow-optimized Laminar Integral Port (LIP) is integrated into the Natural Composite Enclosure (NCE).

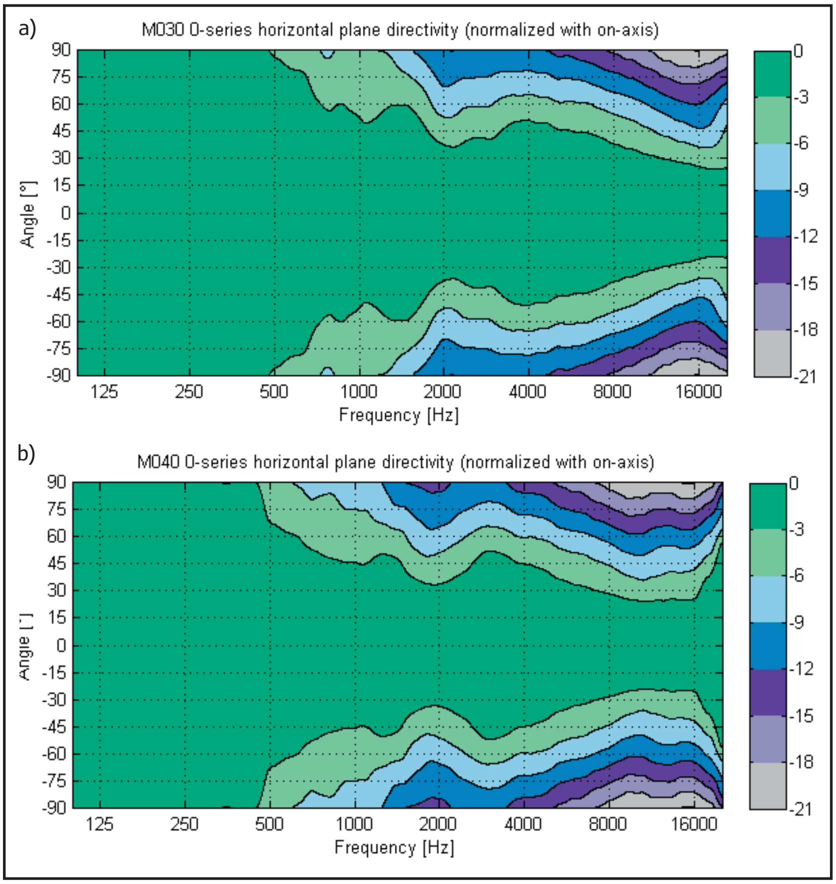


Figure 2: Directivity plots of M Series products including, the M030 (a) and the M040 (b), demonstrate excellent horizontal directivity control.

made its first prototypes of the 1022A , using an enclosure made of hand-laid glass fiber resin layers. The design came into production in 1985. This radical development in enclosure shape paved the way for diffraction-free enclosures and DCWs. In 1996, Genelec introduced its first die-cast aluminum enclosure, with the small 1029A active two-way monitor. All the benefits of cast aluminum were immediately apparent including flexibility in enclosure shape design, structural stiffness, and a large internal volume in relation to external size.

Beginning in 2003, Genelec participated in a university research project studying the use of wood composites for injection molding. In 2006, the first exploratory loudspeaker enclosures in wood-based natural composite material were made. In 2009 Genelec started a program to develop loudspeaker enclosures containing half wood fibers.

The innovative NCE of the M Series products is made of a material that is best described as injection-moldable wood. This material is a natural fiber composite with about half natural wood fibers. Other materials include color pigments, flame retardants, lubricants, and so forth, which are all integrated in a polypropylene matrix. Polypropylene is a recyclable material (polymer resin identification code number 5) that features a low melting point ranging between 130°C and 170°C. This environmentally attractive polymer has been selected for its resistance and durability, its high internal damping and its resilience against impacts and physical damage.

In terms of manufacturing, the injection-molding process allows design of structures with small wall thicknesses and multiple internal support and bracing to increase the enclosure stiffness. Production of acoustically highly optimized outer shapes and forms is also possible. At the same time, the enclosure internal volume is maximized. This is paramount to achieve high acoustic output at low frequencies. The material itself has high internal vibration energy losses, which is advantageous in enclosure design where damping and rigidity is required.

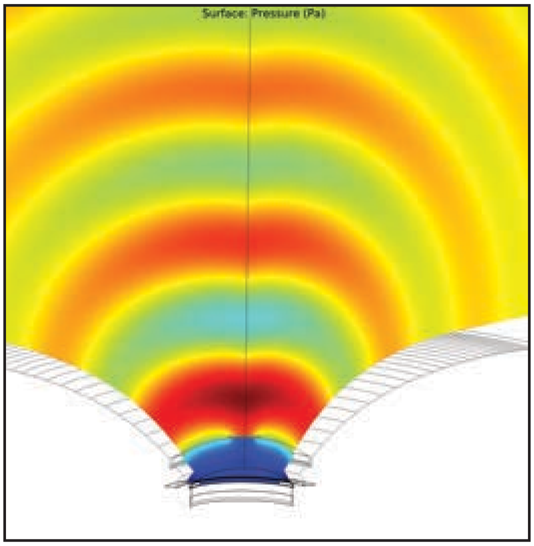


Figure 3: Optimized with Finite Element Analysis (FEA) simulations, the M Series injection-molded Natural Composite Enclosure (NCE) integrates an efficient Directivity Control Waveguide.

Unpainted Enclosure Finish

The highly renewable NCE enclosure material has many of the outstanding acoustical properties found in wood fibers. It is 100% stiffer than the common ABS plastics typically used in loudspeaker enclosures. Due to the high quality of the wood fibers and the virgin polypropylene material used, the resulting NCE raw granulates are clean and odorless. Additionally, the often environmentally hazardous painting has been eliminated from the production process, which saves on transportation and handling. In fact, enclosure front and back parts are manufactured as finished elements, ready for assembly, in factories near Genelec's manufacturing plant. The unique patterned finish is created during the injection-molding process by the interaction and combination between the wood fibers and the polypropylene matrix.

Laminar Reflex Port

Genelec's choice of vented (or bass reflex) enclosures dates back to 1978 and its first active monitor, the S30. The company has pursued research to improve the performance and efficiency of reflex ports ever since. A typical reflex port enclosure features a tube and an opening area. To avoid turbulences in the tube, the airflow should not meet any acute angles, as this would generate noise, compression, distortion and losses of the total radiated energy. To minimize the air flow speed, both the tube and its cross section have to be large. Often, the outer enclosure dimensions become a limitation, because a long tube will not fit in the available volume anymore.

The M Series features two vent tubes with openings stretching across half of the enclosure's depth to address these specific issues. The novel patent-pending LIP has been flow-optimized using computer-based finite element models to achieve low distortion and high efficiency even at very high audio output levels (see **Figure 1**). The reflex ports are integrated in the NCE enclosure during the molding process, avoiding the need for separate additional components. The cross-section plot demonstrates the efficient flow characteristics of the port. The natural installation orientation of the M Series is vertical and to enable easy placement of the monitor against a wall, the M Series ports opening face downwards, in the space under the monitor.

Efficient Directivity Control

Genelec's pioneering DCW technology was first developed in the early 1980s to significantly improve the performance of direct-radiating multi-way monitors. The DCW waveguide is designed to match the frequency response and directivity of midrange driver to woofer and high-frequency driver to midrange. This results in excellent flatness of the overall frequency response for on- and off-axis listening positions (see **Figure 2**). As a consequence, the reflected sound energy at the listening position is reduced. Controlled directivity is associated with neutral uncolored sound, particularly in acoustically compromised spaces.

With a heritage of 30 years of DCW design, the M Series injection-molded NCE enclosure was developed to feature a large, highly efficient DCW integrated into the enclosure front structure (see **Figure 3**). The directivity characteristics have also been matched between the two monitor models, resulting in similar neutral sounding products.

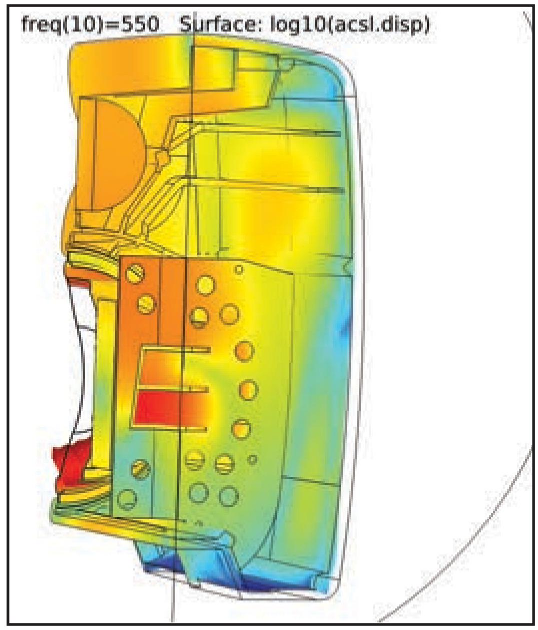


Figure 4: The Genelec M040 enclosure simulation is shown using the finite element analysis (FEA) method.

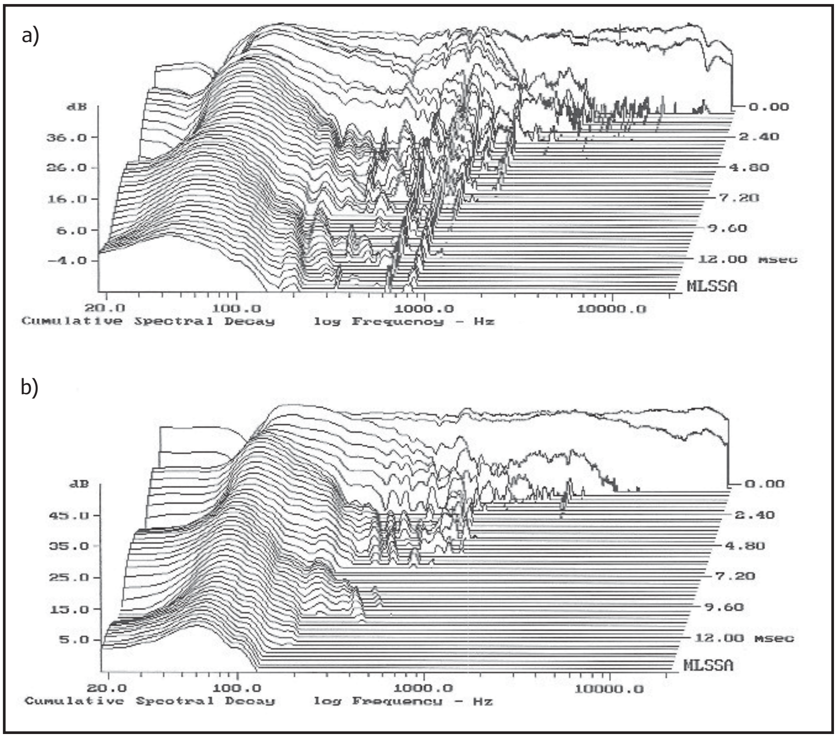


Figure 5: Waterfall plots show the release of resonant energy for a typical two-way active monitor (a) and the Genelec M030 (b).

Low-Diffraction Enclosure

One of the important phenomena affecting sound radiation is called diffraction. Any acoustical discontinuities around the drivers (e.g., enclosure edges) act as secondary sound sources because of diffraction. The typical four edges of the baffle front create a system that has five radiators operating at the frequencies of interest. These are the actual driver and the four secondary sources created by diffraction at the four cabinet edges. The resulting frequency response at the listening position is the sum of these sources. As the listener moves off axis, the time relationships of the different sound sources will change and the degradation to the monitor response will vary with location.

Pioneered by Genelec since 1985, carefully contoured smooth acoustically optimized rounded enclosure shapes can minimize diffraction and improve sound quality. Enclosure corner shapes of the M Series products have been optimized to minimize any sound coloring diffraction from the enclosure edges and corners. In addition to achieving an unsurpassed flatness of the frequency response, minimum diffraction enclosures yield accurate imaging.

Absence of Delayed Resonances

When an audio signal fed to a monitor stops, there should be nothing left but silence. Resonating enclosure structures store mechanical energy. This energy will be slowly released at the resonance frequencies, after the audio stops, and instead of silence there will still be some output. Release of this stored resonant energy can color the audio we hear.

The challenge in enclosure design is to minimize and damp structural resonances (see **Figure 4**). Detailed measurements and study of the release of resonant energy has allowed Genelec to understand were resonances happen and what are their mechanisms (see **Figure 5a**).The M Series acoustical radiating system—consisting of enclosure, drivers, and reflex ports—has been carefully modeled and optimized to keep resonances inaudible. The outcome is audio accuracy and the capability to reproduce details and nuances without masking (see **Figure 5b**).

Distinctive Finnish Engineering

For more than 35 years, Genelec has been guided by a single idea—to make perfect active monitors that deliver neutral and accurate sound in every kind of acoustical environment. The essence of the company's engineering philosophy is to use industrial design to serve the products' acoustical performance. For many years, Genelec has cooperated with renowned Finnish industrial designer Harri Koskinen. His involvement in the designs brings a unique voice to the aesthetics of Genelec's products.

Dedicated to the Music Creation work, Genelec's new M Series provides a unique combination of high linearity, low distortion, clean and neutral audio reproduction. The products feature room compensation adjustments, universal power supplies, and versatile inputs. Adapted to mobile work, M Series products respect sustainability and environmental values, with low life cycle carbon footprints. Genelec's efforts to improve all aspects of monitoring has led to the use of NCE, an environmentally friendly material featuring outstanding mechanical and acoustical properties.

About the Author

Christophe Anet holds a Bachelor of Engineering degree with honors in electroacoustics from the University of Salford, Manchester, UK. Starting his career at Genelec Oy as an R&D trainee, he has held various positions including customer support acoustic engineer and technical editor. He has also worked in environmental acoustics and traffic noise control for the Swiss state administration, as well as for leading architectural acoustics consultant Walters-Storyk Design Group.

Christophe is now the education and training manager at Genelec. He is a regular writer for professional audio magazines.

Fluent in French, English, Italian, and German, Christophe is also proficient in a wide range of software tools, including acoustical measurement equipment and graphic design.

High-Performance Audio Playback from Smartphones

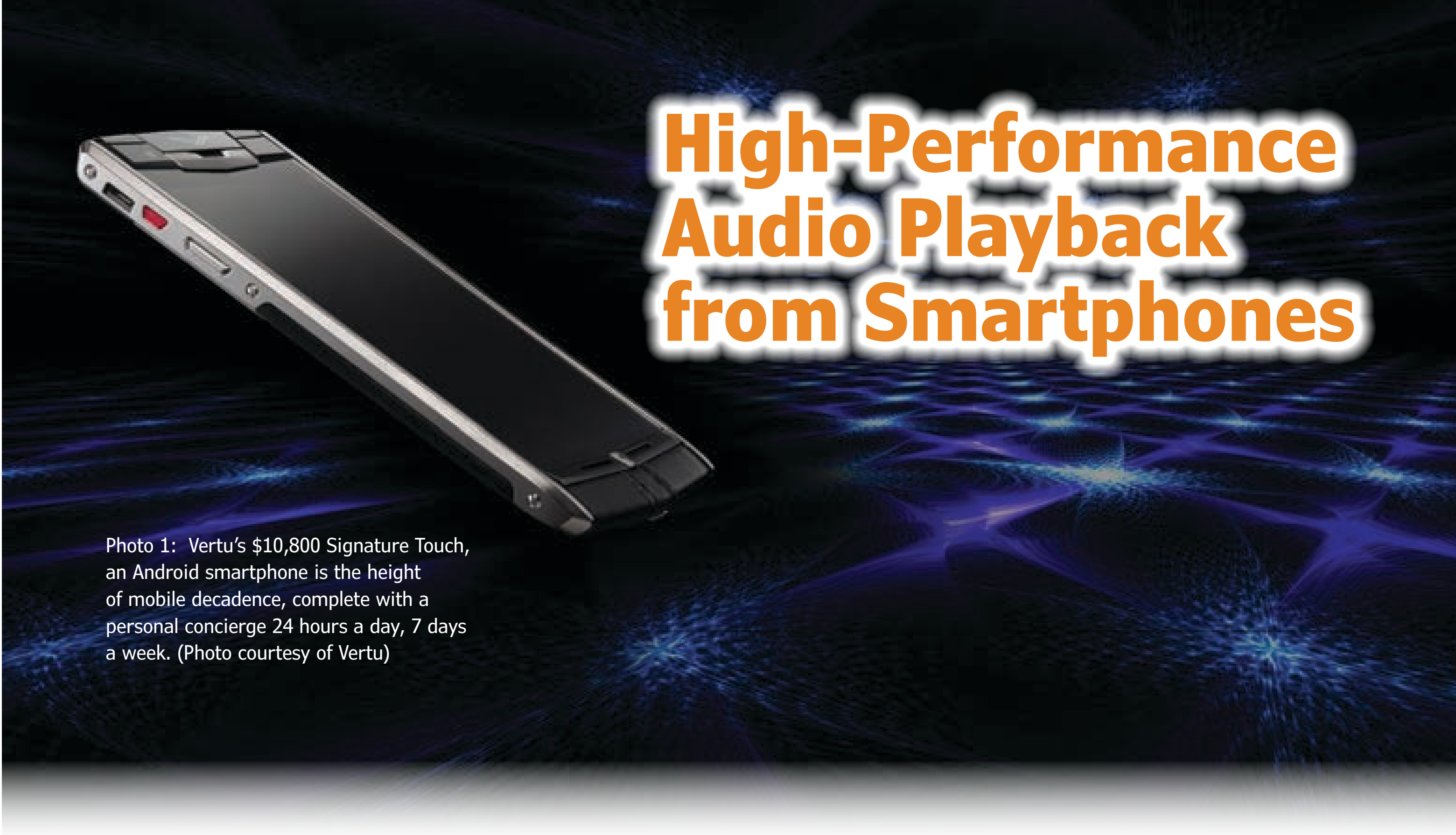


Photo 1: Vertu's $10,800 Signature Touch, an Android smartphone is the height of mobile decadence, complete with a personal concierge 24 hours a day, 7 days a week. (Photo courtesy of Vertu)

This article discusses the average consumer's desire for high-end high-performance smartphones and the ways manufacturers are changing their designs to increase audio quality. The result is feature-filled smartphones that contain high-quality audio delivery systems.

By
Mike Klasco and Steve Tatarunis
(United States)

The market for boutique high-performance smartphones is just starting to appeal to the average consumer. For many years, there have been a limited number of designer phones including the Vertu Signature ($10,000), the Dior ($100,000) and the Tag Heuer Link ($12,000). However, these phones were technologically dated even when they were first introduced.

These products were glorified feature phones or smartphones with a fancy crust (see **Photos 1–3**). The target market was wealthy individuals into haute couture. However, this market is limited because most people with big money also have big brains. They want designer smartphones with great technology, but until recently, these products have been missing from the market.

Early Smartphones

The early smartphones with MP3 stereo music playback as well as the original standalone iPods and MP3 players dropped playback standards down a couple of notches from CD quality by using marginal bit rates. The manufacturers' primary goals were portability and affordability. To achieve these goals, compromises were made particularly in the area of audio playback quality.

In spite of this acceptance of marginal audio quality, there has been a welcome introduction of decent to even awesome on-the-go earphones with built-in microphones for calls. The earphones are part of what was missing in the early smartphones. But now, the pieces are finally coming together to provide superb on-the-go audio delivery systems bundled into feature-filled smartphones.

Apple, Samsung, LG, and HTC are competing to offer the best high-end smartphones that are powered by multi-core processors. The smartphones are loaded with tons of RAM, offer higher-definition displays, and the latest 4G connectivity. Thankfully, the manufacturers are finally expanding their focus to include audio performance, and the winner is the consumer. Even Vertu now offers a technically credible

Photo 2: Christian Dior, world-acclaimed for his Haute-Couture fashion house, launched the Dior Phone Touch in 2011, a state-of-the-art, compact, and high-performance smartphone. (Photo courtesy of Dior)

product with audio "tuning" by Bang & Olufsen called the Constellation T Haute Couture ($6,000–$15,000).

However, there are other features to consider when making a smartphone purchase, including physical robustness (e.g., water resistance, etc.), privacy, and security. But for now, we will focus on smartphone music playback audio quality.

The most important aspect of smartphone audio quality is music playback through earphones and headphones. Almost all smartphone users have some or all of their music collection on their smartphones, and many of them place extraordinary value on this feature.

Photo 3: After more than 150 years of watchmaking, TAG Heuer began offering high-end Android smartphones, including the Tag Heuer Link Smartphone. (Photo courtesy of Tag Heuer)

Headphone listening is still predominantly done through wired connectivity and so we will evaluate what's coming out of that tiny headphone jack and leave discussions on the emerging Bluetooth audio market for another day. Another consideration when it comes to music playback performance—in addition to headset playback quality—is that the smartphone is often used as a music server for home audio systems. Many consumers connect their smartphones to their home systems via a headphone jack.

As a side note, a new Japanese trend goes against this search for all-in-one audio perfection by offering awkward, geeky stack-ups of DACS and headphone amps duct taped together.

But back to the high-end offerings, there are also some very fine premium music players on the market, such as the Astell & Kern AK240 high-resolution digital analog converter player ($2,500) and the Fostex HP-P1 Portable Amp & DAC, which offers combo digital storage and DAC-plus headphone amplifiers. It is very nice, but you still need a smartphone (see **Photo 4**).

There is no mystery when it comes to what company makes the top DACS and headphone amplifier ICs. Yet, the question arises as to why not design a smartphone that uses both the latest smartphone chipset with the top-end DACs and headphone amps? Part of the reason is power consumption, which excludes some of the very high-end DACs. However, it turns out there is an obscure (at least in the US market) Chinese high-end smartphone vendor that has figured out a way to make it work.

