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More of Everything

A year ago, in November 2013, *audioXpress* received a facelift. We expanded the content, added several new columns and feature articles, increased our focus on audio research and development (R&D), product design, professional audio equipment, and acoustic science while maintaining our roots in DIY audio and audio electronics. *audioXpress* is committed to serving the students, the enthusiasts, and the industry professionals and increasingly appealing to the global audio engineering community.

In this very space, I wrote: "We believe *audioXpress* will blossom into a fascinating publication that follows the latest audio innovation trends, independent of the application field, and share a common audience of engineers, consultants, and enthusiasts in the electronics and audio fields, most of who are involved in R&D."

One year ago, I also evoked how Edward T. Dell, Jr. (1923–2013) devoted his life to people with a passion for audio electronics and founded Audio Amateur (rebranded as *Audio Electronics* in 1996), *Glass Audio*, *Speaker Builder*, and in 2000, *audioXpress*, which he edited until 2011. I think it would be appropriate to recall a few of Ed Dell's words and I found just the right ones in his January 2001 editorial (*audioXpress*'s first issue), titled "The Facts of Life for Magazine Publishing."

"This first issue of a new magazine seems a good occasion to review for readers just what is required for a healthy periodical. This one, with its pedigree already established, combining as it does 31 years of predecessor publications, is nonetheless totally dependent on three major ingredients—as well as a fourth one." Ed then explains why a magazine needs Readers—"Readers are, collectively, a kind of community,"—Authors—"In truth, authors are only an advanced form of reader,"—and Advertisers—"I believe no avocational magazine can be healthy or successful without advertising." The fourth "ingredient," Ed learned, is Passion. "My passion for good sound carried me along despite long hours of endless work, very slow growth, and even resorting to co-opting members of my young family. The task of producing a magazine is easier today as far as mechanics are concerned, but I think passion is still fundamental. I still look forward to coming to work each morning. Magazines stir the imagination, they surprise, they stimulate, they amuse. I am happy to report, somewhat immodestly, that I still enjoy that prospect myself."

As I reflect on the past year, I think we have done well in incorporating all four ingredients. Basically, we just need more of everything we have already committed ourselves to do.

And we would like to reaffirm our commitment to evolve and learn how to better serve our readers' interests. We hear you. We know we didn't make some of our high-end audio die-hards particularly excited with all the changes. But we couldn't ignore a new generation of readers who are interested in designing great audio products that include residential systems, headphones, home studio recording equipment, and tube guitar amplifiers. In doing so, we have gained a larger and more energetic community.

We continue to seek authors who want to contribute articles about audio equipment test and measurement, repair and restoration, and innovative new designs. If you feel up to the task, send an email to editor@audioxpress.com. And, of course, we want to offer the best and most effective platform for our advertisers.

As Ed Dell stated, we share the passion and "look forward to coming to work each morning."

João Martins
Editor-in-Chief

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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.

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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.

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Wireless Audio (Part 3)

Seeking Better Quality Wireless Audio Solutions

As we have discussed in the previous articles, Bluetooth and Wi-Fi technologies are powering wireless audio applications particularly since Bluetooth V. 4.0 combined with the High-Speed (802.11) Protocol Adaptation Layer was introduced. While Wi-Fi continues to evolve with more frequency bands, higher bit rates, and device-to-device communication, wireless audio technologies and development platforms are also quickly evolving to support higher quality audio transmissions.

By
João Martins
(Editor-in-Chief)

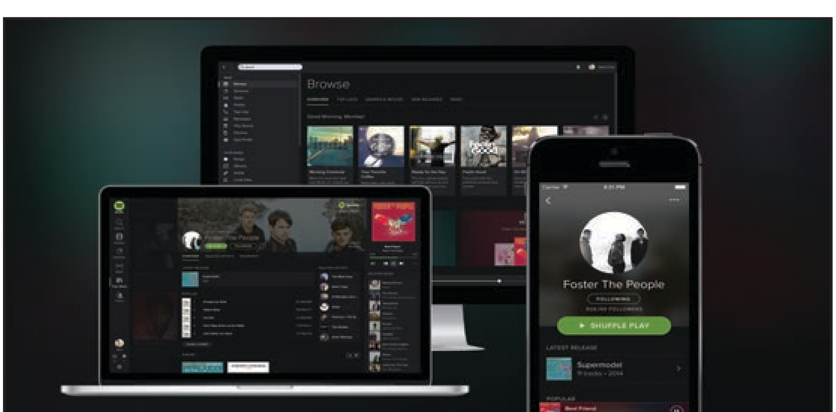
Wireless audio on Bluetooth and Wi-Fi has now surpassed the basic quality thresholds. So, consumers are increasingly adopting wireless audio for the convenience and not worrying so much about reliability. Consumers mainly see cabled connections as simple and reliable but less convenient. Although reliability is a term consumers are only now starting to associate with wireless, when this is coupled with the consumer's perception of quality and convenience, the wireless audio market will expand.

And nothing is contributing more to the consumer's perception that "wireless works" than streaming music from the Internet. Recently, the growth in commercial music streaming services' popularity is creating the turning point from which we can expect wireless audio transmission to become a commodity in the consumer's view. After all, if we are able to stream music from the Internet to any mobile device on a mobile network, why wouldn't consumers expect that same "stream" to continue from the same mobile device through a wireless home network directly to speakers, hi-fi systems, or wireless headphones?

Still, that wireless link to speakers is where the consumer's perception of quality is critical and where the wireless audio market is currently focusing in an effort to combine reliability and convenience.

Sonos started the "revolution" in consumer perception. It was globally promoted by Apple's AirPlay, and it is reaching new levels with services such as Spotify Connect.

Spotify Connect does not require the mobile device to continuously stream content to a wireless speaker or an AV receiver. Spotify Connect's key benefit is that once a track has been selected on the mobile device, using the touchscreen interface, the audio stream is directly delivered from Spotify's servers to a wireless speaker or wireless "receiver" using the local network. The implementation cuts out the middle device, directly streaming from the cloud service to compatible speakers. This frees the



Spotify Connect does not require the mobile device to continuously stream content to a wireless speaker or AV receiver. The key benefit of Spotify Connect is that once a track has been selected, the audio stream is directly delivered from Spotify's servers to the wireless speaker using the local network.

Sennheiser introduced Kleer wireless transmission in its 2008 range of headphones, proving the wireless audio experience could considerably improve compared to Bluetooth. In 2012, Sennheiser, like Sony, adopted direct-sequence spread spectrum (DSSS) technology. DSSS technology enables users to transmit an uncompressed wireless audio signal. Unfortunately, consumers were not interested in simple "cable replacement" approaches due to the benefits of wireless local area networks (WLANs).

mobile device for use during music playback, which greatly reduces battery depletion and enables the mobile device to move anywhere in the network without interrupting music playback. Suddenly, audio devices (e.g., wireless speakers, AV receivers, Internet radios, home theater systems, wireless speakers, and portable music player docking stations with Spotify Connect support) are bridging what could be considered the last frontier in consumer convenience with digital services.

Naturally, within the same local personal audio networks, when consumers connect their own audio sources to wireless speakers or to a complete multi-zone residential network, they will expect higher reliability and quality.

Better Than Cables

As we have seen with perfected digital transmission techniques, professional applications are combining the best of datastreaming protocols and error correction techniques with advanced signal processing. So, why wouldn't audio wireless transmission be as good (or better) than cable transmission? Perhaps, there are bandwidth limitations? We just have to examine current video streaming technologies and over-the-top television experiences—which inspired concepts like Spotify Connect—and the way the dynamic protocols are able to deal with variable bit rate and bandwidth restrictions to understand that we are very close to solving that particular bottleneck. That is, if we still have one, considering all the technologies already in place.

It's clear from recent market developments that wireless audio applications will evolve from the Bluetooth transmission used to connect current smartphones and tablets to speakers/headphones (short distance) into powerful wireless audio networks based on Wi-Fi (IEEE 802.11*xx*) with larger area coverage. There are many potential applications for residential and commercial audio distribution on wireless local area networks (WLANs). And, Wi-Fi will be increasingly used by consumers to create wireless personal area networks (WPANs). Bluetooth and other data protocols (e.g., NFC and ZigBee) will still be a part of those solutions, mainly because they are more efficient with battery-operated devices and can be used for basic pairing and control.

Digital wireless audio transmitters on WLANs can already support multichannel transmissions. Even for basic stereo, the left and right signals on input are kept separate but synchronized on the receiver output. It is also possible to connect multiple receivers to one transmitter, enabling multi-room distribution. The transmission of multichannel signals (surround-sound formats and even immersive 3-D formats) is easy to implement in WLAN solutions, using advanced or lossless codecs and carrying auxiliary metadata to coordinate playout at the receiver end.

Wi-Fi-based wireless audio applications can quickly overcome the quality and reliability issues associated with Bluetooth, while offering all the complementary advantages of data networks including interoperability with Internet streaming, which is vital for subscription services (e.g., Pandora, Spotify, and many others).

Although the industry struggled with bandwidth restrictions up to 2013, in less than two years we now have new WLAN-based solutions offering

Sonos was undoubtedly the pioneer in wireless multiroom audio solutions. The company leveraged all the benefits of wireless technologies to create a dedicated local Sonos network, which enabled the streaming of digital audio to any Sonos device.

Sonos has developed apps for Android and iOS and desktop software for PCs and Macs to control its multiple device wireless system.

uncompressed audio transmission that were previously only available on dedicated radio frequency (RF) systems.

Those dedicated RF solutions—such as the KLEER technology used by Sennheiser or the direct-sequence spread spectrum (DSSS) modulation used by Sony for their wireless living-room headphones—showed it was possible to have high-quality without the interference problems that plagued earlier digital implementations (which were still much better than earlier analog FM transmission solutions). Those systems were basically designed as "cable replacement" systems to wirelessly connect audio sources to speakers/headphones, but their success was limited. Current wireless audio applications go way beyond that and they are already in line with what consumers are starting to perceive as "convenient."