So our challenge for next month's column is to compare the music playback audio quality of a half dozen portable devices. These products include one non-smartphone (the iconic iPod classic), several smartphones known for decent audio quality (e.g., the Samsung Galaxy, the Sony Xperia Z2, the HTC One M8, the iPhone 5s, and the Vivo Xplay 3S. Vivo's main design goal with the Xplay 3S is to bring superior stereo audio playback quality to the Android smartphone platform.

Perhaps HD audio and HD Voice have been the impetus for smartphone manufacturers to think not just about the camera pixels and screen resolution, but also that the phone might be used for voice communications (sorry for the sarcasm). This could be an opportunity for product differentiation. HTC has made a stand with the HTC One M8 (see **Photo 5**). The HTC One M8 raises the audio standard among the Big 3 (Apple, Samsung, and HTC) with its BoomSound—a richer and louder amplifier, a new DSP, and deep chambers housing the speakers.

We have evaluated the devices with earphones, headphones, and playback though speaker systems, as well as comprehensive instrumented tests using the Listen SoundCheck audio analyzer. Both the data and subjective listening test results will be discussed next month.

HD Audio

HD Audio, now commonly known as High-Resolution Audio (HRA), comes into play in a few different areas—stereo music playback through earphones, using the smartphone as a music server, and perhaps someday through the speakers in the smartphone. Of course, we can simply limit our focus on the bit rate and clock speed—24 bit/96 kHz is better than CD quality (16 bit/44 kHz)—and the memory capacity in smartphones is enough to accommodate more music than you can handle in high definition. Lossless formats such as FLAC (or ALAC for Apple users) can be either CD quality (16 bit/44 kHz) or master quality (24 bit/96 kHz).

One by one, smartphone makers have been jumping on the "HD Audio" bandwagon with 24-bit/96-kHz-compatible conversion chips. HD Audio can go further but first there needs to be more widespread availability of program delivery services and then there is the performance of the playback DAC, which needs some attention in this discussion.

Why Care About the DAC?

When Philips introduced the first CD player in early 1980s, the sound was miserable. At first listen, the lack of hiss compared to cassettes was refreshing, but after a while the sound was fatiguing. What was wrong could fill a book, but the last 30 years of digital audio design have addressed most of the issues. One dirty little secret was the intermodulation distortion (IM) that was always missing from the early datasheets of the DACS, often 3% IM distortion or more. IM is nasty and the least musical distortion. It is often the source of listening fatigue—the stuff that makes you reach for the off switch after a while. IM primarily originates in the DAC. There were also another couple of less than pristine performance parameters that afflicted DACs. And, smartphones have so much digital and radio frequency (RF) stuff going on, it is impossible to wrestle the best audio from even great DACs. Nevertheless, recent smartphone introductions not only have top-grade DACs but some smartphone designers have also managed to extract very fine performances from the DACs and the rest of the audio system.

Photo 4: The Fostex HP-P1 portable amplifier and DAC is shown as a "stackup" with an iPhone. (Photo courtesy of Fostex)

Next month, we will look at the performance of six devices. Here's a quick summary of the devices we plan to review:

Apple iPod Classic—The iPod started the whole portable music craze when it was first introduced in October 2001. The current version plays video and audio and has 160 GB of onboard storage. Supported audio formats include AAC, Protected AAC, HE-AAC, MP3, MP3 VBR, Audible (Formats 2, 3, 4, Audible Enhanced Audio, AAX and AAX+), Apple Lossless, AIFF, and WAV. In addition to a headphone jack, an audio signal is also available via the proprietary Apple dock connector.

Samsung Galaxy S5—Samsung's flagship Android phone was released in April 2014 and its strong sales helped Samsung capture over 32% of the global

Photo 5: The HTC One (M8) is the newest version of Android. It is fast and powerful yet simple to use. (Photo courtesy of HTC Corp.)

smartphone market in the second quarter of 2014. The Samsung Galaxy S5 has 32 GB of onboard storage, with another 128 GB available via micro SD. Supported audio formats include MP3, M4A, 3GA, AAC, OGG, OGA, WAV, WMA, AMR, AWB, FLAC, MID, MIDI, XMF, MXMF, IMY, RTTTL, RTX and OTA.

Sony Xperia Z2—Given Sony's audio heritage, it is not surprising to see them place an emphasis on quality audio delivery in smartphones. The Sony Xperia Z2 offers several audio enhancements, 32 GB of onboard storage with an extra 64 GB available via micro SD. Supported audio formats include MP3, 3GPP, MP4, SMF, WAV, OTA and OGGMP3, 3GPP, MP4, SMF, WAV, OTA, OGG MP3, 3GPP, MP4, SMF, WAV, OTA, OGG MP3, 3GPP, MP4, SMF, WAV, OTA, OGG MP3, 3GPP, MP4, SMF, WAV, OTA, and OGG.

Apple iPhone 5s—Apple has always paid attention to smartphone audio quality and the iPhone 5s is no exception. It comes with up to 64 GB of onboard storage and supports AAC, Protected AAC, HE-AAC, MP3, MP3 VBR, Audible (Formats 2, 3, 4, Audible Enhanced Audio, AAX and AAX+), Apple Lossless, AIFF, and WAV formats. In addition to the headphone jack, HD audio is also available from the proprietary Lightning connector. Rumor has it that future iPhones may eliminate the standard headphone jack in favor of Lightning-only connectivity.

HTC One M8—The HTC One M8 has a pair of front facing speakers (one on each side of the display) and HTC touts its on-board "Boom Sound" listening experience. It offers 32 GB of onboard storage with up to 128 GB available via micro SD expansion and will play all popular audio formats.

Vivo Xplay 3S—Vivo is a Chinese brand and its Xplay 3S has the distinction of being the first smartphone offered with a 2K display. An equal emphasis is placed on audio, using the ESS Technology ES9018 DAC, the Texas Instruments OPA2604 amplifier, as well as DTS Headphones:X for surround sound support. The Xplay S3 has 32 GB of onboard storage and expansion via micro SD.

Pristine stereo music playback is a genuine enhancement. But let's not forget clear and intelligible speech communications even in noisy environments is a top priority. After next month's testing for stereo audio quality, we will explore the A/D codecs and voice co-processors used in the latest smartphones.

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Measurement Microphones

To take a scientific measurement with a microphone, we must know its precise sensitivity (in volts per pascal). Since this may change over the device's lifetime, it is necessary to regularly calibrate measurement microphones. This service is offered by some microphone manufacturers and by independent certified testing labs. However, all microphone calibration is ultimately traceable to primary standards at a national measurement institute. This article discusses some of the laboratory-standard microphones calibrated using this method, which are used to calibrate other microphones through comparison calibration techniques.

By
Richard Honeycutt
(United States)

Photo 1: The Western Electric 640AA was the first measurement microphone used at the National Bureau of Standards, which is now known as the National Institute of Standards and Technology (NIST). It is still used by the NIST today.

The first standard measurement microphone was the Western Electric 640AA "silver bullet" capacitor microphone (see **Photo 1**). One characteristic of this microphone that made it suitable for use as a standard test microphone is that its sensitivity—the signal voltage output at a specified sound pressure level (SPL)—can be calculated from its physical dimensions and the polarizing voltage. The National Institute of Standards and Technology (NIST) still offers pressure calibration of test microphones against the 640AA as a standard service.

Although the "Western Electric" name is now owned by a vacuum tube manufacturer/supplier and the company does not manufacture microphones, many other excellent manufacturers do build capacitor measurement microphones. **Photo 2** shows three Brüel & Kjær Sound & Vibration Measurement A/S test microphones attached to preamplifiers—the 4954B and the 4189 are prepolarized electret microphones and the 4191 has a 200-V polarization voltage applied by the preamplifier.

Photo 2: Three Brüel & Kjær measurement microphone capsules—Models 4954B (a), 4189 (b), and 4191 (c)—are shown with preamplifiers. (Photos courtesy of Brüel & Kjær)

The ACO Pacific package (see **Photo 3**) contains an MK2324 prepolarized capsule, a 4048 phantom-powered preamplifier, and a WS1 windscreen. The microphones shown in **Photos 4–6** are electret microphones with integral preamplifiers powered through a phantom power supply. **Photo 7** shows an electret measurement microphone designed for operation with i-mobile devices and powered through the dock connector. The microphones shown in **Photos 2–6** span a price range from under $50 to more than $2,000.

Photo 3: This ACO Pacific MK224PH package includes a measurement microphone, a preamplifier, and a windscreen. (Photo courtesy ACO Pacific)

Measurement Microphone Types

Today, capacitor measurement microphones are available in three varieties. The first is an externally polarized type, requiring an external power supply (typically 200 V), which may be built into a preamplifier that is usually screw-mounted to the capacitor microphone capsule.

The second variety of capacitor microphone is the film-electret type, which uses a prepolarized plastic film diaphragm and requires no external polarizing voltage. The film electret was invented in the 1960s at the Bell Telephone Laboratories by Gerhard Sessler and Jim West. Sessler and West called their invention “foil electret microphones.” After experimenting with a variety of plastic films, they identified a mechanism by which the polarization of a film gradually decreased with time, lowering the microphone’s sensitivity. They determined that fluorinated ethylene propylene (FEP) Teflon had the most stable charge over time of any material they tested. Later, Preston Murphy and Freeman Fraim of Thermo Electron Corp. demonstrated that superior charge stability could be achieved by using polyvinylidene chloride rather than FEP Teflon. Even so, electret microphone sensitivity declined over the years.

Photo 4: The Soundfirst SF101a, shown in its case, is a lower-priced measurement microphone. (Photo courtesy of SoundFirst.com)

Depolarization

Depolarization occurs according to an approximately inverse exponential function. Soon after manufacture, the electret self-depolarizes comparatively rapidly, but the rate of depolarization decreases with time. Depolarization effects can be reduced by exposing the electret material to elevated temperatures, during which most of the early rapid

Photo 5: The AUDIX TM1 measurement microphone has its own clip, calibrator adapter, and calibration data disc. (Photo courtesy of AUDIX Microphones)

depolarization occurs, leaving the polarization more stable over the remaining life of the microphone. This process is called "preconditioning," and the better electret measurement microphones are preconditioned.

Film electret microphone diaphragms must be made of plastic films having good electret properties. This can lead to compromises with regard to the mechanical properties of the diaphragm. So, we now come to the third variety of capacitor microphone: the back electret microphone. In this configuration, the electret material is applied to the microphone capsule's back plate. Metal (nickel, stainless steel, or titanium) or an unpolarized metalized film—often Mylar—is used for the diaphragm.

Figure 1 shows section views of externally polarized, film electret, and back electret microphones. The stability of a capacitor test microphone's sensitivity depends on the stability of the diaphragm tension. For the metal diaphragms used in externally polarized microphones, the sensitivity can be stable to within 1 dB per 1,000 years at 20°C. Stability is not as good at higher temperatures, dropping to about 1 dB per 100 hours at 150°C. Electret microphones suffer from depolarization over time, especially at high temperatures. So while a preconditioned back electret microphone can be stable to within 1 dB per 100 years at 20°C, the stability can drop to 0.5 dB per hour at 150°C.

Film electret microphones can be less expensive than externally polarized models. However, even with preconditioning, film electret microphones have less stable performance over time than do externally polarized microphones. This is true not only because of the electret's depolarization but also because of decreased diaphragm tension causing a change in the capsule's resonance frequency, which affects its frequency response. Any microphone with a plastic film diaphragm is subject to changes in performance when exposed to high temperatures.

Another disadvantage of plastic film diaphragms is their low tensile strength, which limits the tension that can be applied, and thus, limits extreme high-frequency response.

An advantage of electret microphones is that they can be used in high-humidity environments where an externally polarized metal-diaphragm microphone might experience arc-over due to the high polarizing voltage between the diaphragm and back plate.

Our longtime readers will remember crystal and ceramic phono cartridges, and maybe even crystal and ceramic microphones. These devices utilized the phenomenon of piezoelectricity (from the Greek "piezon," to press). This refers to the property by which some materials (e.g., quartz or Rochelle salt crystals and lead zirconate titanate) produce a voltage when physically stressed. Before the development of electret microphones, piezoelectric measurement microphones were used when cost, humidity sensitivity, or mechanical ruggedness were important considerations. The low manufacturing cost of electret microphones

Photo 6: The Dayton EMM-6 measurement microphone is often used for electroacoustic testing. (Photo courtesy of Parts Express International)

Photo 7: The Studio Six Digital iTestMic is designed to provide accurate measurement capability for an iPod or an iPhone. (Photo courtesy of Studio Six Digital)

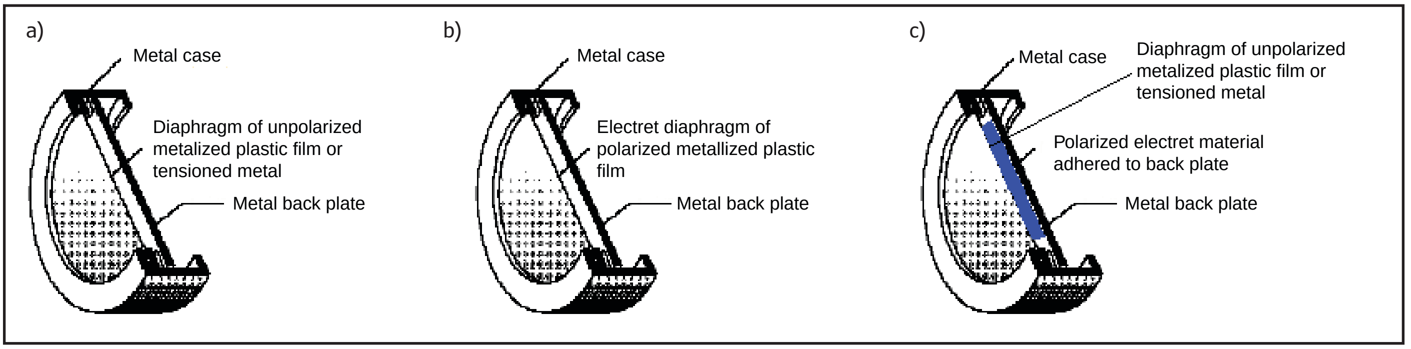


Figure 1: The three varieties of capacitor microphone capsule—the externally polarized capacitor microphone (a), the film electret microphone (b), and the back electret microphone (c)—are similar, but not identical.

Class	Sound Level Meter Accuracy	Calibrator Accuracy
Type 0	±0.4 dB	±0.15 dB
Type 1	±0.7	±0.3 dB
Type 2	±1 dB	±0.5 dB
Type 3	±1.5 dB	What calibrator?
Not spec'd	Who knows?	Oh, come now.

Table 1: ANSI types for measurement microphones indicate differing sound level meter and calibrator accuracy.

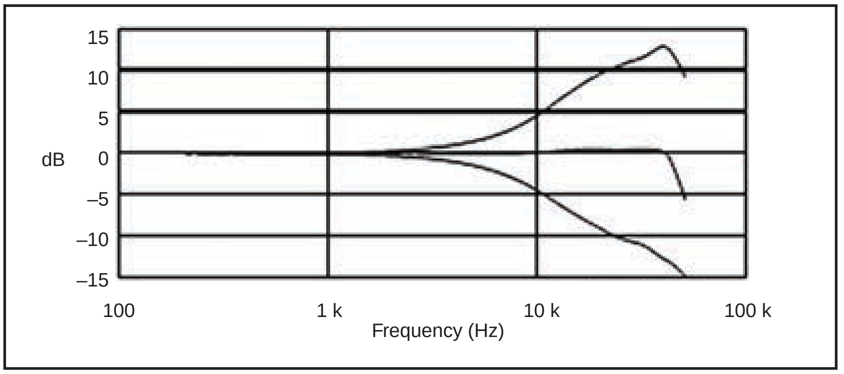


Figure 2: Free-field and pressure frequency responses differ at high frequencies. (Image used with permission from Brüel & Kjær)

has essentially eliminated piezoelectric measuring microphones from the market.

Important Performance Characteristics

The important performance characteristics of a measurement microphone include sensitivity, frequency-response limits and flatness, noise floor, maximum SPL, and the directivity or the angle of incidence at which the frequency response is specified. Sensitivity is specified in volts per pascal (V/Pa), which is the SI unit of sound pressure, or sensitivity response level (SRL), which is the ratio of the microphone's V/Pa output to the 1 V/Pa reference sensitivity expressed in decibels:

$$\text{SRL} = 20\log_{10}\left(\frac{\text{mic's V/Pa}}{1\ \text{V/Pa}}\right)$$

Thus, a microphone that produced an output voltage of 5 mV at a sound pressure of 2 Pa (100 dBSPL) would have an SRL of:

$$20\log_{10}\left(\frac{0.005/2}{1}\right) = -52.04\ \text{dB}$$

Measurement microphone sensitivities range from −80.9 to −20 dB.

Minimum and maximum frequency limits for currently available measurement microphones are 0.5 Hz and 140 kHz, respectively, although

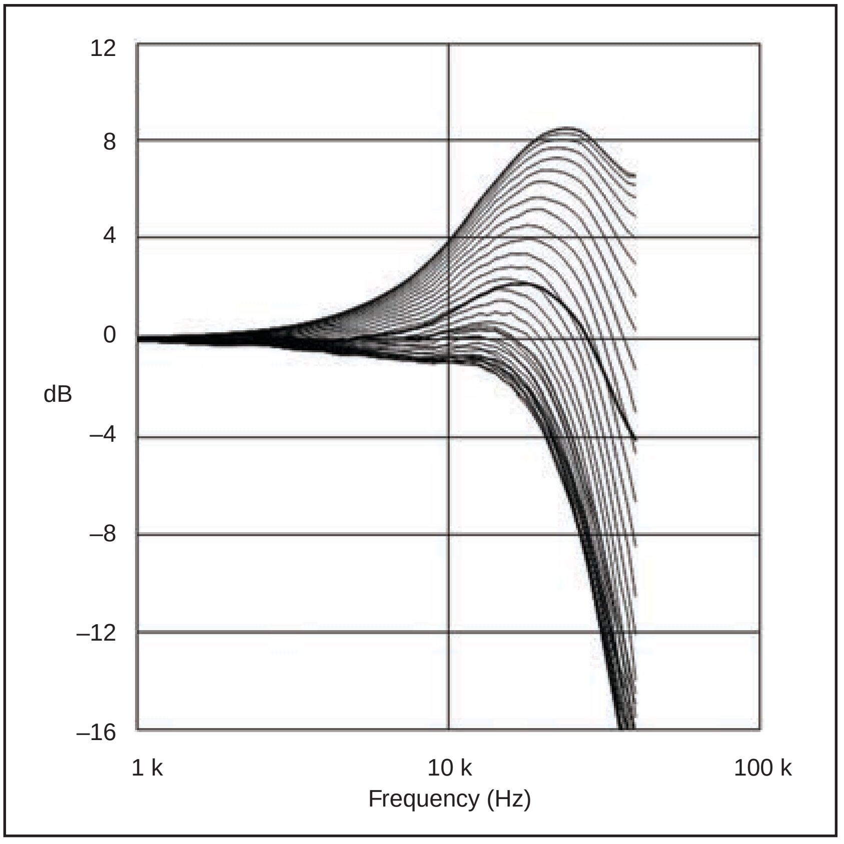


Figure 3: These curves show the free-field corrections for a particular measurement microphone's pressure calibration. (Image used with permission from Brüel & Kjær)

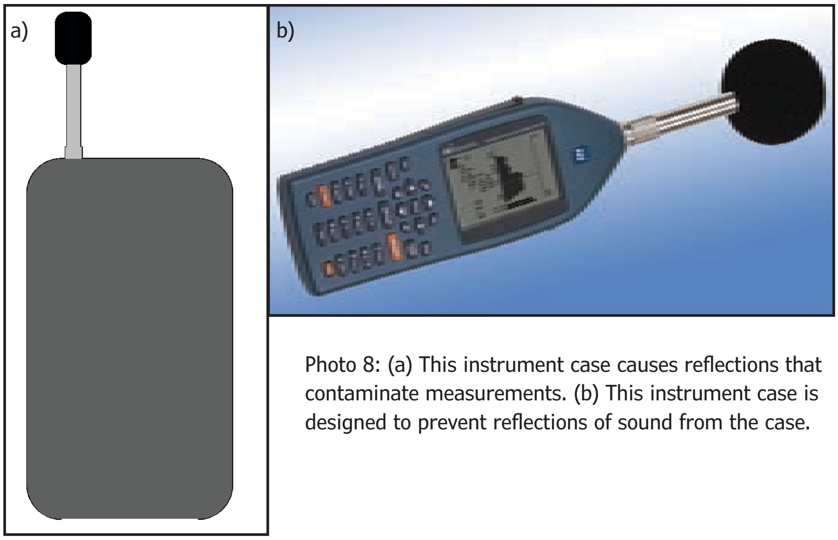


Photo 8: (a) This instrument case causes reflections that contaminate measurements. (b) This instrument case is designed to prevent reflections of sound from the case.

more commonly the limits are 20 Hz and 20 kHz. Frequency-response flatness can be determined by a calibration chart provided in either graphical or numerical form by the manufacturer, or by the American National Standards Institute (ANSI) type. **Table 1** shows the required response flatness that a measurement microphone must exhibit to qualify as a Type 0, 1, or 2.

Measurement microphone sensitivity vs. frequency can be measured and specified for any of three types of sound field. A "pressure field" is a region throughout which the sound pressure has the same magnitude and phase. The small cylindrical couplers used in microphone calibrators and pistonphones produce pressure fields. A "free field" is a region in which the sound waves propagate as plane waves in one direction, with no reflections. A "diffuse field" is a region in which sound waves can arrive with equal probability from any direction.