Sonos the Pioneer

The wireless audio "convenience" concept started in 2002 with Sonos, a consumer electronics company from Santa Barbara, CA. Sonos had a vision to reinvent home audio for the digital age and fill every home with the highest sound quality music. Sonos quickly became the leading wireless music system manufacturer. The Sonos Wireless HiFi System enables users to wirelessly stream music in every room—currently with control from Android and iOS smartphones and tablets, or desktop software on Macs and PCs.

The system can stream music from network attached storage (NAS) drives (including Apple's Time Capsule) and supports several formats including MP3, AAC, Ogg Vorbis, FLAC (up to 16 bit/44.1 kHz), Apple Lossless, WAV, AIFF, and WMA. In addition to playing personal digital music collections, Sonos was also a pioneer in providing direct access to streaming services such as Deezer, JUKE, Last.fm, MOG (which became Beats Music), Napster, Pandora, Rdio, Rhapsody, SiriusXM, Spotify, and many more.

The Sonos Wireless HiFi System uses standard Wi-Fi networks but implemented in a proprietary way (SonosNet). It combines wireless and/or Ethernet connections to connect both Internet sources and files stored on the local network.

Sonos started designing simple modules to combine any analog sources as inputs to the system and make them available for streaming to any other units on the Sonos network, up to a maximum of 32 devices. The concept progressed into different types of Sonos players with CONNECT:AMP modules connecting to unpowered speakers, CONNECT modules connecting to an existing audio amplifier or receiver, integrated wireless speaker/receivers, and even a subwoofer and a playbar for use with a television.

With the Sonos PlayBar, the company entered the home-theater market, combining nine speakers, each with its own Class-D amplifier with support for Dolby Digital input. The speakers also could connect with a Sonos SUB or two Play speakers to form a surround-sound home-theater unit. In fact, Sonos was the first company to address wireless stereo configurations, combining a stereo pair of its Sonos Play:1/3/5 speakers (Zoneplayers).

All the Sonos devices connect to each other using a proprietary peer-to-peer synchronous mesh network with AES encryption, with support for multiple separate zones. A single ZonePlayer or

The Pure Jongo family of portable wireless speakers—developed by Imagination Technologies—was the first available solution powered by the Caskeid family of technologies. Caskeid technology delivers efficient and accurate wireless multiroom and multichannel audio, addressing accurate and advanced synchronization and with effective low latency.

ZoneBridge must be wired to a network for access to LAN and Internet audio sources. SonosNet 2.0 was an evolution of the original 802.11g implementation, integrating MIMO antennas on 802.11n hardware for a more robust connection. Advantages for this approach included less audio dropouts, larger area coverage, placing the speakers farther away from the router.

The proprietary mesh SonosNet network helped create consumer awareness to the wireless audio and multi-zone concept, which effectively opened the market for other players because consumers quickly realized that they needed to improve their home Wi-Fi networks. The fact that every Sonos player or bridge has to constantly maintain a

The AirPlay Advantage

Since Apple always controls its own hardware/software ecosystem we can always expect Apple's approach to wireless audio to flawlessly work with whatever formats are supported by Apple's iTunes and the brand's hardware. Steadily, Apple has been improving AirPlay, being among the first brands in the market to adopt the most recent specifications.

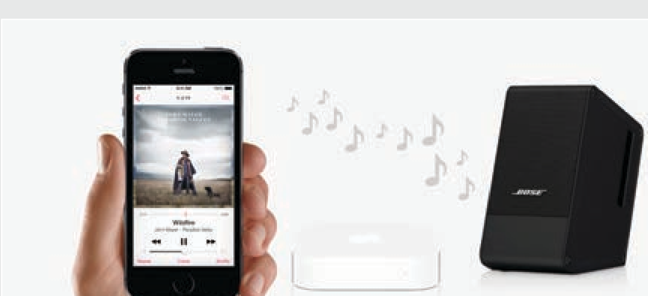
In 2008, Apple upgraded its whole range of products, including the AirPort Express to support simultaneous dual-band 802.11n Wi-Fi at the 2.4-and-5-GHz frequencies. In 2013, Apple introduced three-stream 802.11ac technology to the AirPort Extreme Base Station and progressively added it to its range of Mac computers.

A current MacBook Pro connected to an AirPort Extreme is able to reach data rates of up to 1.3 Gbps (triple the previous 802.11n standard) with double the channel bandwidth, using the new beamforming antenna arrays.

AirPlay technology benefits from those advances and that's one of the reasons why Apple's approach has proved so popular, even though its ecosystem remains restricted to approved third-party apps and it still doesn't support high-resolution audio (HRA)—although we might expect it will soon.

Since the early implementations, AirPlay technology enabled users to work with multiple devices in multiple rooms without any degradation in audio quality (i.e., it streams in Apple Lossless any of the codecs supported by iTunes, including AIFF, WAV, AAC and Apple Lossless). AirPlay doesn't add data compression to stream any audio file that iTunes can play. The streams simply use the Apple Lossless audio codec (ALAC) to transmit two channels at 44.1 kHz, AES encrypted. The stream is buffered before playback begins, resulting in a small delay before audio is output.

AirPlay uses user datagram protocol (UDP) for streaming audio, based on the real-time streaming protocol (RTSP) network control protocol. And since 2011, Apple made the ALAC codec available as open source and royalty free. The AirPlay protocol also supports auxiliary metadata and remote control of the receiver, enabling digital processing to be done only at the receiving end and providing multiple zone distribution without quality degradation.



The protocol supports metadata packets that determine the final output volume on the receiving end. This makes it possible to always send unprocessed audio data at its original full volume, preventing the sound-quality deterioration due to reduction in bit depth that would occur if changes in volume were made to the source stream before transmitting. iTunes also includes sound processing features (e.g., equalization, "sound enhancement," crossfade, and playback volume adjustment) that are user switchable.

As the heart of the solution, iTunes is not the best player in the world—in fact, the interface got much worse with iTunes 11—but at least it runs on Mac OS, iOS, and Windows. And, the configuration is among the best there is, including with third-party AirPlay-compatible wireless speakers, soundbars, and AV receivers.

With Wi-Fi Direct support on AirPlay since 2011, Apple's devices are able to create direct audio streaming among compatible devices, without the need to create a Wi-Fi network.

Apple licenses the AirPlay protocol stack as third-party software component technology to manufacturers and the evolution has basically followed the changes on the underlying Wi-Fi technology without much improvement in terms of wireless audio support for the last four years. Apple has been working on the technology mainly for video streaming support, mirroring, printing, and peer-to-peer file sharing. But at least the AirPlay wireless compatible devices work flawlessly (from companies such as Bose, Yamaha, Philips, Marantz, Bowers & Wilkins, Pioneer, Sony, and dozens of others), with all the song metadata and album artwork transmitted to compatible devices with graphical displays.

AirPlay will probably receive a big boost following Apple's acquisition of Beats. The company is always able to introduce improvements to its most recent products through software and firmware upgrades. Something the other manufacturers are learning is the new reality in consumer electronics, following the high standards set by Apple.

In 2013, Apple introduced three-stream 802.11ac technology to the AirPort Extreme Base Station and progressively added it to its line of Mac computers.

wireless connection, even when in standby mode or connected by cable, created issues with other simultaneous uses of Wi-Fi and introduced users to crowded environments with too many wireless devices.

Sonos paid a price for being a pioneer in the wireless audio market. It created the consumer awareness that enabled Apple to enter the space with AirPlay devices and later many other competing solutions. Still, Sonos is keeping pace with the technology evolution and remaining a very successful wireless audio company.

Advanced Wireless Audio

The huge potential of WLAN technologies for wireless audio has meanwhile been clearly demonstrated by Apple in its AirPlay implementation. Apple was able to increase the bit-rate, coverage efficiency, increase transmission reliability and create device-to-device connections, by simply adopting the most recent IEEE802.11 specifications, following a very standard approach to the use of the available technologies.

AirPlay evolved from the AirTunes audio-only protocol developed by Apple. It currently enables wireless streaming of audio, video, photos and related metadata, including AirPlay Mirroring between devices. Although AirPlay is restricted to Apple's closely guarded ecosystem, the AirPlay protocol technology is licensed enabling manufacturers to integrate the technology into wireless speakers, docks, AV receivers, and audio components, creating a reference implementation that inspired the whole industry.

Of course, AirPlay provides a good perspective of what's possible since Apple controls every aspect of the implementation and is able to easily introduce improvements as it sees appropriate.

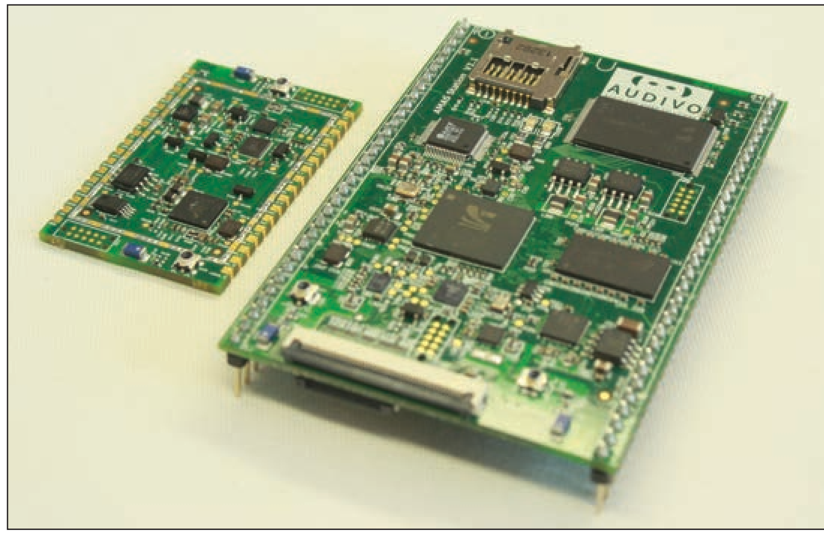
Following Sonos's and Apple's lead, there was a succession of proprietary and short-lived attempts to create similar Wi-Fi audio streaming solutions. Fortunately, the market is currently moving toward compatibility (not yet interoperability) around new (more powerful and versatile) platforms and application program interfaces (APIs).