Thus, microphone specifications can include pressure sensitivity, free-field sensitivity, or diffuse-field sensitivity (sometimes called random-incidence sensitivity). **Figure 2** shows the pressure and free-field sensitivities of a microphone. The middle curve is the free-field response. This is sometimes referred to as the on-axis response. The lower curve is the pressure sensitivity. The upper curve shows the free-field correction, which is applied to the pressure response to get the free-field response. The difference in high-frequency response between the free-field and pressure sensitivities is a result of interference between direct and reflected waves increasing the sound pressure at the diaphragm, beginning at frequencies where the wavelength approaches the diaphragm diameter, and increasing with frequency. The microphone's protection grid also produces its own (but smaller) increase at the highest frequencies. This effect does not occur in a pressure configuration. The diffuse-field correction can be calculated from the pressure response, using the method described in International Electrotechnical Commission (IEC) standard 1183. **Figure 3** shows an example of the free-field corrections for a 0.5" microphone with no protection grid, taken at 5° intervals. These correction curves can be used to obtain the frequency response of a 0.5" measurement microphone if the frequency response of the pressure sensitivity is known. The diffuse-field correction is indicated in the figure by the bold curve.

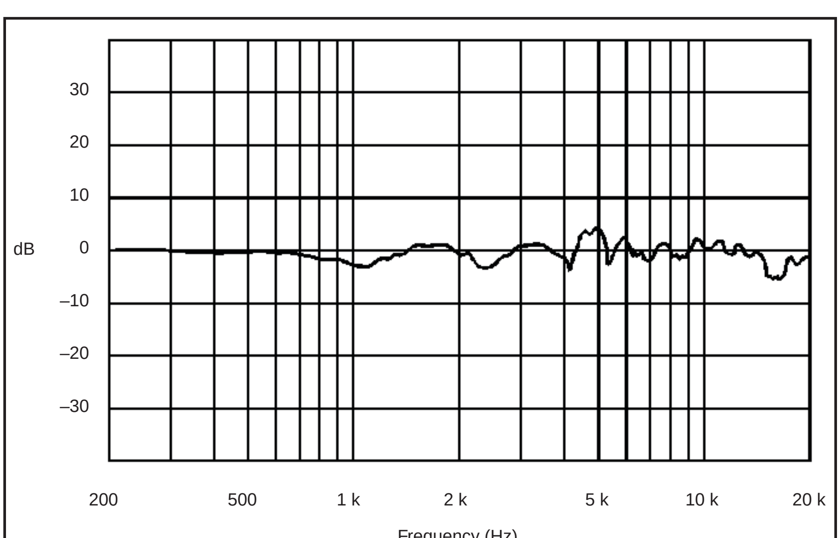


Figure 4: Reflections from an instrument case can contaminate measurements.

Calibration

Any measurement microphone should be regularly calibrated using an NIST-traceable calibrator, to ensure that the sensitivity is known. Otherwise, measurements will not be reliable. Usually, calibration only tests the microphone's sensitivity at one frequency (typically, 400 Hz or 1 kHz), but not the microphone's frequency response. Specialized frequency-response calibrators are available from some test equipment manufacturers.

The minimum SPL at which a measurement microphone is useful is determined by the microphone's equivalent input noise (EIN), which is specified in A-weighted decibels (dBA). This is the A-weighted noise level of a sound field in which a theoretical noiseless microphone having the same sensitivity as the microphone being specified would have to be placed to generate the same electrical output.

In fact, the A-weighting is problematic for two reasons. A-weighting is based on human

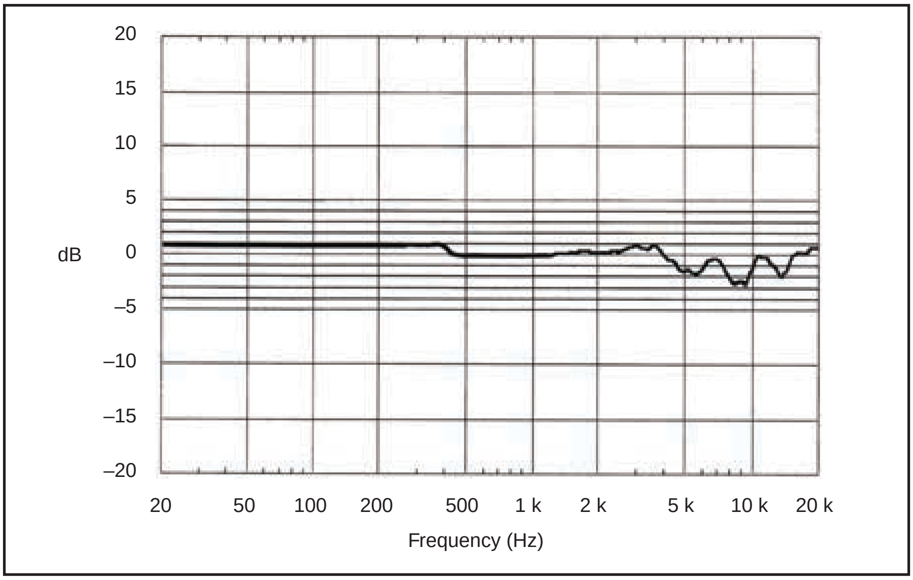


Figure 5: Reflections from a microphone clip can also contaminate measurements.

subjective response to noise occurring in the 50-phon loudness range. The use of A-weighting is questionable for objective measurements (since it is based on subjective criteria). To correctly weight noise based on human annoyance, the subjective response would have to be based on the equal-loudness curve corresponding to the microphone's EIN—within the range from –2.5 to about 50 dBA for currently available test microphones. An even more convincing argument against specifying measurement-microphone self noise in A-weighted decibels is that it is not human response to the noise that is the issue in testing applications. It is the SNR of the microphone's electrical output.

Noted microphone design engineer David Josephson of Josephson Engineering states, "Both the A-weighting and the RMS detector tend to hide poor quality mics, because low-frequency rumble and short-period events [shot noise] in the noise floor don't contribute much to the 'dB(A)' result. There are microphones that are equalized specifically to produce very low A-weighted noise numbers even though the broadband noise floor might be higher than a comparable microphone's figures without such treatment. This was done decades ago by Brüel & Kjær, not for specsmanship but because it can be useful to have a microphone with sub-zero dB SPL noise in the midrange, even at the expense of restricted high-frequency (HF) response or other compromises."

"It is possible to make a FET impedance converter with a low 1/f noise corner, but difficult. FETs are usually specified for noise at 1 kHz or 10 kHz, and FET suppliers seldom want to talk about noise at 10 Hz, much less give a curve for noise vs. frequency."

Equivalent Input Noise

At present, the EIN of measurement microphones is stated in A-weighted decibels, although some manufacturers choose to specify both A-weighted and unweighted noise.

There are two noise sources in a capacitor microphone: the impedance converter (usually a FET in a modern capacitor measurement microphone) and Brownian motion of the air in the capsule. The impedance-converter noise includes thermal or "Johnson" noise having a white (flat) frequency spectrum, and "flicker" noise having a "1/f" spectrum (noise voltage is inversely proportional to frequency). The "corner frequency" above which white noise exceeds 1/f noise is often about 100 Hz. Thus, the $1/f^2$ components are attenuated by the A-weighting function used in specifying self noise.

The noise due to Brownian motion of

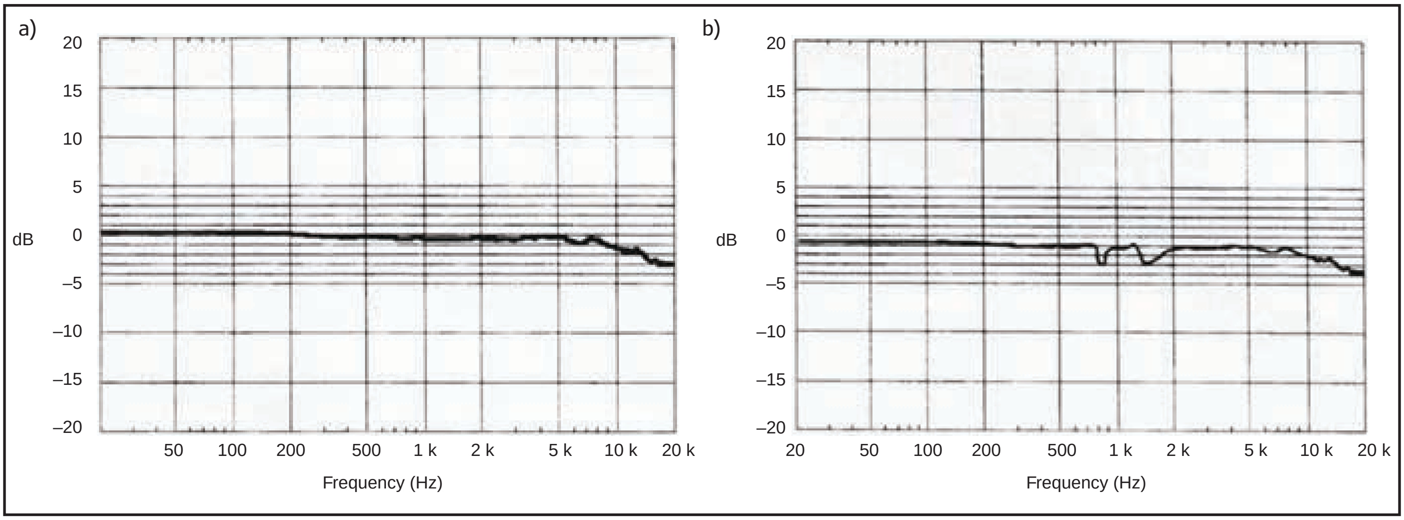


Figure 6: High temperatures can cause permanent damage to electret microphones having film diaphragms.

air in the capsule has a $1/f^2$ spectrum. Doubling the surface area of the microphone diaphragm increases the noise due to Brownian motion by 3 dB, but the microphone's sensitivity to sounds arriving perpendicular to the diaphragm increases 6 dB. The resultant SNR increases by 3 dB for each doubling of the diaphragm surface area, or 6 dB for each doubling of the diaphragm diameter.

A-weighting also understates the effects of $1/f^2$ noise. A capacitor microphone's maximum SPL depends on the linearity of diaphragm displacement vs. instantaneous sound pressure and the distortion characteristics of the impedance converter. Most capacitor measurement microphone capsules have higher maximum SPLs than do the impedance converters, up to 194 dB. Low-cost electret capsules with built-in impedance converters may have 1% distortion limits as low as 90 dB SPL.

Error Sources

Two commonly overlooked error sources in relation to sound level meters are acoustic reflections and temperature abuse. **Photo 8a** shows a multipurpose instrument incorporating a sound-level meter. **Figure 4** shows the deviation from accurate response across the frequency range resulting from acoustic reflections from the case of the instrument. (When the same instrument and the same microphone were tested with the microphone removed from the instrument and connected by a cable, the response was quite flat.) Please note that this instrument is advertised as Type 1.

Photo 8b shows an instrument designed to reduce the effect of reflections from the instrument. However, for truly accurate measurements, it is always best to remove the test microphone from the instrument by several feet to more completely eliminate reflections.

Figure 5 shows the response variations resulting from the use of a microphone clip to hold the test microphone, with no absorptive material used to soak up the reflections. While this amount of error would not significantly affect the accuracy of wideband, octave-band, or even 1/3-octave-band measurements, it is significant in more precise measurements.

The majority of test microphones now used in sound level meters are back electret capacitor microphones with metalized polyester diaphragms. These diaphragms are subject to damage—sometimes quite subtle—as a result of overheating. In a test in which the microphone capsule was subjected to 50°C for eight hours (comparable to the trunk of a car on a parking lot in the summer), the small response anomalies are shown in **Figure 6**. (The curve shown in **Figure 6a** is the original response. The curve shown in **Figure 6b** shows the anomalies caused by heating.) The likely cause of these anomalies was either minor localized stretching of the polyester membrane resulting in uneven tensioning or glue failure in the thin protective nonwoven fabric covering mounted in front of the diaphragm. This is a permanent effect, which remained after the temperature returned to normal. The effect occurred when the microphone was not in use.

Author's Note: Several publications list measurement microphone manufacturers including the Loudspeaker Industry Sourcebook, *published by Segment, LLC. It is available through www.cc-webshop.com.*

Locomotive Audio founder Eric Strouth hand-builds one of his 14B mono tube compressors.

Blending Old-Style Techniques with New Technology

An Interview with Eric Strouth of Locomotive Audio

By
Shannon Becker
(United States)

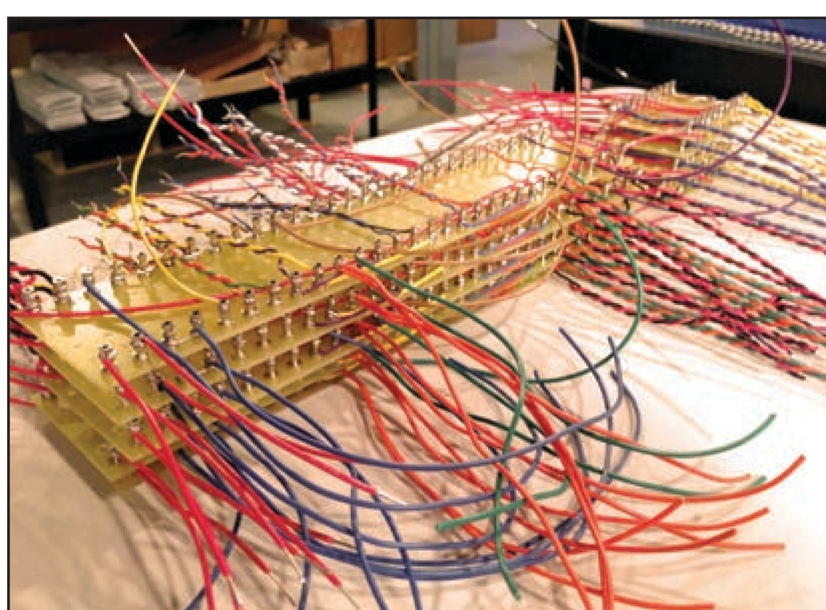
SHANNON BECKER: Tell us about your company Locomotive Audio (www.locomotiveaudio.com).

ERIC STROUTH: I started Locomotive Audio really just as a hobby. But, after I realized I could blend several of my favorite things (audio recording, product design, and building electronics) to start a new career, I decided to quit my day job as a graphic designer and go all-in!

It was a pretty stressful time, but I knew that in the end, I would be able to control my own destiny and nothing seemed more exciting than to be my own boss! I've already learned a lot about myself since starting the company. Learning to push myself and be excited about what I am doing daily is something I was really looking for. And for me, being able to channel several of my talents into one vision has been a real joy.

SHANNON: Describe the handmade analog audio recording equipment that your company designs?

ERIC: One of the main things that excites me about manufacturing audio recording equipment is the fact that I can do it how I want! After building an LA-2A style compressor, which is hand-wired and makes use of the turret board technique, I realized that it offers such flexibility. I could change my designs at moment's notice to fix little issues. And they are a modder's paradise! With a circuit board, you are pretty much wed to the design after it arrives at your facility. Not to mention, turret boards are much more reliable and look so cool when loaded up with all the components! At Locomotive, we plan to use this approach for all of our products. Although the build time is much longer than it is with a standard circuit board, we feel it is a necessary component of who we are and want users to know that somebody actually



All Locomotive Audio designs are hand-wired using a turret board technique.

The dual-channel 286A, shown in front (a) and rear (b) views, has a single-ended Class-A tube signal path. Inspired by the vintage Ampex 601 vacuum tube tape recorder, the 286A features two tubes per channel—one EF806 and one 12AY7 (c).

sat with that piece of equipment for many hours and it became their "baby!"

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

ERIC: We just debuted the Model 286A dual-tube microphone preamplifier and direct input (DI) and the Model 14B tube compressor. Both are built like tanks and use some pretty high-quality components.

The dual-channel 286A preamp offers a huge sonic palette for engineers and producers to explore. Its single-ended Class-A tube signal path is what makes this preamplifier special. The 286A features two tubes per channel—one EF806 and one 12AY7. The 286A can play nice by acting clean and natural with gentle even-order harmonics for a smooth, sweet touch. Or, you can get more aggressive with the "Gain" knob and drive the first 12AY7 stage for a warm and colored thickening effect. You can also go all-out and slam the final 12AY7 stage and output transformer through the "Level" knob for a fully-driven, saturated tube amp sound! Clean and natural to overdriven and fuzzy warm, it's your choice!

Our single-channel 14B compressor and limiter is based on a very old, but still exciting concept of rebiasing the input tube with a side-chain control voltage. The 14B mixes ideas based on a few of our favorite compressors such as the UA 176, the Gates Sta-Level, and the Collins 26U. Its modern features allow it to achieve a unique sound in its own right. Starting with the semi-remote cutoff 6BC8 tube, a 12AY7 then drives two powerful push-pull 6V6 output tubes. The 14B is quite a versatile unit, with the ability to run soft and gentle or hard and aggressive. The side-chain low cut control offers even more flexibility, especially when used on the mixbus and linked with another 14B. This unit can be used with all kinds of source material including vocals, drums, guitars, bass, piano, and horns. You will notice an enhancement on any material run through this unit, even when not compressing.

SHANNON: Were there any challenges involved with the product design?

ERIC: The main hurdle was designing a chassis that could easily accommodate the turret board approach and wouldn't hinder the build process, such as soldering the wires from the turret board to the components that are already loaded on the chassis. Having a graphic design background and being very technically minded with measurements and fabrication, I designed the chassis in Adobe Illustrator and exported my drawings to an AutoCad format for the metal fabricators to use. I got pretty close, but of course when you are dealing with several pieces of metal, each with their own tolerance, holes that should match up will need to be bumped around a bit. So after a couple iterations of the prototype, we got it right! I couldn't be happier with the overall look and efficiency the chassis gives the build.

SHANNON: Why did you choose the Kickstarter route to fund it?

ERIC: My wife Jeni had mentioned it, and at first I dismissed the idea because I thought Kickstarter was only for little projects like, "help me fund a CD" or "give me some money for a short film." But after taking a look and seeing some pretty amazing high-dollar campaigns, I realized she was right. I figured people would feel safer going there and giving a complete stranger their hard-earned money than

The single-channel 14B compressor and limiter, shown in front (a) and rear (b) views, is based on an old concept of rebiasing the input tube with a sidechain control voltage. The 14B offers a semi-remote cutoff 6BC8 tube (c) and a 12AY7 drives two powerful push-pull 6V6 output tubes (d).

just pushing a PayPal button on my website.

SHANNON: Was this your first Kickstarter experience?

ERIC: Yes, and overall, I would say it was great! I did learn that Kickstarter, in general, does much better for lower-priced items that are more accessible to the general population of people. For instance, a $500 3-D printer on Kickstarter is much more exciting to a lot of people than a $2,000 audio compressor. So most, if not, all of the "backers" came from my own advertising on audio forums and Facebook. If anything, as I said before, the credibility of Kickstarter is what may have made some people decide to actually put up their money.

SHANNON: Would you recommend Kickstarter to other entrepreneurs?

ERIC: Absolutely! I would definitely recommend it as a great way to get your product quickly seen by many people. Your product has to be just right for it, but the upside is that things are pretty well organized and the funds are direct deposited into your checking account. So, it is a pretty streamlined process. Some things to keep in mind are that Kickstarter takes 5% and the actual payment processor, Amazon, takes 3–5%. So, the creator needs to build that into the price. The way I see it though, is that the exposure is worth the fees and without Kickstarter, the creator may not see any sales!

SHANNON: Has your product started shipping? What kind of response has it received from your backers?

ERIC: Yes, in fact, we recently sent our first package containing two 14B compressors to Sweden! I've also sent a demo unit to a local St. Louis studio (Firebrand Recording), which was very well received. Firebrand Recording will be getting its own unit this week!

SHANNON: How long have you been designing audio electronics? When did you become interested?

ERIC: I have been tinkering since I was a child, but really designing for about four years now. As with most people from my generation, I started "in the box" and got fed up with using a computer screen and mouse to control my audio recordings. I hated the fact that I couldn't fit everything on my screen and that I couldn't just grab the damn knob with my fingers and turn it! After a trip to Memphis, TN, and seeing all the beautiful old analog equipment in Sun Studio, I was hooked. I came home and purchased a reel-to-reel tape recorder and started learning. I became a little obsessed actually! And here I am, designing old-school tube equipment. Although I use a tape recorder instead of a DAW now, I honestly think a hybrid approach with new and old technology is the way to go.

SHANNON: What's next for Locomotive Audio?

ERIC: Well, the most exciting news for us is that we are attending the Audio Engineering Society (AES) International Convention (October 9–12)! We hope to make some pretty awesome connections and sell a few pieces of gear! I'm so excited!

SHANNON: Do you have any advice for *audioXpress* readers who are thinking of building their own sound systems?

ERIC: Although the high voltages can be very dangerous and a little soldering experience is a must, I would definitely recommend doing a project like an LA-2A clone. Powering up a high-quality piece of equipment is very gratifying after soldering that last wire!

Build Your Own Audio Mixer (Part 1)

Start with a Basic Mixer

Although we live in the digital era and we are surrounded by sophisticated computerized gadgets, there is still a fascination with putting together a few resistors, capacitors, and semiconductors to make a simple electronic circuit. In this article, we discuss how to design and build an analog audio mixer using common electronic components. It is really surprising how a handful of components can perform a useful function. And, the satisfaction of building it yourself is incalculable.

By
George Adamidis
(Greece)

An audio mixer, which is also called a mixing console, is an electronic device that combines and modifies audio signals. The modified audio signals produce combined output signals. Audio mixers can be analog or digital. Digital mixing consoles use DSP concepts and analog mixers are usually based on op-amp electronic circuits. Although DSP is the current trend, analog circuits are still used due to their simplicity and low cost.

To illustrate the approach, we will describe our recent efforts to develop a modular design (see **Photo 1**). Modular design is a form of splitting an object into smaller portions so that they can be joined together to form one complete system. A modular audio mixer is formed by assembling some main modules that can vary in number and/or disposition to suit individual needs. In a modular mixing console, the designer can decide how many inputs should be provided. The input audio signals could be anything from microphones, CD players, or PC sound cards to any other analog audio source.

Photo 1: This is my homemade, modular audio mixing console.

The Basic Mixer

Our basic mixer will be able to combine signals from different signal sources, change the volume of each input channel, and change the output's overall volume. **Figure 1** shows summing circuit, which is actually the heart of the mixing console. This circuit is also known as the summing amplifier. It consists of an op-amp, n input resistors (R1, R2 … R_n) and a feedback resistor (R_f). A summing amplifier sums several (weighted) voltages. Its output voltage V_o is given by the formula: $V_o = -(A_1 \times V_1 + A_2 \times V_2 + A_3 \times V_3 + \ldots + A_N \times V_N)$. A*x* is the voltage gain for the xth input and it is equal to R_F/R_X. So:

$$V_{OUT} = -Rf\left(\frac{V_1}{R_1} + \frac{V_2}{R_2} + \ldots + \frac{V_N}{R_N}\right)$$

When all input resistors (R1, R2, ... R_N) and the feedback resistor R_F, have the same value, then the voltage gain for each input channel becomes equal to the unity. The formula becomes: $V_O = -(V_1 + V_2 + V_3 + \ldots + V_N)$. The minus sign indicates that the summing output is inverted or otherwise phase shifted by 180°. However, phase shift has no audible effect.

If the value of the feedback resistor, R_F, becomes greater than the value of any input resistor, R_x, then the gain for channel *x*, will be greater than unity (and equal to R_F/R_x).

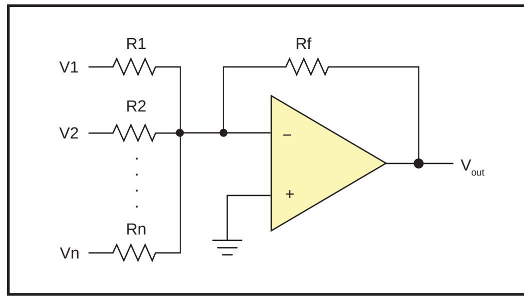


Figure 1: The summing amplifier is itself a mixer.

The summing circuit is actually an adder, and a basic mixer is really nothing more than an adder too, so the summing circuit is itself a mixer. However, the summing circuit doesn't have any elements for adjusting sound volume (voltage levels). An audio mixer usually does, so adding them should be the next challenge (see **Photo 2**).

To make our mixer a little more sophisticated than a simple adder, let's add more circuits using the block diagram shown in **Figure 2**. We will add a matching circuit at every input to ensure that no signal source is unduly overloaded. Every matching circuit will serve as a preamplifier and a volume level adjuster for each input channel. Moving forward we will call it the "input module." The outputs of all the input modules will be combined in a summing amplifier. We will also add a potentiometer to our summing amplifier, which will be the "master" volume adjuster (used to adjust the output volume).

Line Input Modules

Most audio signals used to transmit analog sound between audio components (e.g., CD and DVD players, TVs, audio amplifiers, and mixing consoles) are line-level signals. There are also weaker audio signals (e.g., those from microphones and instrument pickups) and stronger signals (e.g., those used to drive headphones and loudspeakers). This project uses a line-level input module. Later, we will design additional input modules for weaker audio signals.

To design a line-type input module, we must consider two facts. Cables between line output and line input are generally short compared to the audio signal wavelength in the cable. So, no transmission line effects or impedance matching is required. However, line-level circuits use the impedance bridging principle, in which a low-impedance output drives a high-impedance input. A typical line-out connection has an output impedance from 100 to 600 Ω. Line inputs present a much higher impedance, typically 10 kΩ or more. The two impedances form a voltage divider (see **Figure 3**). The shunt element (R_{IN}) has a large resistor value relative to the resistor value of the series element (R_{OUT}), which ensures that little of the signal is shunted to ground and that current requirements are minimized. Most of the voltage asserted by the output appears across the input impedance (R_{IN}) and almost none of the voltage is dropped across the output impedance (R_{OUT}).

$$V_{IN} = V_{OUT} \times \left(\frac{R_{IN}}{R_{IN} + R_{OUT}} \right)$$

The approximate nominal voltage level of a line-type signal is about 320 mVRMS or 1.2 VRMS for consumer and professional audio equipment, respectively. An appropriate input module circuit that conforms with the above criteria is shown in **Figure 4**.

The circuit shown in **Figure 4** is actually an inverting amplifier. An inverting amplifier is a common circuit, based on an op-amp. Its basic function is to scale (or amplify) and invert the input signal. The inversion is equivalent to a phase shift and has no audible effect.

Referring to the left-hand channel (the right-hand channel is, of course, identical) and as long as the op-amp gain is very large, the amplifier gain is determined by the external resistors (the feedback resistors R2A and R3 and the input resistor R1). The voltage gain is equal to the ratio of (R2A + R3)/R1. Moreover, the circuit's input impedance (R_{IN}) is approximately equal to R1 because the operational amplifier's inverting input is a virtual ground.

We chose R1 to be equal to 10 kΩ to comply

Photo 2: My homemade, simple stereo-mixer has one microphone input, four line-inputs, a voltage unit (VU) meter, bass and treble tone controls, and a headphone monitor.

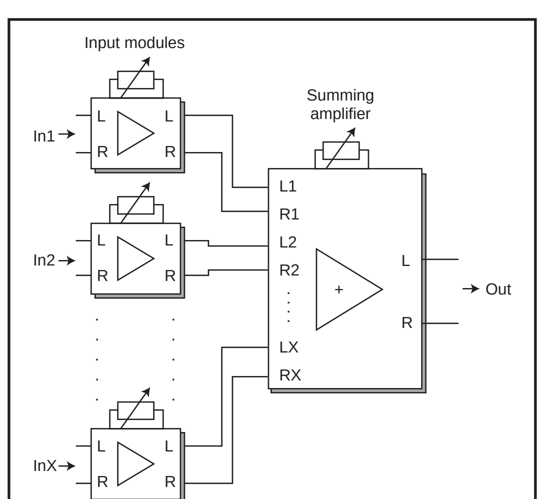


Figure 2: The block diagram shows a simple stereo-mixer.

About the Author

George Adamidis is a Physicist and an Electronics Engineer. He has a degree in Physics (2000) and a MSc in Electronic Physics (2002) from Aristotle University in Thessaloniki, Greece. He currently teaches high school Physics and Electronics and he is an Assistant Researcher at the Centre of Technological Research of Crete. His research interests include embedded systems, RF technology, smart antennas, antenna design, and electromagnetic compatibility.

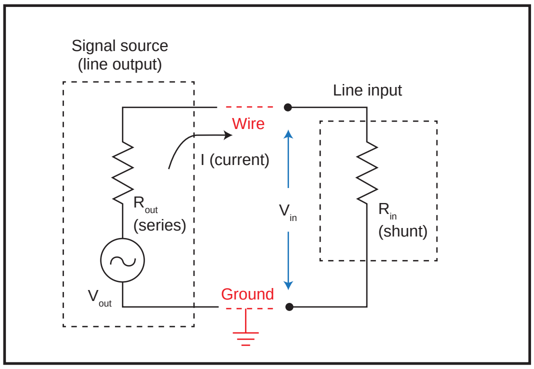


Figure 3: The line-output and line-input impedances form a voltage divider.

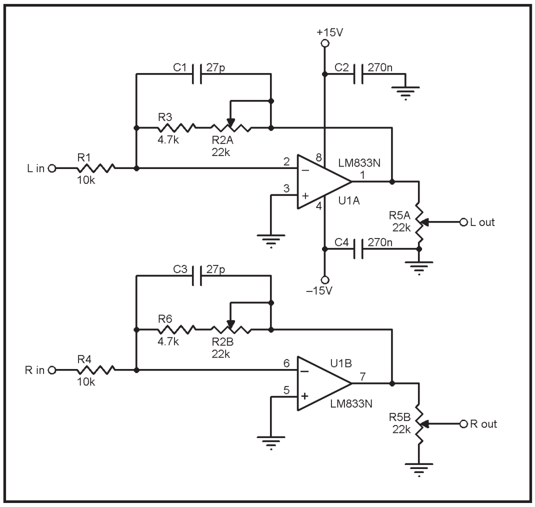


Figure 4: A line-type input module has identical left and right channels. R2 and R5 are logarithmic-type stereo sliding-potentiometers.

with the criterion of the minimum input impedance required on a line-type input. We intentionally used a minimum value for R4 to avoid thermal noise (we will discuss this later). R3 is equal to 4.7 kΩ, and R2A can be varied from 0 to 22 kΩ, so that the voltage amplification (R2A + R3)/R1 can be varied from about 0.5 to 2.6 (from −6.5 to 8.5 dB). This way, R2 acts as a gain adjuster, and the nominal output level can be adjusted from 160 to 830 mVRMS or from 0.6 to 3.12 VRMS, for consumer and professional audio equipment, respectively.

C1 is used for high-frequency noise filtering and for preventing oscillations. Together with R2 and R3, it forms a low-pass filter. The filter's cut-off frequency is above the upper limit of the audible range (20 Hz to 20 kHz), and varies inversely proportional to the change in value of R2 (see **Figure 5**). The R5 potentiometer implements an adjustable voltage divider and acts as the volume level adjuster.

The circuit shown in **Figure 4** is a good example of a line-input module in a mixer. Of course, many other circuits can also used for the same purpose. Any voltage preamplifier that has adequate input impedance (equal or higher than 10 kΩ) and a volume level adjuster can be used at the mixer's input stages. Additionally, the input stages must not

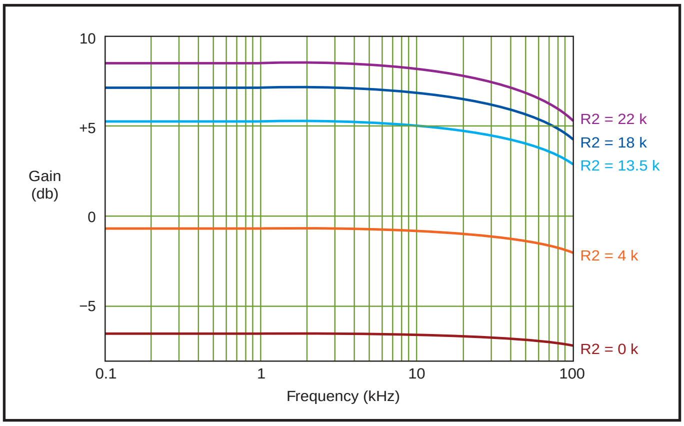


Figure 5: The line-input module frequency response varies for different wiper positions of R2.

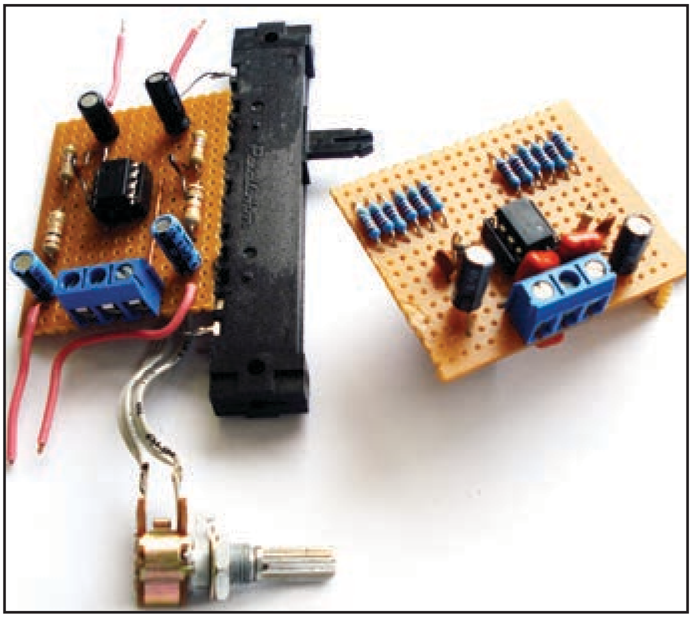


Photo 3: The line-input module (left) and the summing amplifier module (right) are built on prototyping matrix boards.

produce any noise or distortion. They must have flat frequency response from 20 Hz to 20 kHz (which is the audible range) and they must also be stable. Almost everything depends on the op-amp choice, the filtering, and using small resistor values.

In the prototype, we use the classic LM833 dual op-amp that has been designed with particular emphasis on performance in audio systems (see **Photo 3**). We also use as small as possible resistor values to avoid thermal noise. It is well known that any resistor produces some thermal noise voltage, which is promotional to the resistor's value. So, minimizing resistor values is essential for minimizing thermal noise level. Unfortunately, our choices for a single-stage circuit are limited when we need the input impedance to be at least 10 kΩ.

Thermal noise power is also proportional to the total bandwidth. Keeping the total bandwidth as low as possible is essential for reducing noise level. For this purpose, low-pass-filtering is used in every audio mixer to reject spectral content above 20 kHz (the upper limit of the audible range). Although filtering improves the overall SNR, it also affects the preamplifier's frequency response. So, it is somehow difficult to simultaneously ensure good filtering and flat frequency response (through the entire 20-Hz-to-20-kHz audio range). In our circuit, we use rather simple filtering and focus on achieving flat response (see **Figure 5**).

In our circuit, we use DC-coupling for flat response even at very low frequencies. DC-coupling is an advantage in most cases, but it could eventually become a problem if the input source has any DC-leakage (offset). Any DC at the input will be amplified, will pass at the output, and it could saturate the next stages. So, in the specific case of having a DC at the input, we recommend the use of a DC-blocking capacitor (in series with R1) to prevent the next stages from saturation. Adding a DC-blocking capacitor will limit the flatness at very low frequencies, and some low-frequency content may be attenuated, unless a large enough value is used. (The recommended value is about 10 μF or more.)

The Summing Amplifier

Figure 6 shows the summing amplifier circuit for our mixing console. Referring to the left-hand channel (the right-hand channel is identical), R7A is the feedback resistor and R1L, R2L,....RXL are the input resistors. The feedback resistor is a 4.7-kΩ stereo potentiometer (R7) that enables the output level to be matched to the sensitivity of the unit to which the mixer is connected (i.e., R7 acts as the master-level adjuster).

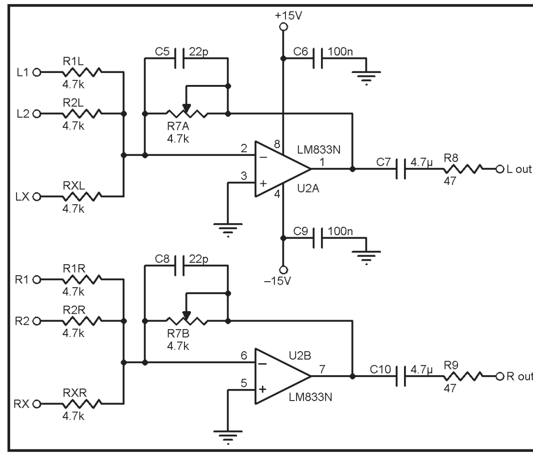


Figure 6: The summing amplifier circuit has identical left and right channels. R7 is a logarithmic-type stereo sliding-potentiometer.

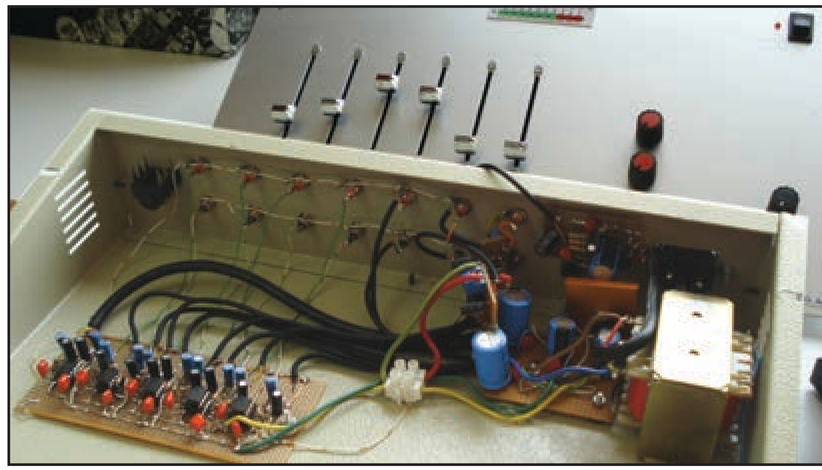


Photo 4: The mixer uses a modular-design concept. However, building some modules on the same board minimizes the number of cable connections required.

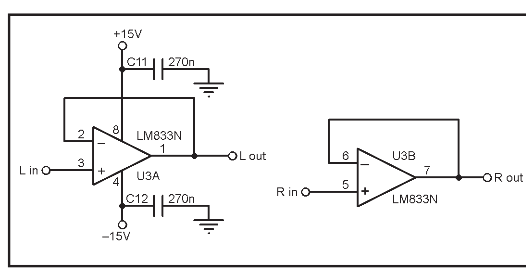


Figure 7: The voltage-follower circuit is actually a unity-gain voltage amplifier, which provides maximum isolation between its input and output.

Photo 5: I tested the mixer at the lab.

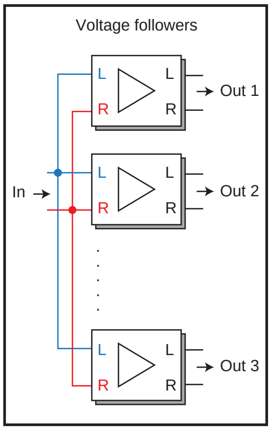


Figure 8: Use one or more voltage followers to add outputs.

The input resistors (R1L–RXL or R1R–RXR) have the same value. They are all equal to 4.7 kΩ, so that the gain—which is equal to R7/R1—can be varied between 0 and 1 (negative infinity to 0 dB). C8 performs basic filtering and is used to attenuate high-frequency signals to reduce noise level. There is also a series RC network at the output. The purpose of this network is to prevent any DC-offset from appearing at the mixer's output and also obviate any oscillation tendencies caused by a capacitive load (e.g., long screened cables). Using the modular-design concept, the summing amplifier module can be easily assembled on a prototyping matrix-board (see **Photo 4**).

Additional Outputs

Besides the main output, most mixers usually have additional outputs. The main output is usually connected on a final amplifier and other outputs are typically intended to be used as signal sources for monitoring or recording equipment.

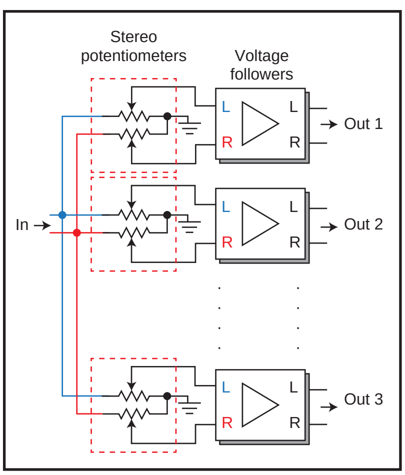


Figure 9: Using potentiometers at the inputs of the voltage followers enables independent volume adjustment for each output.

Adding outputs is rather simple. It is based on the concept of using one or more voltage followers. A voltage follower is a basic op-amp circuit. It is called a "follower" due to its ability to "follow" its input without any loss. It is actually a unity-gain voltage amplifier that simultaneously has very high input impedance and a very low output impedance, thus providing maximum isolation between its input and its output. **Figure 7** shows the voltage follower circuit.

By adding one or more voltage followers, with their inputs connected on the summing amplifier's output, we will be able to build a mixer with as many outputs as we want (see **Photo 5**). Due to the voltage-follower circuit's high input impedance, we can connect as many followers as we need on the summing amplifier's output, without overloading it (see **Figure 8**). All outputs from the followers will provide the same audio signal, but they will be totally isolated one from another.

If we want to independently adjust the signal level in each output, we can use a potentiometer at the input of every voltage follower. The wiper of each potentiometer should be connected to the input of each voltage follower (and the other two pins of each potentiometer should be connected to the summing amplifier's output and the ground, respectively). **Figure 9** shows the arrangement. These potentiometers will act as volume adjusters for each output, and one of them can serve as the "master" volume adjuster. In this arrangement, R7 is not needed as a volume adjuster, and it can be replaced with a fixed-value resistor (or a trimmer resistor).

Using potentiometers at the inputs of the voltage followers reduces their input impedances. The total impedance "seen" by the summing amplifier will also be reduced. It will be about R_{POT}/N (where R_{POT} is the resistor value of each potentiometer, and N is the total number of potentiometers). This will somehow limit the maximum number of voltage followers that can be connected at the main output. However by using potentiometers of about 47 kΩ each, adding up to four or five outputs will not be a problem.

In the second part of our article series, we will create a more complex structure by adding some circuits for audio equalization. We will add circuits for attenuating or boosting a range of frequencies (e.g., bass, midrange, and treble) on each audio channel and use a general graphic equalizer to perform general equalization control at the output. We will also include a volume unit (VU) meter and a headphone monitor.

Frequently Asked Questions

What if I wish to modify the gain of the input modules?

Referring to **Figure 4**, the voltage gain in each input module is equal to (R2A + R3)/R1 or equal to 20 × LOG[(R2A + R3)/R1] in decibels. Increasing, or decreasing the value of R2A or R3 will increase, or decrease the gain.

What is a DC-blocking capacitor?

A DC-blocking capacitor is a capacitor used for direct current (DC) blocking at a preamplifier's input. It is connected in series (the current passes through it). You may need to use DC-blocking capacitors in your audio mixer if you face any problems due to DC-offsets.

The reference designs in **Figure 4**, **Figure 6**, and **Figure 7** use DC-coupling in their inputs for achieving flat response even at very low frequencies, and they do not have DC-blocking capacitors. This is an advantage as far as the input sources do not have DC-leakages (offset). However, if there is any DC present at an input, it will be amplified and will pass at the output. Most DC connected preamps will simply saturate. For example, a gain of only six times (16 dB close enough) will convert a 2.5 VDC-leakage into the full 15-V maximum output from the preamplifier. So, using DC-coupling capacitors is essential in some applications. A DC-blocking capacitor may limit somehow the gain at very low frequencies, unless a large enough value is used (the recommended value for the reference designs is about 10 µF or more).

Do I need to use low tolerance components?

Using high quality and low-tolerance components is essential in a stereo-mixer. Otherwise, voltage gain or frequency response may not be the same on the right and the left audio channels. In the prototypes, we use 1% tolerance resistors and 5% tolerance capacitors.