Semiconductor companies currently offer complete development platforms for wireless audio, such as Texas Instruments's (TI) PurePath Wireless audio products. The PurePath Wireless audio products feature 2.4-GHz audio ICs that transmit uncompressed CD-quality (16-bit 44.1/48 kHz) wireless audio over an RF link, with error-free, lossless transmission in two- and four-channel system-on-chips (SoC) integrated solutions and USB audio support for all major operating systems.

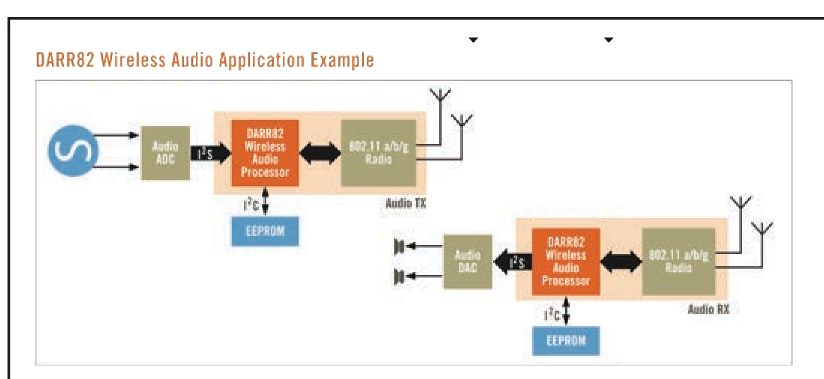
Other manufacturers are now offering new generation SoC wireless audio and sound processor solutions to power this evolution in wireless audio. Imagination Technologies is offering its Caskeid technology for synchronized wireless multiroom connected audio streaming technology.

Caskeid solutions work over Wi-Fi with low-latency synchronization, which the company claims to be "far superior to any other multispeaker solution on the market today and is indistinguishable from the timing accuracy of a wired system." The unique technology delivers stereo playback with less than 25-µs delay and supports wireless surround sound systems.

Caskeid technology is available for licensing. Imagination Technologies also offers custom SoC designs, based on MIPS and Ensigma wireless connectivity technologies, and includes a Software Developer Kit (SDK) enabling agnostic app support over Wi-Fi. Caskeid-enabled speakers can be used with any music streaming services including Pandora, Rdio, Deezer, and Spotify combining Bluetooth and Wi-Fi. Both PEAQ and Pure are already shipping Caskeid-enabled speakers and Onkyo and



The Audivo development platform integrates CSR's VibeHub networked audio platform to significantly ease development of whole-home audio solutions. This system-on-chip (SoC) module is a Wi-Fi and Bluetooth 4.1 turnkey solution that enables ODMs to quickly develop and easily integrate networked audio capabilities into their products and support multiple platforms, including Apple's AirPlay, UPnP and Digital Living Network Alliance (DLNA) sources via Ethernet/WLAN and Bluetooth with CSR aptX coding.



SMSC's DARR82 wireless audio processor (part of Microchip's portfolio) provides up to four uncompressed stereo audio channels with low latency and bidirectional audio support using dual-band Wi-Fi (2.4/5 GHz) with support for 16- and 24-bit audio up to 96-kHz sampling and multipoint-to-multipoint transmission, enabling multiple receivers to listen to the same audio stream.

Microchip's JukeBlox SoC and SDK platform is currently one of the strongest options available to manufacturers to implement wireless audio in consumer electronics products. JukeBlox offers integrated Wi-Fi and support for direct streaming from the Internet, including Spotify Connect. Some of the biggest brands in consumer electronics, including Denon, Pioneer, Harman-Kardon, Nokia, and Philips use JukeBlox technology in their products.

Canadian company Bluesound is one of many manufacturers targeting high-quality wireless audio applications, including support for high-resolution audio (HRA) sources or at least 24-bit digital audio.

Resources

Apple, Inc., www.apple.com.

Auralic, Ltd., www.auralic.com.

Bluesound International, a division of Lenbrook Industries, Ltd., www.bluesound.com.

Imagination Technologies, Ltd., "An Introduction to Caskeid—Wireless Stream Synchronization IP, www.imgtec.com/ensigma/caskeid.

KLEER, "Wireless Digital Audio Quality for Portable Audio Applications," December 2007, Microchip Technology, Inc., www.kleer.com.

Microchip Technology, Inc., www.microchip.com.

MOOS Audio, www.moosaudio.com/wireless-speaker-technology.

Sonos, Inc., www.sonos.com.

Meridian Audio will use the technology in future products. Many more licensees are in the pipeline to integrate Caskeid or selected technologies from the Caskeid family.

Other solutions currently in the market (e.g., DTS's Play-Fi) explored the same advantages of the Wi-Fi technology and currently the market offers several competing and very similar implementations, including Qualcomm's AllPlay, which was announced in January 2014.

Marketing Promises

As we said, many companies have tried to emulate Sonos's success with similar proprietary approaches, including Bose, Denon, KEF, LG, Samsung, and others—using Wi-Fi or Bluetooth+Wi-Fi as the foundation. Another company, Bluesound, tried to go a step further by streaming 24-bit digital audio and offering native support for high-resolution audio files using 802.11n Wi-Fi.