DAVOS? THE ILLUMINATI? BILLIONAIRE SUMMITS?
THEY'VE GOT NOTHING ON THE POWER OF THIS PLACE.
When it comes to gatherings of powerful people who shape business and innovation, CES is in truly elite company. Best of all, you don't have to be incredibly wealthy to be here. Then again, the contacts you make here might just get you in that club. Register at CESweb.org.
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You Can DIY!

The TERESA Amplifier

Photo 1: The TERESA amplifier is a simple and cascadable Class-AB design.

TERESA is an acronym for the manufacturers Texas Instruments (TE) and Renesas (RESA) because these companies provided the parts for the core of this amplifier. In this article, I used the latest Audio-MOSFETs from Renesas and a relatively new high-end drive component from National Semiconductor (now part of Texas Instruments). The result is worth seeing and hearing! This simple and cascadable Class-AB design with professional audio qualities does not require any special experience in detailed analog circuit technology to build.

By
Harald Frank
(Germany)

The initial idea for this amplifier came from my own wish, as an engineer, to build an amplifier at least once in my life. But if you have spent all your working life in the world of microcontrollers and their programming, you are no better prepared for such an adventure than any other analog layman. In total, I spent about two years researching, building, and improving my TERESA amplifier (see **Photo 1**).

For my first attempt, I used classic Hitachi MOSFETs—the 2SK1058 (N-Channel) and the 2SJ162 (P-Channel). I also used a National Semiconductor (now Texas Instruments) LM4702 high-end stereo drive part. This worked quite well. You can find the circuit in the examplary application notes for the LM4702 (AN1490, AN1645). Jack Walton's article "High-End 120-W MOSFET IC Driven Amp," (*audioXpress,* July 2007) also provides information on the LM4702 (see Resources).

During my second attempt, I used techniques that I learned from the first build, coupled with my natural curiosity. Apparently, no one has combined Renasas's relatively unknown audio MOSFETs 2SK2221/2SJ352 with Texas Instruments's LME49811 premium mono drive part (see **Figure 1**). I also found that the oscillation tendencies often attributed to MOSFET amplifiers seem significantly less with this combination. If you are fond of loud music, I should mention that the new driver part delivers about double the gate drive power to the MOSFET output stage (per channel). This is less important for an amplifier with one or two pairs of output MOSFETs but if you use three or more pairs, you have enough gate drive power to easily exceed 500-W output power.

The new MOSFETs can also deliver slightly higher output currents of 8 A vs. 7 A for the previous types. However, TERESA's quality should not suffer from the power specifications. The LME49811, with its 0.00035% distortion specification, is well suited for this application and will help the amplifier reach professional audio quality (i.e., the amplifier's relevant parameters). High-quality connectors, capacitors, trimmers, resistors, and wiring also make this a successful project. Several output pairs can be cascaded. The choice depends on the amplifier's expected use.

The basic amplifier with a single output pair configuration will have about 10-to-15-W idle dissipation, and can, with ±50 V supplies, deliver about 200-W sine wave power in 8 Ω (see **Photo 2**).

The goal is to produce an affordable high-quality project with the simplest possible design. With only a few exceptions, all the resistors are 1%, 0.25-W metal film types. In a few places, silver-mica capacitors have been used. These have good properties for audio applications. All the silver-mica capacitors are 30 pF, which will make sourcing easier. Also, all the capacitors on the LME49811 should be placed as close as possible to the IC pins.

The Concept

Using a coupling capacitor for the input signal to the LME49811 (C1) is safe, but it might also slightly lower the performance. However, in protecting the input from DC, such a capacitor significantly improves reliability. This capacitor could be left out if you are sure the source provides a DC-free signal. But, if you have to use one, it should be high quality. I decided to use a 10-µF upper-mid-class Mundorf capacitor. These capacitors are specially manufactured for crossover filter and coupling capacitor applications.

In the process, I learned a lot about capacitors. For example, they must be “broken in” to reach and maintain their full sound spectrum. So, use two identical capacitors for the two amplifier channels in a stereo setup.

The principle of the TERESA amplifier is simple and easy to understand. If necessary, the input signal can be attenuated with a high-quality volume control-type potentiometer. The signal is routed to the LME49811 +IN (noninverting) input via the coupling capacitor C1 and the resistor R3. The feedback is connected from the output via R2 to the –IN (inverting) input. The gain is calculated with the equation N = R2/R1. This value should be adapted to the expected input signal level and the power supply voltage to avoid clipping. For this amplifier, the gain should be between 20 and 35. Resistors R3 and R4 should be similarly dimensioned.

The LME49811 drives the gates of the MOSFETs, configured as source followers, from its SINK- and SOURCE- outputs, through gate stopper resistors. Any unsymmetries in the MOSFETs of a single pair will be compensated for by the feedback. Therefore, it is not necessary to match them. An RC-snubber is used at the output to suppress high-frequency signals. For bandwidth measurements, this snubber should be temporarily disconnected because the resistor can become very hot with prolonged high-frequency high-level measurements.

Power Supply and Power Supply Voltages

The ground (GND) connection should be wired to a star point with large-gauge wire (see **Photo 3**). The power supply’s dimensioning can vary with the specific application. I recommend using a panel-mounted mains entry block consisting of a DP switch, a fuse holder, and a mains filter. This immediately solves several mechanical and electrical problems. The toroid mains transformer (which has less electromagnetic stray field) ideally should have extended secondary windings. In my amplifier, it is a 280 VA, 2 × 38 V and 2 × 43 V. You can find these and similar reasonably priced transformers on the Internet.

There are four main voltages, V+, V–, V++ and V–– (see **Figure 2**). Voltages V++ and V–– are for LME49811’s symmetrical supply. They should be about 10 V above V+ and V– for the output MOSFETs. Otherwise, you lose 10-V output level because the MOSFET gates need to be driven 10 V above their source voltage. Voltages V–– and V++ should, according to the LME49811’s specifications, range between 20 and 100 V. The MOSFET supply voltages V– and V+ should be 10 V less than that. This is a relatively large range that can be set to personal preference or a specific application. For V++ and V––, I recommend a reservoir capacitor of approximately 10,000 µF. For V+ and V–, I recommend a reservoir capacitor of approximately 30,000 to 40,000 µF. The LME49811 suppresses voltage variations on the supply lines rather well (see **Photo 4**). Therefore,

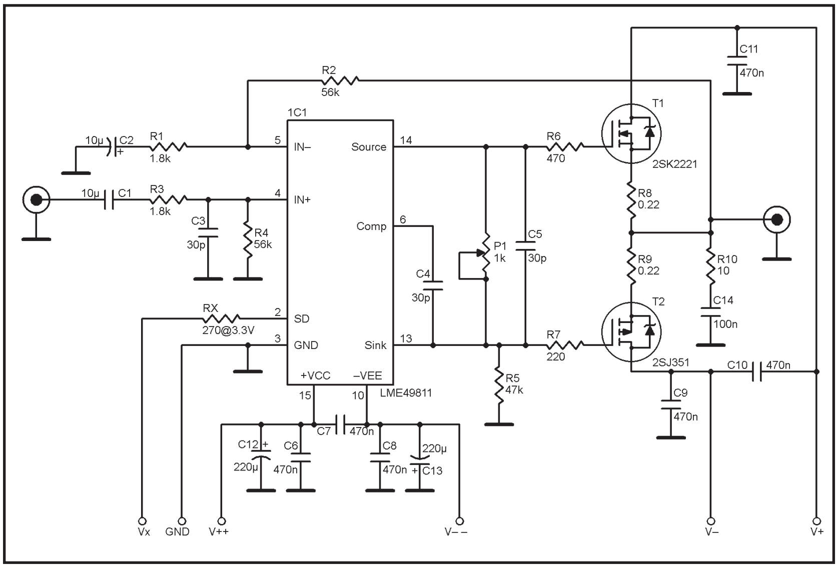


Figure 1: This design combines Renasas’s 2SK2221/2SJ352 MOSFETs with Texas Instruments’s LME49811 premium mono drive part.

Photo 2: The completed amplifier is shown prior to buttoning up the enclosure.

Photo 3: The TERESA amplifier can drive three pairs of transistors.

you could connect V– to V–– and V+ to V++ if that would make your power supply simpler (but it would have a lower maximum output level).

Anti-Plop Provision

I have put an extra diode (D3) after the rectifier for the V+ voltage to create a fast decaying voltage. This feeds a low-value reservoir capacitor (C17) and is limited with a Zener diode. This Zener diode is only required if the voltage will be above 50 V because the following regulator (IC2) can accept input levels up to 50 V. To ensure the Zener diode correctly operates and the reservoir capacitor discharges relatively quickly, a discharge load is required. The discharge load is usefully combined with the Power-ON LED. About 5 to 10 mA will be sufficient here.

For the regulator, I have selected an S-1142B33 from Seiko Instruments, which can withstand a 50 V_{IN}, and delivers 3.3 V at up to 200 mA. This regulator has a Schmitt-trigger ON/OFF input (we need the –B part here, which has a positive going ON/OFF switching logic). A simple trimmer at this input provides a well-defined regulator switch-off point after the mains has been switched off. The regulator's output is connected through a carefully dimensioned resistor to the LME49811's shutdown-input. So, the whole amplifier is cleanly switched on or off through the regulator ON/OFF input voltage, with no switching pops or crackling (see **Photo 5**). It is simple and effective!

Important note: The LME49811 SD-input is quite sensitive and can, when overdriven, damage the chip. It is important to limit the current flowing into this input to between 1 and 2 mA. The SD signal should also have a reasonable rise time and not change levels too slowly. The combinations for the supply voltages, the input currents, and the LME49811's operational status are shown in **Figure 3**.

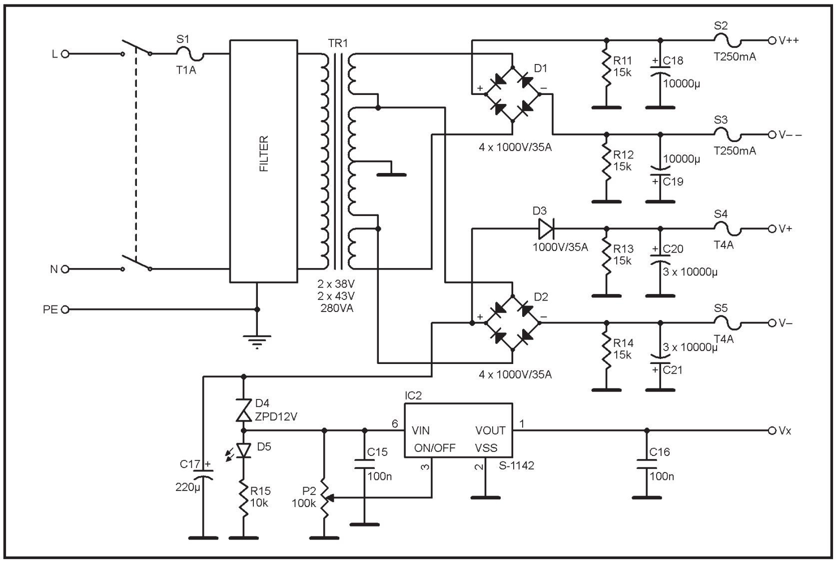
You can calculate the required series resistor or determine it through trial and error. Keep the input current as close as possible to 1.5 mA. The equation for this is: R_{SERIES} = (input voltage – 2.9 V)/1.5 mA. However, when you work with changing voltage levels during the test phase or when you're not sure what the input voltage will be exactly, the circuit and values shown in **Figure 1** and **Figure 2** will provide fail-safe operation.

Bias Adjustment

With this driver part, the bias generation is really simple. It consists of a single trimmer potentiometer between the LME49811's SINK and SOURCE pins, which is shown as P1 in **Figure 1**. Normally, you would use a VBE-multiplier circuit to generate the bias voltage and temperature-compensation to bias the output MOSFETs. However, the lateral structures in the Renesas audio MOSFETs show a change from positive-to-negative temperature coefficient at a specific bias point. The amplifier's operating point should be set at exactly this point, which would be between 80 and 100 mA. At this point, the amplifier will be temperature stable and show no runaway. When the amplifier heats up, it will move into the negative-temperature coefficient range, which will result in a reduction of the bias current and avoid thermal runaway.

Compensation is effected through a 30-pF capacitor between the compensation input and the SINK output. Use a high-quality mica capacitor and solder it as close as possible to the pins. Before switching on the amplifier, the bias trimmer should be set to minimum resistance. The amplifier input should also be shorted and and then the bias should

Figure 2: The TERESA amplifier has four main voltages: V+, V–, V++, and V––.



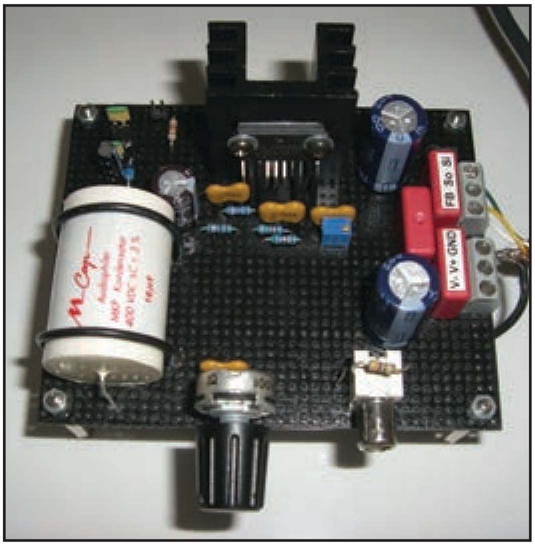


Photo 4: The LME49811 stage is shown without the power stage.

be slowly turned up. You can measure the bias current by measuring the voltage drop across the output emitter resistors (R8, R9, 0.22 Ω). Set the current through the P-Channel MOSFET to 100 mA.

Author's Note: At this point, I recommend that you purchase an inexpensive multimeter if you don't have one. Multimeters are available for approximately $30 and are worth the money. You can then verify supply voltages and correctly set the bias.

Physically Separate Construction

You can also physically split the amplifier at the point where the MOSFET gates are connected to the LME49811's SINK and SOURCE pins. You can easily use some screened cable between the output stage and the LME49811 to enable a decentralized construction layout (see **Photo 6**). In this case, the connection cable's screen should only be grounded at the LME side to avoid any ground loops that could cause hum. You would need three wires inside this cable for SINK, SOURCE, and the feedback connection to R2.

You can further decentralize this if you want to mount the P- and N-Channel output MOSFETs on separate heatsinks. You need to use three separate, screened cables, all with the screen connected on one side to the same ground point. Use them to connect the three sub-assemblies. Of course, you also need to connect the MOSFETs to build the output node but you can do that with a single heavy wire. The various signal cables should be connected to the gate stopper resistors and the driver outputs. This configuration enables you to build rather unconventional amplifier assemblies. There is another good reason to build the circuit as separate assemblies—any oscillation tendencies are also dependent on the distance of the high-impedance input to the low-impedance output and how the signal routings are implemented. So, the more separated these routes are, the better.

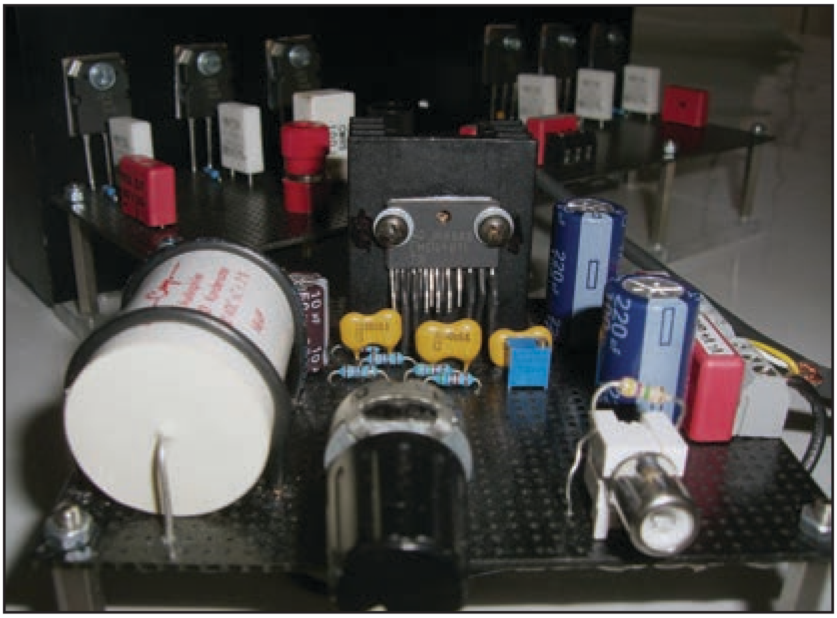
I determined the value for the gate-stopper resistors on the MOSFETs by trial and error. If, with a square-wave signal input, the output signal's rising and leading edges show instabilities around their mid-points, the gate stopper resistor values should be adjusted. If you don't have access to an oscilloscope, use the values in the schematic. With a single MOSFET pair, you could even do without the emitter resistors R8 and R9 (0.22 Ω) although the correct bias adjustment would become more difficult.

Most knowledgeable DIYers recommend Futaba MPC74 5-W resistors for the emitter resistors. When using several parallel output pairs, emitter resistors are required to ensure correct power sharing between pairs. Matching the MOSFETs for a single pair is not very useful because the LME49811 will flatten out any differences in parameters. However, with several parallel pairs, matching between same-polarity (N- or P-Channel) MOSFETs in an amplifier, ensures a better sharing of output power. Finally, a high-frequency damping combination of R10 and C14 is connected across the amplifier output.

Heatsinking

With a 100-mA bias current and a ±50 V MOSFET power supply voltage, there will be about 5 W idle dissipation per output MOSFET (in a single output pair), so adequate heatsinking must be provided. During actual operation, a single MOSFET pair can provide an output power of 150 to 200 W, and this should also determine the required heatsinking. The LME49811 should be mounted on a heatsink, which can be much smaller. The part has an internal temperature protection circuit that should only be activated with insufficient cooling. Both the MOSFETs

Photo 5: The LME49811 stage is shown in the front with the power stage at the back.



About the Author

Harald Frank studied in Duisburg, Germany, and became an Engineer in 1989. He started working as a field application engineer (FAE) for microcontrollers. He now works as an FAE and as the office manager for GYLN in Germany.

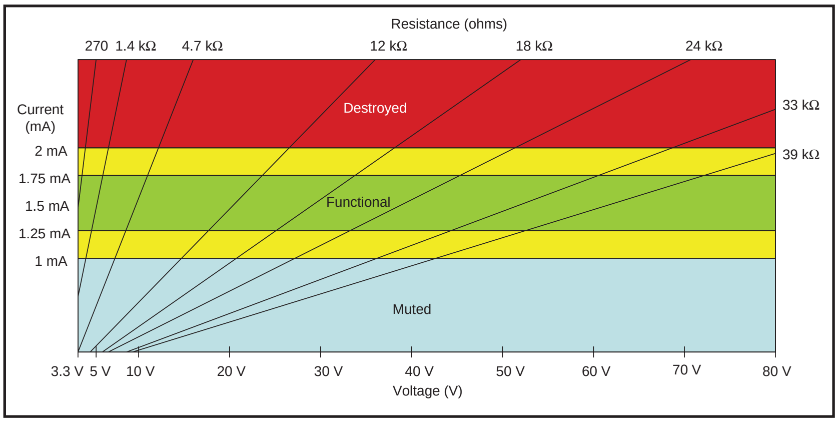


Figure 3: The LME49811 SD input diagram shows the current, voltage, and resistance.

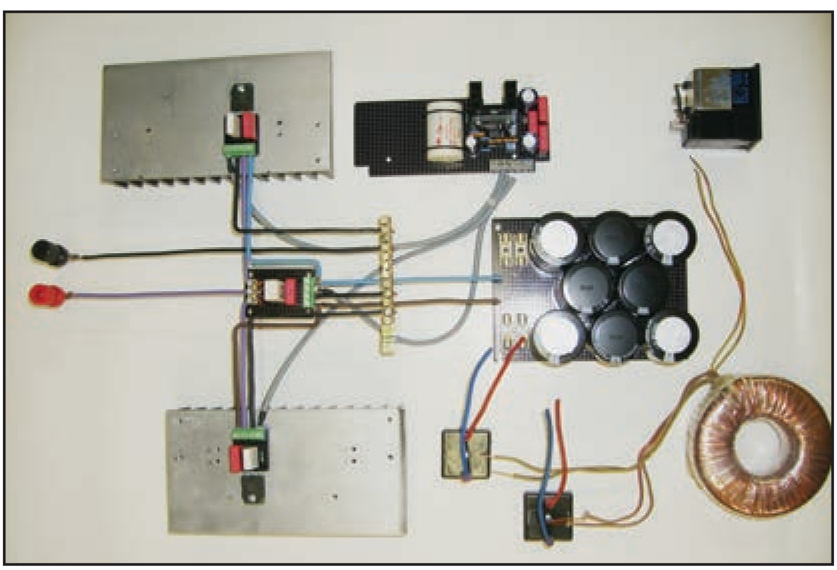


Photo 6: I recommend a decentralized construction, especially for the output transistors.

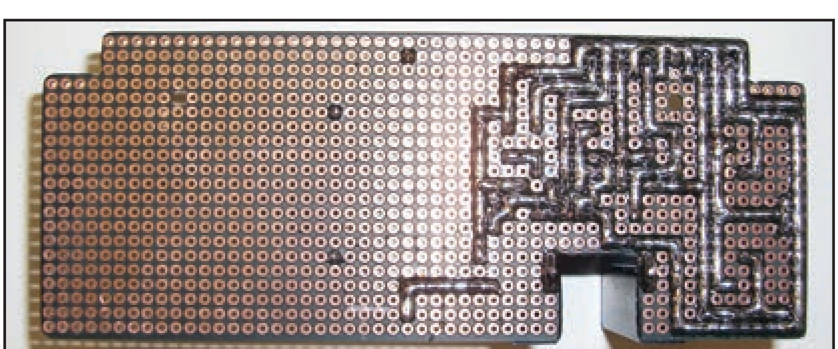


Photo 7: My PCB shows one practical way to design the heart of the amplifier.

and the driver chip have non-isolated metal heat transfer surfaces, which should be isolated from the metal heatsinks with heat-conducting isolation pads.

DC Protection

In many amplifiers, a lot of effort is put into ensuring that no DC can reach the speaker. In the TERESA amplifier, there is only one high-quality input coupling capacitor to ensure no DC enters the amplifier. Actually, DC at the output can only result from defective parts (which is not likely with a newly built amplifier). Destroyed parts, in my experience, are possible when an LME malfunctions or has the wrong voltage. However, good DC protection is worthwhile and I will propose such a system in a follow up article.

Short-Circuit Protection

TERESA has no problems with either a 4- or 8-Ω load. Builders should match the number of required MOSFET pairs, the expected speaker load, the heatsinking, and the power supply parameters so the combination results in a reliable amplifier system. A sufficient protection against short circuiting can be provided by a fuse at the power supply output. With all my tests so far, not a single MOSFET has been destroyed.

Assembly

I recommend a decentralized but classic assembly in a metal enclosure. Bridge rectifiers with a central mounting hole can simplify wiring and cooling. For the star GND point, I used a length of mains earth rail. All cabling is flexible with the largest diameter possible. The heatsinks are from an old amplifier. Each MOSFET has its own heatsink and doesn't really get warm at room listening levels. All the other metal parts for the assembly and the enclosure sides are 2-mm aluminum, which are easily sourced and can be easily worked.

In analog designs, PCBs are seldom perfect on the first try and usually require several iterations. Therefore, PCBs only make sense for this amplifier if they are compact and modular enough to enable a decentralized assembly (see **Photo 7**).

I used proven technology with the well-known matrix experimenter PCBs to find the optimum parts placement. This helped me quickly find a good solution. I painted the top surface of the boards black with lacquer paint before assembly, which gives them a professional look. The traces are built from thick solder layers, which work well after some trials. However, silver wire or anything similar will work as well. The advantage of using this technology is a clear layout that can easily be

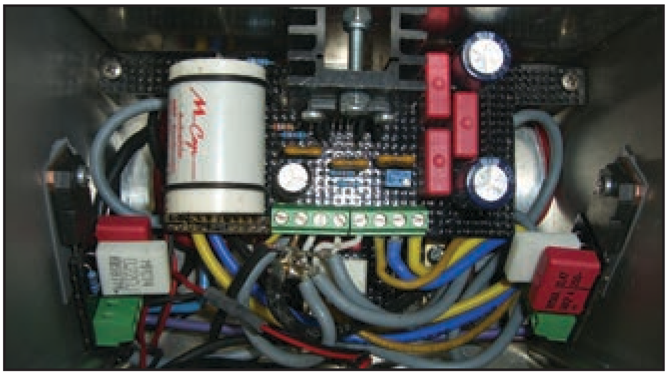


Photo 8: The TERESA amplifier mechanicals and electrical details are in place.

planned. The thick connections are also advantageous, especially for GND connections. Here, too, you should star-wire all GND connections and traces. If you have hum, it often is a result from GND loops (see **Photo 8**).

If the amplifier is built in a closed box its performance can slightly change for better or worse. For example, if you have hum or oscillatory tendencies, it may be possible to connect the open input to GND with a resistor. This resistor value should be as high as possible and as low as necessary. An important point is the connection of GND to the enclosure. It is necessary for the enclosure to have a single GND point and all metal parts connect together. Some amplifiers have an external option to connect the enclosure to the amplifier GND to fix problems. A 1-kΩ resistor from GND to enclosure can often work miracles.

Final Observations

This amplifier circuit is not perfect and surely harbors some significant improvement possibilities. However, it functions well and can provide an excellent starting point for many applications. An amplifier is only as good as its parts. Use good quality parts and you will reap the benefits.

The cascading options with several output MOSFET pairs make it difficult to design an efficient PCB for all the possibilities, and at this point, none is available. Also, I don't have the necessary test equipment to measure the TERESA's main parameters. This could be done by interested builders who have the necessary equipment. I would be very interested in your findings!

If, when listening to the TERESA, you hear sounds that you haven't previously heard, it could be because of the high cut-off frequency, which should be around 80 to 100 kHz with a resulting slew rate of about 18 to 22 V/µs. However, it could also be the speakers! Be aware of the power when you connect your speakers! A tweeter or other parts can be damaged when overdriven. I have discovered this the hard way.

If you have checked all of these factors, you can enjoy your music at the highest level. I would be very happy if this article helps improve the sound quality in as many listening rooms as possible! Constructive criticism, comments and improvement proposals are always welcome.

Editor's Note: In a follow-up article, Harald Frank will describe a protection circuit for this amplifier to guard against output DC, over current/short circuit, and temperature. The article will also feature level control, fade in/out, and switch-on delay options that are all controlled by a small microcontroller.

Resources

"LM4702 (ACTIVE) Stereo High Fidelity 200 Volt Driver with Mute," Texas Instruments, www.ti.com/product/lm4702.

J. Walton, "High-End 120-W MOSFET IC Driven Amp," *audioXpress*, July 2007.

Sources

MPC74 Resistors
Futaba Electric Co., Ltd. | www.fu-futaba.co.jp/english

2SK1058 N-Channel and 2SJ162 P-Channel MOSFETs
Hitachi Global | www.hitachi.com

2SK2221 N-Channel and 2SJ352 P-Channel audio MOSFETs
RENESAS Electronics Corp. | http://am.renesas.com

S-1142B33-E6T1U Voltage regulator
Seiko Instruments, Inc. | http://www.seikoinstruments.com

LM4702 Speaker amplifier and modulator and LME49811 audio power amplifier input stage
Texas Instruments, Inc. | www.ti.com

Bill of Materials

Capacitors	
C1	10 µF; 400 VDC; 3%; M-Cap audiophile MKP capacitor
C2	10 µF; 16 V; 10%; elektrolytic capacitor
C3, C4, C5	30 pF; 300 V; 5%; silver-mica capacitor
C6, C7, C8, C9, C10, C11	470 nF; 250 V; foil capacitor
C12, C13, C17	220 µF; 100 V; 10%; elektrolytic capacitor
C14, C15, C16	100 nF; 100 V; foil capacitor
C18, C19	10,000 µF; 80 V; elektrolytic capacitor
C20, C21	3 × 10,000 µF; 80 V; elektrolytic capacitor
Diodes	
D1, D2	Panel mount bridge rectifier 35 A/1,000 V
D3	Single diode 35 A /1,000 V
D4	Zener diode ZPD 12; 0.5 W (optional, voltage dependent)
D5	LED according to preference, approximately 5 mA
Fuses	
S1	Fuse, slow-blow, 1 A/250 V
S2, S3	Fuse, slow-blow, 250 mA/250 V
S4, S5	Fuse, slow-blow, 4 A/250 V
Potentiometers	
P1	1 kΩ; 0.25 W; Trimpot
P2	100 kΩ, 0.25 W; Trimpot
Regulators	
IC1	LME49811; ±100 V; Audio Power Amplifier (Texas Instruments)
IC2	S-1142B33-E6T1U Voltage regulator 50 V to 3.3 V (Seiko Instruments)
Resistors	
R1, R3	1.8 kΩ; 0.25 W; 1%; metal film
R2, R4	56 kΩ; 0.25 W; 1%; metal film
R5	47 kΩ; 0.25 W; 1%; Metal film
R6	470 Ω; 0.25 W; 1%; Metalfilm
R7	220 Ω; 0.25 W; 1%; Metal film
R11, R12, R13, R14	15 kΩ; 0.25 W; 1%; metal film
R15	10 kΩ; 0.25 W; 1%; metal film
R8, R9	0.22 Ω; 5 W; 5%; MPC74 metal band FUTABA
R10	10 Ω; 5 W; 5%; CWR-5 wire wound resistor FUTABA
Rx	270 Ω; 0.25 W; 1%; metal film at Vx = 3.3 V
Transformer	
Tr1	Toroidal mains transformer two 38 V/43 V; 280 VA
Transistors	
T1	2SK2221; 200V / 8A; N-Channel Audio-MOSFET (RENESAS)
T2	2SJ352; 200V / 8A; P-Channel Audio-MOSFET (RENESAS)

Note: The enclosure, heatsinks, fan (if needed), screw/solder terminals, PCB material, cabling, mains filter, mains switch, and other not-specified amplifier parts depend on individual implementation and should be determined by the builder. It is not always useful to spell everything out and suffocate all creativity!

Build a Powerful Tube Amplifier

A Push-Pull AB Amplifier Using KT 120 Vacuum Tubes

Powerful 120 vacuum tubes are used in this push-pull amplifier design. The KT 120 tube is a more powerful version of the known KT 88. Adjustments to the KT 88's circuits enable them to be used for the KT 120. The mains transformer and output transformer must also be adjusted for the KT 120's higher output. Using of the KT 120 tubes enables us to build powerful power amplifiers for hi-fi equipment, guitar amplifiers, and PA systems.

By
Gerhard Haas
(Germany)
Original article published in German in Elektor's Roehren Sonderheft 9*, May 2013*

Photo 1: The push-pull AB amplifier project is shown with four KT 120 vacuum tubes.

Prior to building this powerful push-pull AB amplifier, it is important to learn more about the tubes that will power it (see **Photo 1**). The KT 88 and the KT 120 are power tetrodes. The suppressor grid, which in this case consists of discharge plates, is firmly connected to the cathode on the inside. The pinout for both tube types is the same. The metallic shielding shroud on the tube's lower end is also connected to the cathode. **Table 1** shows the tube data for the amplifier. **Table 2** shows the tube data for the power transformer.

The KT 120 yields higher limit values than the KT 88, which means a higher performance can be expected. However, the screen grid is the amplifier design's bottleneck. With both tubes, the maximum screen grid power loss must be no higher than 8 W. But with the KT 120, the screen grid voltage can be set to 600 V. The KT 88's screen grid voltage is only 450 V. Therefore, a slightly higher operating voltage can be used—which in combination with the slightly higher cathode current and the higher admissible anode power loss—promises a higher performance output. A major distinctive feature of the KT 120 is that with a fixed-grid bias, the maximum grid bleeder resistor must be relatively low—about 51 kΩ—so the grid circuit also needs to be adjusted. In contrast to other tubes, the required negative-grid bias is quite high, which requires high-voltage resistant capacitors. A component can only provide the best results if it is properly used. For more information, refer to available datasheets (see **Resources**).

The Two-Tube Design

Figure 1 shows the amplifier circuit with two KT 120 vacuum tubes, creating a powerful power amplifier. Since the KT 120 requires a lot of voltage for saturation, the driver stage needs to be designed accordingly. V1A provides the bandpass gain. V1B is the phase inverter. Both tube systems are operated by means of the voltage, which is stabilized by the Zener diodes D2–D5 and well filtered via the R27 and C3 components. R5 is the discharge resistor for the electrolyte capacitors C3 and C10.

Amplifier Data	2 x KT 120	4 x KT 120
Output power at 1 % THD	>120 W	>240 W
Power voltage	630 V	630 V
Bias voltage	-120 V	-120 V
Filament voltage	12.6 V	12.6 V
Filament current	2.1 A	3.9 A
Noise voltage, unweighted	≤400 µV	≤600 µV
Plate-to-plate resistance	3.8 kΩ	1.9 kΩ
Output transformer's core size	MD 102 B	PMZ 135/50

Table 1: The basic data for the two power amplifier versions is provided.

Power Transformer Data (AC)		
Power voltage	460 V/0.5 A	460 V/0.8 A
Bias voltage	85 V/50 mA	85 V/50 mA
Filament voltage, AC operation	12.6 V/2.3 A	12.6 V/4.2 A
Filament voltage, regulated DC operation	13 V/3.2 A	13 V/5.5 A
DC regulator's supply voltage	22 V/50 mA	22 V/50 mA
Core size	MD 102 B	PMZ 135/50

Table 2: The data for the transformers used in this design are shown.

The phase-inverter output signal is amplified by the differential amplifier V2 to the level required to fully drive the power tubes. The balance can be set via P1. The KT 120's grid bleeder resistor must not exceed a maximum of 51 kΩ at the fixed-grid bias. This requires a lower-resistance layout of the trimpots needed to set the idle current along with the associated resistors. The coupling capacitors in front of the control grids need to be sufficiently dimensioned so the amplifier's lower cut-off frequency inside the amplifier itself is not reduced.

The circuit board is designed to operate other tubes (e.g., the 6L6, the KT 88, the EL 34, and other models) using the same pinout. However, the pinout and the operating voltages must be adapted to the respective tube's requirements. Depending on the wire jumpers layout, triode, and pentode ultra-linear operation is possible. But, it is necessary to observe the conditions required by the respective tube type.

In this circuit, we used a pentode operation in an ultra-linear layout. This means a wire jumper for the suppressor grid is not necessary because all the beam tetrodes (i.e, the KT 120s) already have it connected to the cathode inside. The screen grids are connected to the respective taps for ultra-linear operation at the output transformer. The filter capacitor C9 for the negative grid bias has been designed for 100 µF and a dielectric strength of 160 V. This ensures a sufficient filtering effect. To set the required 65-mA idle current, the grid bias must be relatively high. This is also necessary because a high operating voltage (620–630 V) is present.

The screen grids are operated at their upper limit. However, they are protected by the current-limiting resistors R25 and R26. Similarly, the voltage slightly drops at the copper resistance of the transformer winding. The power losses remain well below the admissible maximum values, both in the idle and the saturation modes. They are utilized to a maximum of approximately 60% to 70%. The configuration using the KT 120 is not prone to high-frequency oscillation. In most cases, the correction capacitor C11 can be omitted. If it is necessary, opt for values between 10 pF and 100 pF.

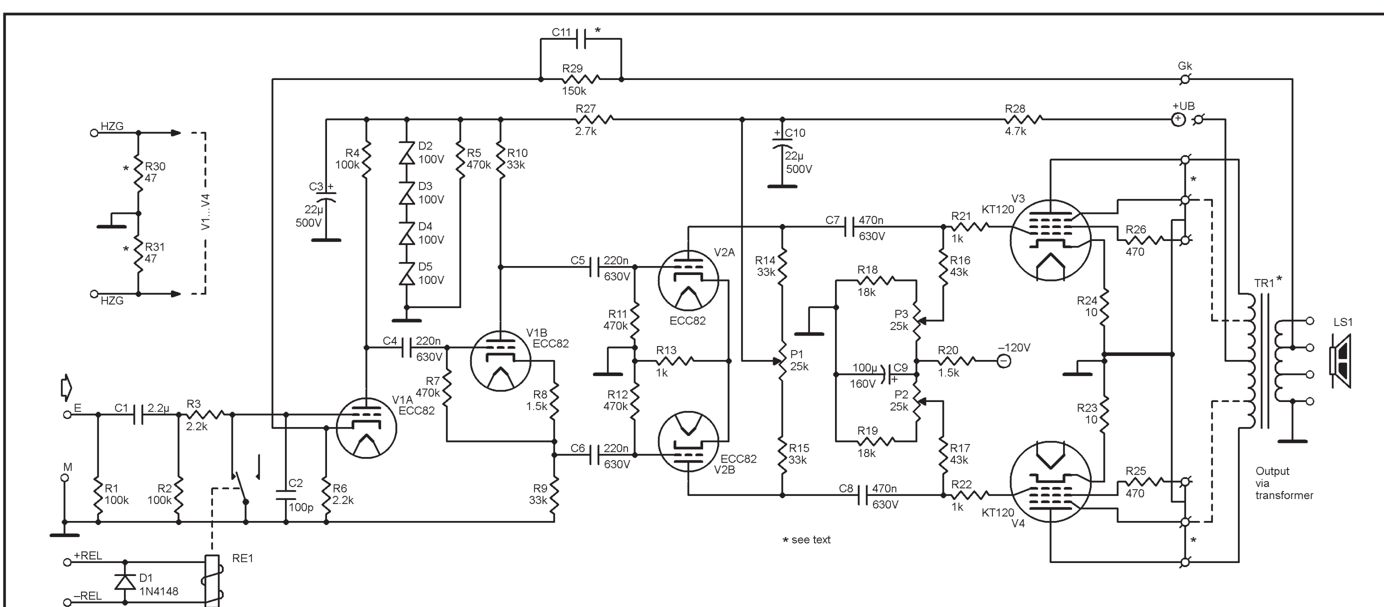


Figure 1: This amplifier circuit is designed using two KT 120 vacuum tubes. V1A, V1B, V2A, and V2B represent the ECC82 vacuum tubes. V3 and V4 represent the KT120 vacuum tubes.

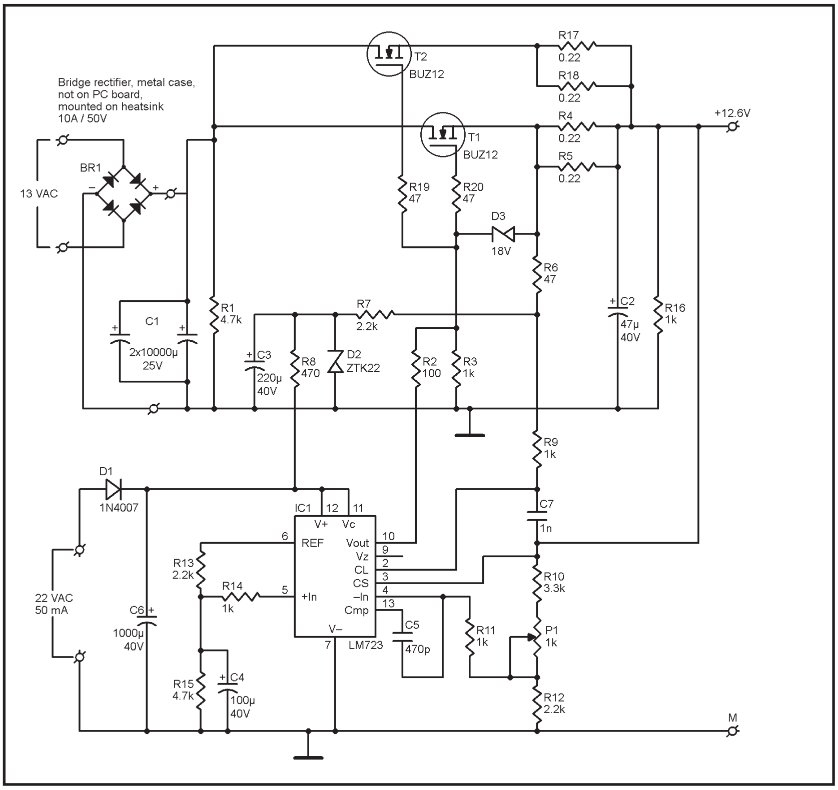


Figure 2: The power supply is shown with each power amplifier individually protected.

Tube heating on the circuit board is wired for 12.6 V. Use the resistors R30 and R31 if alternating current (AC) is used for heating. When heating is performed using direct current (DC), keep R30 and R31 if the voltage is fed at the two points labeled HZG. However, if the heating voltage is used for further purposes (e.g., operating relays), the two resistors do not apply. Connect the negative terminal of the DC heating to ground. This connection is absolutely necessary because the tube heating needs to have a ground reference for potential equalization. Also, the entire heating circuit can act like an antenna and cause substantial hum even with DC heating.

Figure 2 shows a suitable DC heating. It is designed in a way that the power loss and the heat generation are very low. It also ensures a gentle soft start of the tube heating and effective hum suppression. The transistors T1 and T2 have to be isolated and mounted on a heatsink with a thermal resistance of less than 3.5°C/W. The rectifier also needs to be mounted on a heatsink. We recommend using thermal compound between the heatsink and the rectifier.

A muting relay is supposed to be placed at the circuit's input. In idle mode, the input is short circuited. Once the relay coil receives power, the contact opens, enabling the signal to be amplified. This type of circuit's relay contact short circuits the input when switched off to avoid clicks. The relay can be operated via the DC heating with 12.6 V. R3 and C2 form a low pass that suppresses high-frequency

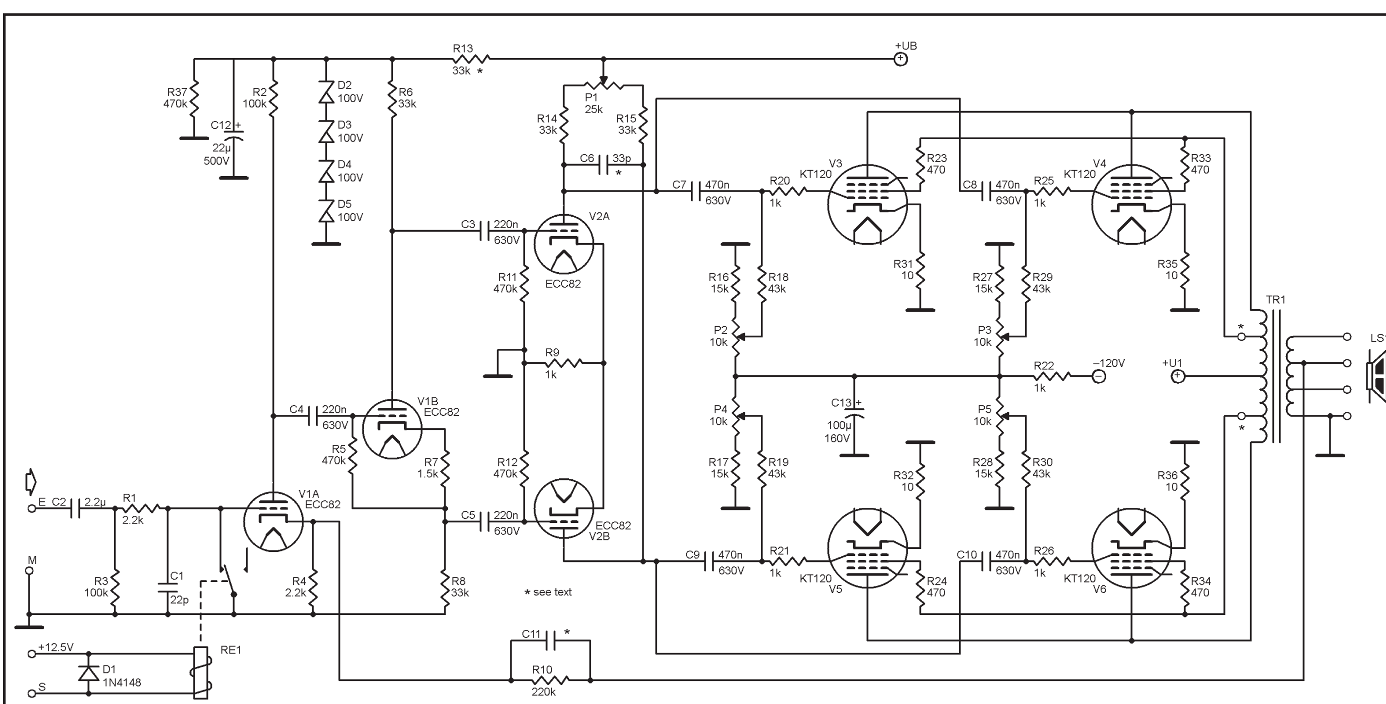


Figure 3: This project's schematics are designed to ensure low power loss and low heat generation.

disturbances. R3 also ensure that with a closed relay contact the source output does not have a total short circuit, which could lead to damage.