Of course, Bluesound does not guarantee that high-resolution audio (HRA) files will be streamed. The company just says that the Bluesound controller app "is designed to grab the highest possible bit rate. The Wi-Fi streaming of high-resolution files requires a robust and fast wireless network."

While Wi-Fi is evolving faster in terms of bit rate, consumers will need to be wary of marketing promises for a while. While it may be true that some of these systems are able to play HRA files that does not mean the wireless audio transmissions will occur in a quality-mode that is compatible with the high-resolution of the original files—especially when other transmission technologies are involved.

Bluesound at least has designed its new generation solutions to be firmware upgradable and that indicates it is aware of the need to support fast-developing WLAN technologies.

Because of the competitive environment and consumer awareness about HRA, Sonos recently announced plans to support HRA through the Deezer Elite streaming media service, which provides access to free lossless audio codec (FLAC) files at a standard of 1,411 kbps or higher. As they clearly state, we are talking about "5X the top bit rate of other streaming services."

It is clear that Sonos is being cautious regarding how far its next hardware and platform upgrade should go. Sonos does not want to alienate its existing client base, yet at the same time, it wants to answer increasing requests to support something more demanding than simply carrying "uncompressed" audio transmission of already compressed files (even if they are lossless).

The Continuing Quest for Realistic Recorded Sound Reproduction

By
Ron Tipton
(United States)

The quest for realistic reproduction of recorded sound is ongoing and we may never be able to fully recreate the ambience of a live performance. But we will keep trying. My goal for this series is to trace the evolution from mono to present-day binaural recording with an emphasis on how you can recreate and enjoy many of the steps along the way with free or low-cost software or hardware. I hope you enjoy the trip!

In the beginning, there were mono (single channel) recordings. First on tinfoil, then wax cylinders, shellac discs, vinyl discs, magnetic wire, and magnetic tape. The first commercial shellac discs, a little more than 100 years ago, brought professional music into people's homes and were considered nearly miraculous. Recording began with one mechanical pickup and one mechanical reproducer, both acoustic horns. With the advent of electronics this became one pickup microphone producing a single output regardless of the number of loudspeakers used. Marvelous, yes, but lacking the ambience and spatial realism of a live performance.

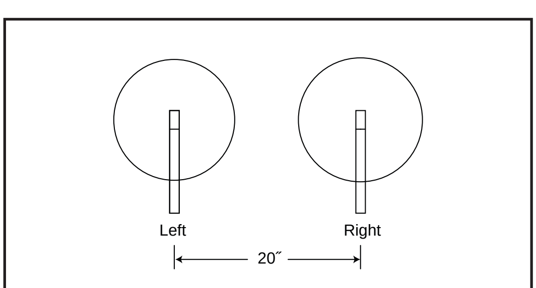


Figure 1: The usual microphone placement for a conventional A-B stereo pickup is shown.

Conventional stereo uses two microphones to record two files, usually (but not always) the left and right channels. (I'll address the "not always" later.) This seems simple, but the type of microphone and its arrangement is important because it determines the width of the stereo image and your ability to localize sounds within the image.

There are basically three microphone types: omnidirectional, which is a circular pattern with the microphone at the center; unidirectional, which is a cardioid or heart-shaped pattern; and bidirectional, a figure-eight pattern. There are four common arrangements: A-B, ORTF, X-Y, and Mid-Side.

There are two other types, which I will explore later. The line or "shotgun" microphone is a unidirectional microphone with a narrowed pickup pattern, that is, more directional. There is no standard for microphones of this type so pattern width varies among manufacturers and models. The pattern shape is usually graphically or numerically described on the specification sheet.

The other microphone type is the quad with four pickup cartridges arranged at right angles to each other. I will go into greater detail about the quad microphone when I discuss multichannel recording.

The A-B Arrangement

Figure 1 shows an A-B arrangement. Two omnidirectional or unidirectional microphones are placed in parallel with a usual spacing of about 20″ although the spacing is not too critical as I'll later demonstrate with some recorded sound files. The spacing is not an independent variable because the stereo image also depends on the distance and direction between the microphones and the sound source. The microphones should be matched in amplitude and frequency response, but this is usually accomplished by using microphones of the same make and model.

Stereo file **helicopterA2.wav** was recorded with a 24″ spacing between two DAK UEM-83R unidirectional electret condenser microphones. I increased the spacing to 48″ for file **helicopterA4.wav**. The "flight path" was side-to-side parallel to the microphone fronts and about 15′ away. In both files, you can clearly hear the sound passing from left to right.

Photo 1 shows the measurement setup. A detailed description of how I recorded the files as well as the files to this and the articles that will follow in this series can be found in the Supplemental Material section on the *audioXpress* website (www.audioxpress.com).