An Alternate Four-Tube Design

Figure 3 shows the circuit with four power tubes. The principle is the same as the design shown in **Figure 1** except twice the number of power tubes are used, providing more output power. The balance resistors for heating were removed from the circuit board can be externally mounted, if needed. Here again, C11 is only required in case of high-frequency oscillating tendencies. Otherwise, it can be omitted. C6 is planned as well. Depending on the design, oscillations can sometimes occur. C6 can eliminate them. Use 22 to 100 pF as reference values. Basically, if no oscillating tendencies occur, compensation capacitors are not needed. C6 must have a dielectric strength of 500 V. The power amplifier can also be muted with the Reed relay connected to the input. The relay can be operated via the DC heating.

The Power Supply

Powerful power amplifiers also require a corresponding power supply. **Figure 4** shows a circuit that can be used in several applications. On the AC side, Relay 1 (Rel1) is designed for switching the high voltage, which enables stand-by operation. Likewise, this relay can switch the power amplifier off if an idle run protection circuit is used. The protection circuit automatically identifies whether a speaker is connected or not. The high voltage is filtered via filter choke L1 and C3/C4.

The operating voltage of the preamplifier and the driver stage is further filtered via R5 and C5/C6. R5 divides the voltage to the potential required for the preamp. In case the power supply is used for a stereo power amplifier, the high voltage can be tapped via the fuses Si2 and Si3. Additional protection of the power tubes is useful. If a power tube kicks off because of overloading or ionization, the power amplifier is also switched off to prevent further damage. Basically, the high voltage is protected by a fuse on the AC side. Whether a power tube, electrolyte capacitors, or the rectifier gets damaged, an effective switch-off is important. With the high operating voltage and the relatively high transformer current, several things can literally go up in smoke if damaged. In such a case, a mains transformer can deliver a significantly higher current (about double its rated power) for a short time. In case more than 400 V with up to 1 A are present, resistors, semiconductors, electrolyte capacitors, and other elements could explode.

Because of the transformer alone, power amplifiers of this size are pretty substantial. The smaller version (with two power tubes each) can still be implemented as stereo power amplifier. However, the power amplifiers should be designed as monoblocks. It may require more effort to build the power supply and the housing, but an excellent channel separation contributes to an optimum sound resolution. If the power supply used for the stereo application contains two KT 120 tubes, each power amplifier can be individually protected (see **Figure 4**). With the larger version, one power supply for each power amplifier is recommended because of the higher demand for current. The fuse values given in the components

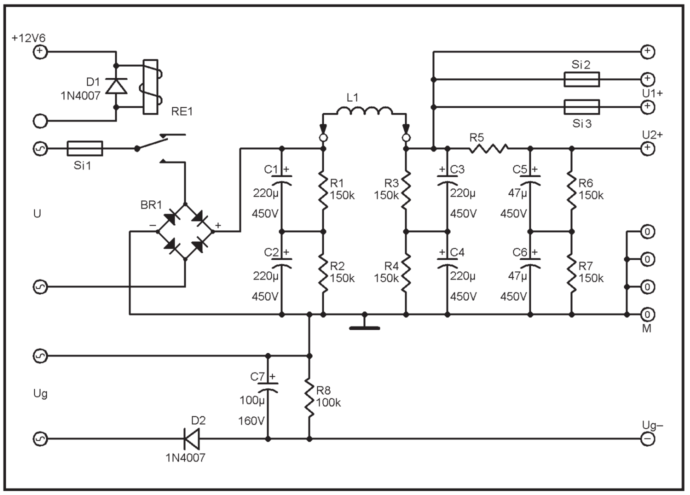


Figure 4: This power supply will work with an amplifier using four KT 120 vacuum tubes.

list are dimensioned relatively tightly, based on the theory that a tripping fuse is better than an amplifier going up in smoke. If the fuses continually trip when the amplifier switches on or in the event of high saturation levels, their values can be increased by one to two standard values without risking major damage.

Output Power

Next, let's determine how much output power can be achieved using the KT 120. The output power is always referred to as a maximum allowable total harmonic distortion (THD). Historically,

Photo 2: The KT 120 vacuum tubes are big tubes with big electrodes.

About the Author

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

the power of single-ended A-type power amplifiers is specified with a THD of 10%. With push-pull AB amplifiers, it is generally 5%, but often values between 3% and 10% are specified. Even-order distortion at high power is desirable. Another goal is to obtain as much power as possible with the least possible effort. The lower harmonic distortion of push-pull amplifiers can be achieved because the second harmonic k2—which is characteristic for single-ended power amplifiers—will be drastically reduced due to the symmetrical output stage. Even harmonics typically cancel themselves in push-pull configurations. The THD is lower, but the uneven harmonics—above the third harmonic k3—are more dominant, which often leads to a "dirty" sound. Skillful dimensioning, quality components, and suitable quiescent points ensures that the total distortion remains low without excessively heavy feedback. But such a configuration can still be made second-harmonic dominated. The goal for a good hi-fi power amplifier is not primarily to achieve the highest possible output power, but rather to give up some power output in favor of good sound.

In this amplifier, once the optimum symmetry is set by means of the symmetry potentiometer P1, maximum power can be reached with low harmonic distortion. As mentioned, in this case, the uneven harmonic distortion components above the third harmonic become more dominant. When the symmetry potentiometer is slightly misadjusted, the total distortion will increase early, with the even harmonics above the second becoming more dominant, improving the sound quality. The power amplifiers with the KT 120 vacuum tubes show smooth clipping characteristics and a moderate harmonic distortion curve over the output level, which provides a pleasant and smooth sound.

The feedback resistors R29 shown in **Figure 1** and R10 shown in **Figure 3** are designed in such a way that the power amplifiers can be driven to full output with about 1 to 1.5 V input signal. The feedback is connected to the transformer's 8-Ω tap. To achieve the lowest interference voltage, a shielded input cable is recommended. However, it should not be run inside a cable harness together with the AC mains leads. For a power amplifier with two KT 120 vacuum tubes, an interference voltage of about 400 µV can be expected if properly designed and used with DC heating. A power amplifier with four KT 120 vacuum tubes would have an interference voltage of about 600 µV. The interference voltage primarily consists of noise and is measured at the 8-Ω terminal.

Testing the Amplifier

When starting the amplifier, the standard rules must be observed. First, check the heating and the negative grid bias. The grid bias must be present everywhere and should be set to the highest negative value as a precaution. The output transformer must be provided with a load resistance. Then, after heating the tubes, the high voltage can be switched on. To set the idle current levels, the voltage drop is measured at the cathode resistors with 10 Ω each. The trimpot is used to set a value of 650 mV, which corresponds to a current of 65 mA. This procedure has to be repeated several times. Then, operate the power amplifier at medium performance for about 10 min. Next, operate the power amplifier in idle mode for about 2 min. Now, the idle current levels are set. This procedure is necessary as the KT 120 is a big tube with big electrodes (see **Photo 2**). It takes time for the cathode to really heat delivering the full current, so that the idle current must be set following this procedure. Furthermore, it helps make the new tubes more stable.

Table 1 contains important basic data of the two power amplifier versions. Details on the frequency response are not given as the overall result depends on the transformer's characteristics. It is of utmost importance that the transformers are able to cleanly transmit 20 Hz under load. Advertising or datasheets often feature almost fantastic frequency responses. Closer examination reveals that those frequency responses are measured at 1 W or even at 1 mW, using a signal generator. This has nothing to do with practical applications and is misleading. At about 1 dB under maximum output level—about 100 W with two KT 120 vacuum tubes and 200 W with four KT 120 vacuum tubes—the frequency response should be between 20 Hz (–0.5 dB) and more than 40 kHz (–1 dB), respectively. The recommendations shown in **Table 1** regarding the core size of the transformers are provided so that with appropriate design, the required value of 20 Hz is achieved. The recommended transformers also feature the best-grain-oriented sheet alloys and show low losses, enabling a compact design at high performance levels. Likewise, **Table 2** shows the recommended mains transformer cores and the required currents and voltages.

Author's Note: Complete kits and special parts (e.g., transformers, mains transformers, filter chokes, and circuit boards) can be purchased through several audio manufacturers including Experience Electronics (www.experience-electronics.de).

Project File

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

Resources

Genelax, "The KT88 Beam Tetrode," http://www.r-type.org/pdfs/kt88.pdf.

Tung Sol Electron Tubes, "Tung Sol KT120 Pentode," http://www.tungsol.com/tungsol/specs/kt120-tung-sol-specs-curves.pdf.

Parts List for Amplifiers with KT 120 Vacuum Tubes

Amplifier with 2 x KT 120 (Figure 1)

Capacitors

C1	2,2 µF/50 V, nonpolar
C2	100 pF, ceramic
C3	22 µF/500 V
C4,5,6	0.22 µF/630 V MKP 10
C7,8	0.47 µF/630 V MKP 10
C9	100 µF/160 V
C10	22 µF/500 V
C11	see text

Diodes

D1	1 N 4148
D2-5	Zener diode 100 V/1.3 W

Miscellaneous

Rel 1	relay 12 V, gold-plated contacts
2	noval sockets, print mounting
2	octal sockets, print monting

Potentiometers

P1	trimpot PT 15 Lv 25 kΩ, cermet
P2,3	trimpot PT 10 Lv 10 kΩ, cermet

Resistors

R1,2	100 kΩ
R3	2.2 kΩ
R4	100 kΩ, MO
R5	470 kΩ, MO
R6	2.2 kΩ
R7	470 kΩ
R8	1.5 kΩ
R9,10	33 kΩ, MO
R11,12	470 kΩ
R13	1 kΩ
R14,15	33 kΩ, MO
R16,17	43 kΩ
R18,19	18 kΩ
R20	1.5 kΩ
R21,22	1 kΩ
R23,24	10 W, MO
R25,26	470 Ω, MOX
R27	2.7 kΩ, MO
R28	4.7 kΩ, MOX
R29	150 kΩ
R30,31	47 Ω (only used at AC heating)

Tubes

V1,V2	ECC82
V3,V4	KT 120

* PC board	epoxy resin, 70-µm copper
Board size	150 mm × 160 mm
EE-310 A	(board type number)

DC-Heating (Figure 2)

Capacitors

C1	2 x 10.000 µ/25 V
C2	47 µF/40 V
C3	220 µF/40 V
C4	100 µF/40 V
C5	470 pF ceramic
C6	1,000 µF/40 V
C7	1 nF, ceramic

Miscellaneous

IC1	723 DIL
T1,2	BUZ 12
Gl1	bridge rectifier, metal case, 10 A 50 V
D1	1 N 4007
D2	BZX 79 C 22
D3	Z-Diode 18 V/1.3 W
1	IC socket DIL 14

Potentiometer

P1	1 kΩ trimpot, cermet

Resistors

R1	4.7 kΩ
R2	100 Ω
R3	1 Ω
R4, R5	0.22 Ω/5 W, metal band
R6	47 Ω
R7	2.7 kΩ
R8	470 Ω
R9	1 kΩ
R10	3.3 kΩ
R11	1 kΩ
R12,13	2.2 kΩ
R14	1 kΩ
R15	4.7 kΩ
R16	1 k
R17,R18	0.22 Ω/5 W, metal band

*PC board	epoxy resin with glass fiber, 70-µm copper
Board size	75 mm × 100 mm
EE-212 B	(board type number)

Amplifier with 4 x KT 120 (Figure 3)

Capacitors

C1	22 pF ceramic
C2	2.2 µF/50 V nonpolar
C3,4,5	0.22 µF/630 V MKP 10
C6	see text
C7,8,9,10	0.47 µF/630 V MKP 10
C11	10–100 pF, see text

Diodes

D1	1 N 4148
D2,3,4,5	ZY 100

Miscellaneous

Rel 1	relay 12 V, gold-plated contacts
2	noval sockets, print mounting
4	octal sockets, print mounting

Potentiometers

P1	trimpot PT 15 Lv 25 kΩ, cermet
P2,3,4,5	trimpot PT 10 Lv 10 kΩ, cermet

Resistors

R1	2.2 kΩ
R2	100 kΩ, MO
R3	100 kΩ
R4	2.2 kΩ
R5	470 kΩ
R6	33 kΩ, MO
R7	1.5 kΩ
R8	33 kΩ, MO
R9	1 kΩ
R10	220 kΩ
R11,12	470 kΩ
R13,14,15	33 kΩ, MO
R16,17	15 kΩ
R18,19	43 kΩ
R20,21,22	1 kΩ
R23,24	470 Ω, MOX
R25,26	1 kΩ
R27,28	15 kΩ
R29,30	43 kΩ
R31,32	10 Ω, MO
R33,34	470 Ω, MOX
R35,36	10 Ω, MO
R37	470 kΩ, MO

Tubes

V1,V2	ECC82
V3,V4,V5,V6	KT 120

*PC board	epoxy resin, 70-µm copper
Board size	126 mm × 343 mm
EE-129D	(board type number)

Power Supply (Figure 4)

Capacitors

C1,2,3,4	220 µF/450 V
C5,6	47 µF/450 V
C7	100 µF/160 V

Miscellaneous

Gl1	GBU 4 M
D1,2	1 N 4007
Rel1	relay 12 V spst goldplated contacts
Dr1	D-3275 3,2 H/0,6 A
Si1	0.63 A slow blow for 2 x KT 120 1 A slow blow for 4 x KT 120
Si2,3	0.4 A slow blow for 2 x KT 120 0.8 A slow blow for 4 x KT 120
3 pairs	fuse clips

Resistors

R1,2,3,4	150 kΩ, MO
R5	22 kΩ/9 W for 2 x KT 120 15 kΩ/11 W for 4 x KT 120
R6,7	150 kΩ, MO
R8	100 kΩ

* PC board	epoxy resin with glass fiber, 70-µm copper
Board size	130 mm × 170 mm
EE-220 B	(board type number)

**The PC boards are detailed on the* audioXpress *website see Supplemental Materials (http://audioxpress.com/page/audioXpress-Supplementary-Material.html)*

Resistors unless otherwise marked are metal film 0.6 W, 1 % tolerance
MO = Metal oxide 2 W, 2 % tolerance
MOX = Metal oxide 4 W, 5 % tolerance

Starved Amplifier Operation

Triode vacuum tubes can be used with very low plate voltages. Creating such designs may require data that is not available from the datasheet and include a different design process.

By
Richard Honeycutt
(United States)

I first encountered the term "starved amplifiers" in a 1960s *Popular Electronics* article. At the time, I was only interested in high-quality audio, since the article mentioned that starved amplifiers tended to have high distortion, it was only of academic interest to me. (Although I played guitar, distortion was considered undesirable so the article did not "grab" me from the musical electronics standpoint either.)

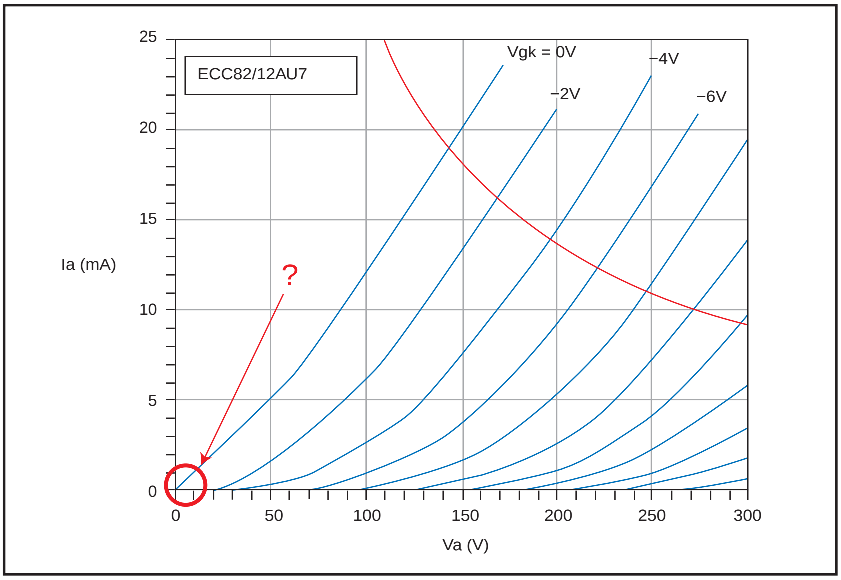


Figure 1: Sub-12 V operation is well off the beaten track. (Image courtesy of Merlin Blencowe)

I have since learned a few things about the term "starved," which can have two different meanings. Starved amplifiers can use either triodes or pentodes. And there are techniques for avoiding the high distortion the *Popular Electronics* article mentioned.

Historically, a "starved" amplifier was a pentode amplifier using a normal B+, but a very low screen-grid voltage. These circuits produced very high gains and were sometimes used in instrumentation amplifiers.

The more common use of the term "starved" now refers to a triode or pentode amplifier with a very low B+ voltage. Since the most common preamp tubes have 12.6-V filaments, a "12-V" wall wart supply can be used for both filament and B+ power. However, the B+ must be thoroughly decoupled from the filament supply to prevent unwanted feedback.

Triode Amplifiers

Let's start by discussing starved triode amplifiers. Our first challenge in designing a starved triode amplifier is finding the data. As shown in **Figure 1**, starved operation lives well away from the normally available data curves. The ordinate of this graph and those following is identified as "Ia," meaning anode current, which is the same as plate current.

Measurements of four popular tubes for plate voltages up to 10 V yielded the curves shown in **Figure 2**. Notice that the curves do not look much like high-voltage curves. Instead, they resemble a cross between high-voltage triode curves and pentode curves. This results from the control-grid bias voltage being very low. Also notice the knee in the curves at about V_P = 1 V for the 12A*7 dual triode vacuum tube family, and about 2 V for the 6DJ8 dual triode vacuum tube. One can expect the tube's behavior to change when the operating point traverses the knee, indicating that the distortion characteristics will change if the instantaneous plate voltage drops below 1 or 2 V.

The slope of the curves shown in **Figure 2** is the change in plate current divided by the change in control-grid voltage that caused the plate current to change. In other words, the slope of each curve at any given point is the tube's transconductance g_m at that point. If you notice the different scale of the ordinate in the curve for the 6DJ8, you will see that this tube has the highest g_m of any of the tubes measured. The amplification factor μ is the change in plate voltage divided by the change in control-grid voltage, with the plate current held constant. (As discussed in previous articles in this column, a constant-plate-current condition would require an infinitely large plate resistor or a circuit that mimics an infinitely large plate resistor.) Looking at the 12AX7 curve for a plate current of 100 μA, we see that as the control-grid voltage changes from −0.1 V to −0.2 V, the plate voltage changes from about 3.5 V to 8 V, giving a μ of (8 – 3.5)/0.1 = 45, not 100 as normally specified. We see that starved operation results in a lower μ for this "high-mu" triode. By comparison, the 6DJ8 has a μ of 27 for the same range of control-grid voltages.

The triode's plate resistance r_P is the inverse slope of the plate characteristic curve at the Q point:

$$\frac{\Delta V_P}{\Delta I_P}$$

With a −0.1 V control-grid voltage and 5-V plate voltage, the 12AX7 has a plate resistance of 80 kΩ. Under the same conditions, the 6DJ8 has a plate resistance of 5.7 kΩ.

As a point of interest, measurements show that "starved" operating characteristics of different tubes of the same type show quite a bit of unit-to-unit variation, although the individual triodes of a dual-triode tube such as the 12A*7 family tend to be pretty well-matched.