I used a pair of EM-8800 unidirectional electret condenser microphones spaced 24″ apart to repeat this test. These microphones have a narrowed cardioid pattern and tend toward the so-called "shotgun" pattern. For this test, the path was again side-to-side and about 6′ in front of the microphones. The resulting file is called **helicopterB2.wav**

The ORTF arrangement was devised in about 1960 by the Office de Radiodiffusion Télévision Française (ORTF) at Radio France. I tried it with the microphones spread out at 110°. **Photo 2** shows this variation of the A-B arrangement. It spreads the effective microphone spacing without increasing their physical separation. This may have some practical advantage because only one microphone stand or one set of suspension wires would be needed. Cardioid microphones are typically used, although omnidirectional microphones are sometimes employed.

The X-Y Arrangement

Figure 2 shows the X-Y arrangement, which is commonly used with cardioid microphones because of the convenience in mounting or suspending them.

Photo 1: This setup for stereo recording uses two DAK UEM-83R microphones spaced 24″ apart. The cube with the speaker is my portable helicopter, which is fully described in the Supplemental Material.

Photo 2: The ORTF microphone arrangement uses an "over-under" pattern on a single tripod.

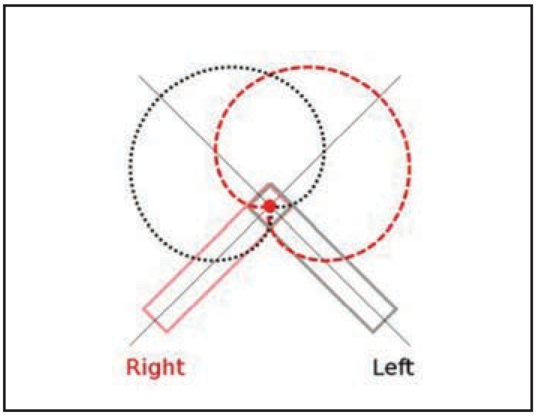


Figure 2: This drawing shows the X-Y microphone placement. The angle between the microphones is usually 90°. They are typically mounted one above the other (see Photo 2).

Photo 3: I used a a Behringer EMC8000 cardioid for the mid microphone and an Apex 415 set to figure-eight (it has a switch that also enables you to chose omnidirectional or cardioid) for the side microphone in this Mid-Side setup.

Project File

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

Resources

M. Nymand, "Directional vs. Omnidirectional Microphones," (a Microphone University file), www.dpamicrophones.com.

Sources

Cantabile Solo
Cantabile Software | www.cantabilesoftware.com

REAPER
Cockos, Inc. | www.cockos.com/reaper

Apex 415 microphone
FrontEnd Audio | http://apexelectronics.com

Behringer EMC8000 microphone
Parts Express | www.parts-express.com

Cubase software
Steinberg Media Technologies GmbH | www.steinberg.net

DR-05 Linear PCM Recorder
Tascam | www.tascam.com

Model 411 Microphone preamplifier
TDL Technology, Inc. | www.tdl-tech.com

MSED free VST plug-in
Voxengo | www.voxengo.com

However, the stereo image is poor compared to the A-B arrangement because there is no time of arrival difference. The only difference is the loudness of the sound in one channel compared to the other. This microphone placement was probably used when you hear an instrument's sound "wander" around in the stereo field. I have often seen this placement used to record a symphony orchestra because it's easy to do. However, the stereo recording would be improved with a better arrangement.

The Mid-Side Arrangement

Photo 3 shows the Mid-Side arrangement, which is typically used to record the middle and side channels rather than the left and right channels. The reason for this pattern is that it enables engineers to control the width of the stereo image in post-processing with a Digital Audio Workstation (DAW) or using hardware.

The Mid microphone is usually a cardioid, although an omnidirectional is sometimes used. Its output is just the mono output from the source. However, the Side microphone must have a bidirectional (figure-eight) pattern with the microphone front facing to the left so the microphone rear faces right. This pattern places the two sides 180° out of phase but they are added together in the microphone so there is just one output signal.

The Side output is really stereo but when the sound source is directly in front of the microphone pair, it is facing the Side microphone's side. This means the right and left pickups are at least nearly equal and the output sums to about zero. Therefore, the Mid signal is needed to reconstruct the stereo image and the Side channel must be phase inverted to create a –Side signal:

$$\text{Left out} = \text{Mid} + (+\text{Side})$$
$$\text{Right out} = \text{Mid} + (-\text{Side})$$

The stereo width can be controlled by how much Side signals are added to the Mid signals.

Alan Blumlein devised this microphone placement and patented it in 1933. **Photo 3** shows the Apex 415 figure-eight microphone (Side) with a Behringer EMC8000 cardioid (Mid) that I used, along with a pair of TDL Model 411 microphone preamplifiers and a Tascam DR-05 to record the sample provided in **helicopterC.wav**. This is a stereo file with Mid in the right channel and Side in the left channel.

The Cantabile Solo, a DAW from Australia, and Voxengo's MSED, a free VST plug-in, MSED, provide an inexpensive way to extract stereo and explore image width changes. The procedure for Cantabile Solo and

Sony Audio Studio is included in the Supplemental Material. Other DAWs can be freely downloaded. REAPER, another inexpensive DAW from Australia, is also a good choice.

There are many opinions about stereo microphone placement, which shows that a "good" arrangement depends on a variety of factors (e.g., the size and shape of the venue, the type of sound (music) to be recorded, the type of microphones used, and maybe a few other considerations).