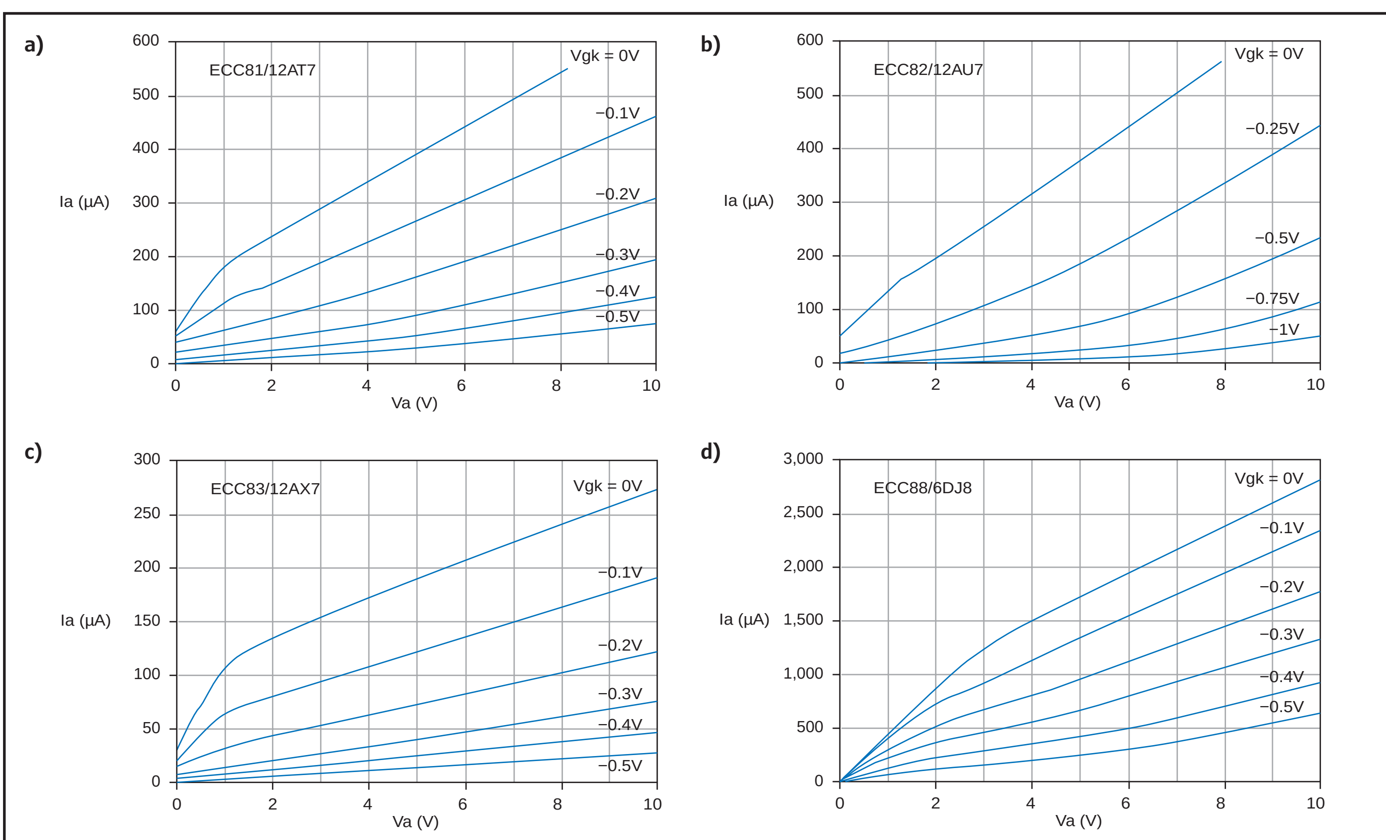


Figure 2: Starved anode characteristics of four popular triodes—the ECC81/12AT7 (a), the ECC82/12AU7 (b), the ECC83/12AX7 (c), and the ECC88/6DJ8 (d)—are shown. (Image courtesy of Merlin Blencowe)

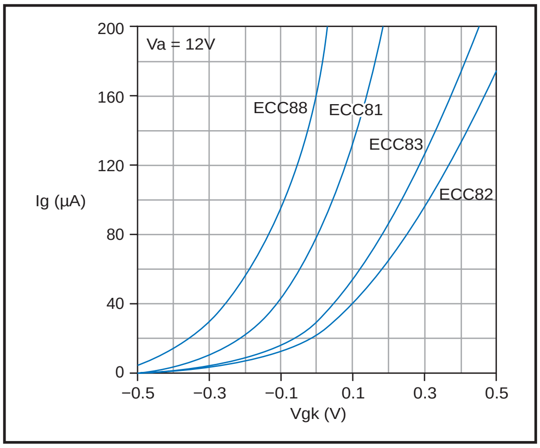


Figure 3: Average grid-current characteristics are shown for the same four triodes—ECC88/6DJ8, ECC81/12AT7, ECC83/12AX7, and ECC82/12AU7. (Image courtesy of Merlin Blencowe)

Low Control-Grid Voltages

When considering low control-grid voltages, we often make the unjustified approximation that the grid current is zero as long as the control grid is negative with respect to the cathode. However, the "pitchfork-in-the-rain effect" that makes grid-leak biasing work also operates with low negative control-grid voltages. **Figure 3** shows the grid current curves—taken at a plate voltage of 12 V—of the same four tube types used in **Figure 2**. Each curve represents the average values of several samples.

One thing we can learn from the grid characteristics is the triode stage's input resistance. In normal non-starved operation, we consider the tube's input resistance to be infinite so the amplifier stage's input resistance is just the grid resistor's value. The grid resistor's value is usually in the hundreds of kilohms except when grid-leak biasing is used, in which case, it is a few megohms. Not so with starved amplifiers. The grid characteristic curve's inverse slope at the operating point is the tube's input resistance. For the 12AU7 (ECC82), with a −0.1 V control-grid bias, the input resistance is about 5 kΩ. For the other types, the input resistance is even lower. This begins to make a starved triode look something like a bipolar transistor.

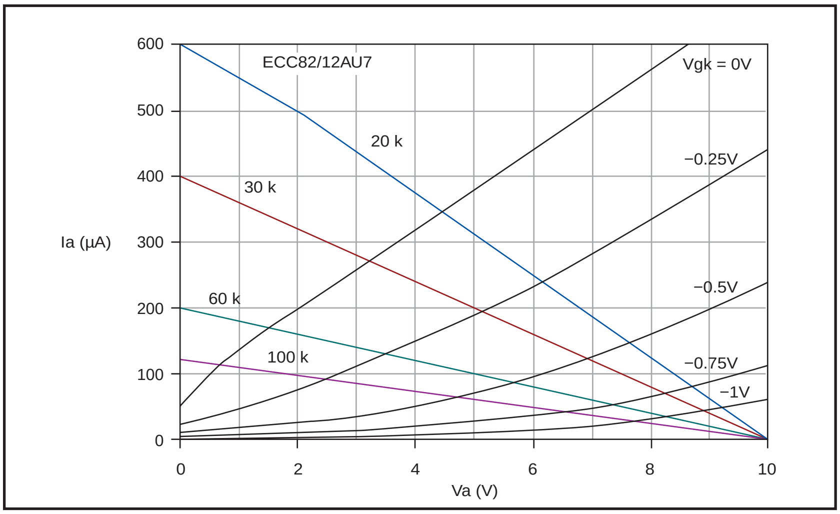


Figure 4: The graph shows an analysis using the load lines for an ECC82/12AU7 triode.

Control-grid biasing is another consideration in which a starved triode amplifier has a somewhat "transistor-ish" behavior. Since the useful values of control-grid bias voltage are so low, the effect of grid current in normal-valued grid resistors cannot be neglected. If we are biasing a 12AU7 control grid at −0.25 V, and we have a plate voltage of 12 V, a 100-kΩ grid resistor would be too large, since the resulting negative grid voltage would be enough to drive the triode into cutoff. To achieve a −0.25 V control-grid bias, we would need a grid resistor (in this case, acting as a grid-leak bias resistor) of 0.25 V/3.7 µA (the bias voltage divided by grid current from **Figure 3**), which gives us 62,500 Ω. When we calculate the parallel combination of this resistor and the 5-kΩ input resistance to the tube, we find that the amplifier stage's input resistance is just under 4.3 kΩ.

There are not many uses for a tube preamp stage with an input resistance this low. However, we could use a different approach to biasing: pull-up grid-leak biasing. What we do is connect the grid-leak resistor from grid to B+, rather than from grid to ground. Instead of only dropping 0.25 V across the grid-leak resistor, we drop maybe 12.25 V, assuming that we are using a 12 V B+. This increases its value to 874 kΩ. However, if the grid-leak resistor is 1% above or below the target value, the error in control-grid voltage will be of the same order as the bias voltage, making our bias voltage accurate only within ±100%, which is not very good. A possible solution is using a potentiometer for the grid-leak resistor.

At first glance, you might think that pull-up grid-leak biasing cannot work because it is like getting negative voltage from a positive supply. It is more accurate to think of the B+ connection as a voltage reference and the drop across the grid-leak resistor as a voltage source in series with the voltage reference. Thus, in our example, we have $V_{REF} + V_{GRID\text{-}LEAK} = 12\text{ V} - 12.25\text{ V} = -0.25\text{ V}$.

Figure 4 shows the 12AU7's plate characteristics with load lines plotted for several different values of plate resistor.

The harmonic distortion produced by a triode is primarily second-harmonic. The percentage of second-harmonic distortion (2HD) can be calculated by:

$$\%2HD = \frac{\Delta V_{LARGER} - \Delta V_{SMALLER}}{2\left(\Delta V_{LARGER} + \Delta V_{SMALLER}\right)} \times 100$$

Using a control-grid bias of –0.25 V and a maximum input of 0.5 V_{P-P}, we find the distortion for each value of plate resistor to be as shown in **Table 1**.

The 60-kΩ plate resistor gives a good combination of gain and maximum output voltage, and we will soon see how to reduce the distortion.

Figure 5 shows a starved triode amplifier stage. The resistor connected in series with the input is called a "grid stopper." Its purpose is to reduce distortion. Without this resistor, the stage produces normal triode distortion, consisting mainly of second and fourth harmonics with some third and some higher-order harmonics.

The grid current reflects these nonlinearities also, and the nonlinear grid current flowing in the grid stopper creates distortion components that are exactly out of phase with the components causing the distortion. Thus, the distortion appearing at the plate is canceled out without incurring the gain reduction that would be caused by using negative feedback. The grid stopper's value must be experimentally determined, but it is generally about 5 kΩ for a circuit such as the one we're considering. Increasing the grid stopper's value overcompensates, thereby increasing distortion and adding significant odd-order components.

Actually, the grid stopper does reduce the apparent stage gain, since it and the grid-leak resistor paralleled with the stage's input resistance act as a voltage divider for the input signal. In our example using pull-up grid-leak biasing, the attenuation caused by this voltage divider is a factor of about 0.44.

The output resistance of a triode stage is the parallel combination of the plate resistance r_P and the value of the plate resistor R_P. We can calculate r_P from the plate characteristic curve. It is approximately 23.5 kΩ for the conditions we are analyzing. The output resistance of the stage is about 16.9 kΩ. While this is not particularly high for triode stage, we would prefer a lower output resistance if the stage is to feed a subsequent op-amp or transistor stage.

Adding a Dual Op-Amp

One way to increase the input resistance, decrease the output resistance, and simplify the control-grid biasing arrangement is to add a dual op-amp to our circuit. By carefully choosing our gain, we can ensure that any distortion in the circuit is produced by the triode, using the op-amp only for clean gain, impedance translation, and a precise bias voltage.

Figure 6 shows such a circuit, using a Linear Technology LTC6241HV dual low-noise op-amp with rail-to-rail output capability. This complementary metal-oxide-semiconductor (CMOS)-input op-amp has a typical bias current of 1 pA, virtually eliminating output-offset-voltage concerns that could result from high resistances such as the 1-MΩ resistor from the inverting input to ground. The LTC6241 provides a rail-to-rail output, so it can be

R_P	V_{LARGER}	V_Q	$V_{SMALLER}$	%2HD	$A_{SMALLER}$	V_{OUTP-P}
20,000	7.7	6.15	4.5	1.5625	6.4	3.2
30,000	6.95	5.1	3.3	0.684932	7.3	3.65
60,000	5.7	3.6	1.6	1.219512	8.2	4.1
100,000	4.7	2.55	0.7	3.75	6	3

Table 1: This table compares the performance of the circuit using various values of plate resistor.

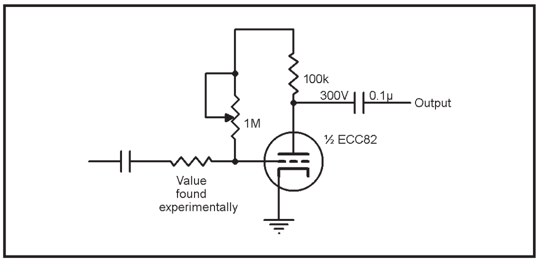


Figure 5: This is one possible circuit for a starved triode amplifier stage.

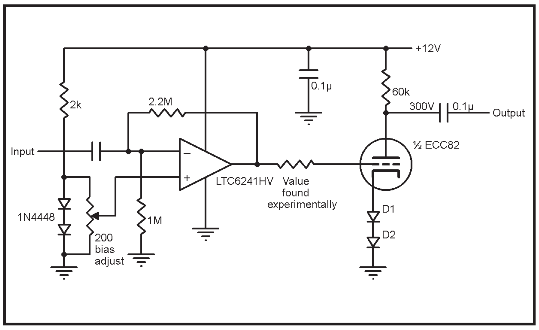


Figure 6: Adding an op-amp buffer stabilizes bias and increases input resistance.

operated with a quiescent DC output very near 0 V, the negative rail of a single-ended power supply. As drawn, the 12AU7's cathode is about 1 to 1.5 V above ground.

The 1N4448 diode sets the op-amp's non-inverting input at 1.24 V (typical) to 1.46 V (maximum), with a forward current of 5 mA, which is just about what the 2-kΩ dropping resistor allows. The op-amp's gain is 2.2 (2.2 MΩ/1 MΩ). With zero input voltage at the inverting input, the output voltage of the op-amp is 2.2 times the voltage at the non-inverting input. If the 200-Ω bias adjust potentiometer is correctly adjusted, the op-amp output will be 0.25 V less than the cathode voltage. This amounts to a control-grid bias of −0.25 V.

Since the LTC6241 has a maximum supply voltage of 12 V, a regulated power supply should be used. The 12AU7 filament requires 12.6 V at 115 mA, so a 12.6 V, 200-mA unregulated wall wart can be used, with an outboard regulator set to 12.6 V. Then a 1N4448 diode can be placed in series with the regulator output before feeding the +12 V point shown in **Figure 6**. Of course, a 12AU7 can operate with an unregulated supply, but these often put out more than the rated voltage, which could lessen tube life. Or, the 12AU7 could be run on a 12-V filament supply, instead of 12.6 V, but the gain would drop a bit, and noise would increase.

The 0.1-µF capacitor is for decoupling the power supply: making it act as a true AC ground at the op-amp's 12-V input. The op-amp adds back in the 0.44X attenuation caused by the grid stopper; increases the input resistance to a value friendly to a guitar pickup, high-impedance microphone, or any line-level source; and stabilizes the control-grid bias.

Since the LTC6241 is a dual op-amp, the second unit can be used as an output buffer, lowering the output resistance if desired. It can be made to supply added gain as well, but in this case, the gain should be set so that the op-amp's clipping point occurs at a level well above where the maximum desired distortion from the triode stage has been passed.

For example, the circuit shown in **Figure 6** has a total gain of $A_{OPAMP}A_{GRIDSTOPPER}A_{TRIODE} = 2.2 \times 0.44 \times 8.2 \approx 7.9$. At an output voltage of $4.1V_{P-P}$, this circuit produces about 1.2% 2HD. This would imply an input voltage of $4.1/7.9 = 0.519\ V_{P-P}$, or 183 mVRMS. If we wanted more distortion , or the same amount of distortion with less input voltage, we could increase the feedback resistor for the op-amp above the 2.2-MΩ value shown in **Figure 6**. The bias adjust potentiometer can be changed to provide the correct control-grid bias voltage even with a higher op-amp gain. But we are still limited to a maximum 4.1 V_{P-P} output voltage.

An LTC6241 output stage could increase this to almost 12 V_{P-P} before clipping is encountered, provided that the op-amp output is biased precisely at the midpoint between the positive (12 V) and negative (ground) rails. This means a maximum gain of $12/4.31 \approx 2.8$. In such a circuit, the tube distortion would prevail up to about 1.2% 2HD. Then, the op-amp would go into hard clipping. More gain in the input op-amp and less in the output stage would allow a wider range of triode distortion, as long as the negative excursion of the output from the first op-amp stage did not exceed the quiescent op-amp output voltage (the sum of the quiescent cathode voltage and 0.25 V), which is where the first op-amp stage would clip on the negative side.

Applying Method to Pentodes

This same approach could be applied to a starved pentode stage design, but the distortion percentages and signature (even- or odd-order) would be different. I do not know of any available plate- and grid-current curves for pentodes operated under starved conditions, and SPICE models may not be accurate under starved conditions. Also, the screen-grid bias adds another variable to consider. Therefore, anyone who wants to try a starved pentode design will have to do a lot of experimenting, which might include measuring tube characteristics.

Resource

M. Belencowe, "Triodes at Low Voltages: Linear Amplifiers Under Starved Conditions," www.freewebs.com/valvewizard2/Triodes_at_low_voltages_Blencowe.pdf.

Source

LTC6241HV op-amp
Linear Technology Corp. | www.linear.com

Member Profile

This project is awaiting completion. The view offers a glimpse of Dirk's workspace.

Member Name:
Dirk Sloos

Location:
Eindhoven, The Netherlands

Education:
HBO electronics

Occupation:
Dirk runs his own business building, modifying, and repairing electronic sound devices.

Member Status:
Dirk subscribed to *audioXpress* because both his main hobby and his occupation focuses on sound and electronics.

Audio Organization Affiliations:
He is a affiliated with Muze and Muze Records in The Netherlands.

Audio Interests:
Among his many audio interests are music, electronics, and tubes. He also enjoys good sounding gear.

Most Recent Audio Purchase:
Dirk's latest acquisition is a Wembley EL84PP amplifier from the 1960s. He said it is very musical, but it definitely needs some restoration. Dirk added that he has several of these amps from the 1960s with EL84 tubes because of their high-quality sound.

Current Audio Projects:
Recent projects include a new-old stock (NOS) tubed DAC, a 6SN7 preamplifier with a remote control, and a 211SE amplifier that he is building for a friend.

Dream System:
Dirk said he would need a lot of time and a lot of money because he would build multiple dream systems for several different purposes.

Industry Calendar

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

January 3–4, 2015 **ALMA International Winter Symposium 2015**
Tuscany Suites & Casino, Las Vegas, NV, US
www.almainternational.org

Founded in 1961, the Association of Loudspeaker Manufacturing and Acoustics (ALMA) International is the only international trade association dedicated to improving the design and manufacture of loudspeakers. The main gathering for the ALMA members occurs every year, just before the International CES. The ALMA Winter Symposium at the Tuscany Suites and Casino in Las Vegas is already established as a premier event for the loudspeaker and acoustics industry and the 2015 event will feature an expanded program that includes technical and business tracks, ending with the traditional cocktail party on Sunday.

January 6–9, 2015 **2015 International CES**
Las Vegas Convention Center, Las Vegas, NV, US
www.cesweb.org

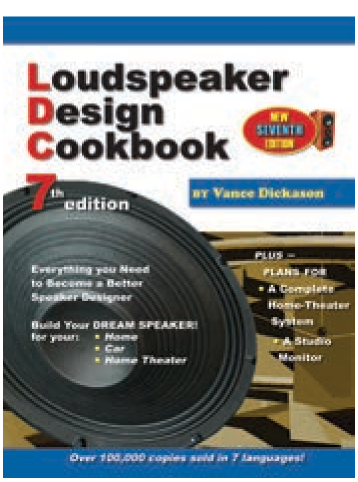
Once again the 2015 International Consumer Electronics Show (CES) will be the center focus for the consumer electronic industries, featuring several audio consumer vendors and the latest innovations, including everything from high-resolution audio to wearable devices and wireless streaming technologies. Immersive audio formats will certainly be one of the highlights at the 2015 CES with Dolby Laboratories showcasing it's Dolby Atmos for the home solutions, in parallel with the latest proposals in Multi-Dimensional Audio, which is the competing open platform solution proposed by the MDA Cinema Proponents Group. The solution is currently being implemented under the auspices of the Society of Motion Picture & Television Engineers (SMPTE). Promoted by the Consumer Electronics Association (CEA), the 2015 International CES is expected to attract more than 150,000 visitors and more than 3,500 exhibitors.

January 22–25, 2015 **2015 NAMM Show**
Anaheim Convention Center, Anaheim, CA, US
www.namm.org

The Winter National Association of Music Merchants (NAMM) Show in Anaheim is one of the most important annual industry events, mixing the passion for music, audio production, and sound in the most vibrant and happiest environment there is. Not only is this one of the most vibrant trade shows, full of product announcements and innovations, it is also a key opportunity for brands to build their images and reach the most important decision makers in the world. The show evolved very favorably in 2014 with great outdoor events and new educational sessions, while the NAMM Technical Excellence and Creativity (TEC) Awards have become a key moment for the industry to recognizing creative and technical achievements of audio professionals.

AX Shop

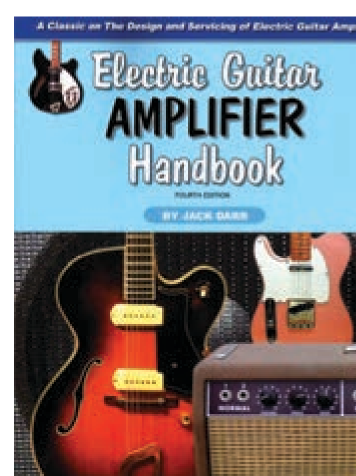
For more information and ordering **www.cc-webshop.com**



1 LOUDSPEAKER DESIGN COOKBOOK

This book's 7th edition provides you with everything you need to become a better speaker designer. It now includes Klippel analysis of drivers, loudspeaker voicing, advice on testing and crossover changes, and so much more! It is an excellent reference for loudspeaker engineering professionals.

Author: Vance Dickason | SKU #AA-BKAA068



2 ELECTRIC GUITAR AMPLIFIER HANDBOOK

The new reprint of the 4th edition contains three sections: "How Guitars Work," "Service Procedures and Techniques," and "Commercial Instrument Amplifiers." It is everything you need to know about how each part of the amplifier works and what to do if it doesn't work. The final section of the *Handbook* offers schematics, line drawings, and photos for many guitar amplifiers including models from Ampeg, Bogen, Baldwin, Fender, Gibson, Gretsch, and others.

Author: Jack Darr | SKU #AA-BKAA069

3 AUDIOXPRESS ARCHIVE CD

For added insight into woofers, tweeters, amplifiers, and all things audio, get the archive that goes where you go. *audioXpress* delivers a complete collection of project articles, product reviews, and audio engineering and technology information. It is also available in discounted, multi-year sets! Visit www.cc-webshop.com for details.

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