All my test files demonstrate a stereo sound field, some more than others. My point is that you should experiment to find out what works best. Although the recorded sound can usually be improved in post-processing, it's better to start with the best recording you can make.

There are other microphone placements for multichannel recording, which I will describe in future articles in this series.

However, I have a few more comments about recording. Many people are still in love with analog recording to magnetic tape for its presumed "warmth" and generally pleasant sound. But even with just two tracks on a 0.5" wide tape running at 15 inches per second (ips), the frequency response is not especially flat and the SNR is about 65 dB at best—and these machines are expensive.

Digital recording is much less expensive. Its frequency response depends on the microphone and preamp, not on the recorder. Using a 24-bit depth and a high-sampling frequency, the fidelity is quite good and the SNR can be 85 dB or better. The Tascam recorder I mentioned earlier records stereo .wav files at 44.1, 48 or 96 kilosamples per second (ksps) with a 16- or 24-bit depth using the built-in or external microphones and with a SNR of 92 dB or better. All this performance is available in a hand-held and tripod mount unit for less than $100. There are also many other digital recorders in every price range.

About the Author

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957, he started Testronic Development Laboratory (now TDL Technology) to develop audio electronics. He is still the TDL president and principal designer.

Nordic Audio Tales (Part 4)

Morel—Nordic Influence Thrives in Israel

The next article in our Nordic series explains how loudspeaker influences from this northern region appear in Morel's designs. Due to its Nordic roots, the Israel-based Morel is highlighted in this series.

By
Mike Klasco
(United States)

A common element among most Danish speaker companies is Ejvind Skaaning. According to bits and pieces from noted speaker engineer Claus Futtrup's online blog, "Mr. Ejvind Skaaning was born on April 17 1929, in the small town of Vejen, Denmark. Recording, audio, and speakers were hobbies but by the mid 1960s, Mr. Skaaning focused on starting his loudspeaker business. In 1968, Skaaning was the subcontractor to Scandyna for the famous Dynaco A-25 speakers using SEAS drivers. Soon after in 1970 Scanspeak was founded and Skaaning was in charge of operations and transducer development."

RMS

By 1974, Harman had acquired Ortofon and liquidated its Scan-Speak group. Scan-Speak was to

Meir Mordechai (a) and Ejvind Skaaning (b) sign the first woofer made at the Morel plant in 1979. (Photo courtesy of Morel) be easily dismissed. But at the same time, Skaaning entered a joint-venture with partners Gerhard Richter and Meir Mordechai to form RMS (which stood for Richter, Mordechai, and Skanning). Based in Israel, the partners wanted to develop next-generation concepts in drive unit construction. In 1979, RMS started manufacturing raw drivers (at what is now Morel's plant) in Ness-Ziona, which is a suburb of Tel Aviv, Israel. The company, which has become Morel, still resides at the original site. A freeway, now in its final construction stages, will soon be adjacent to the factory.

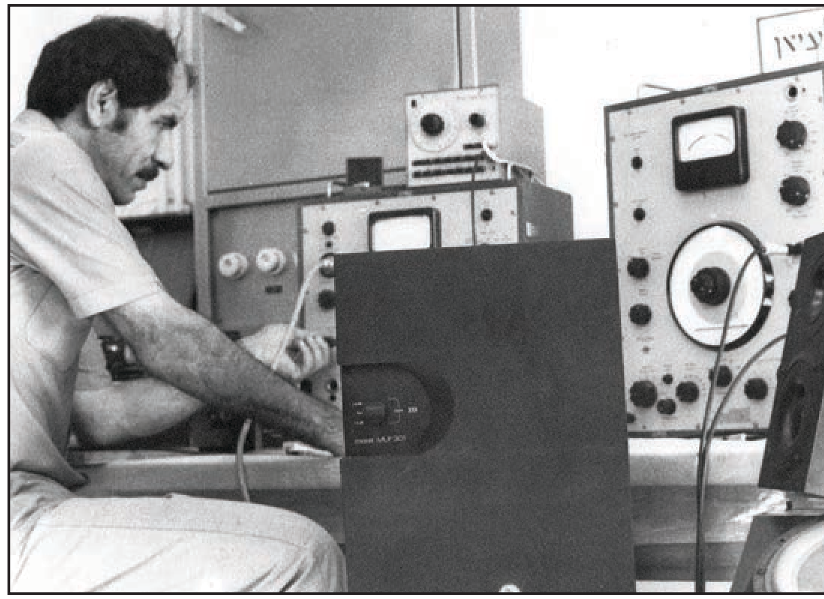
Following the RMS partnership, the Middle East became progressively unstable and the future seemed bleak especially in Lebanon where Skaaning had heavily invested. With the civil war in Lebanon and Skaaning's other Lebanese investments at risk, he was ready to cash out and go home. Mordechai, an Israeli, bought Skaaning's (and Richter's) shares after two years in business. Skaaning returned to Denmark with his Lebanese wife.

Skaaning and Richter (a successful German speaker distributor) remained partners and took the same design approach they had used with the RMS product line (which was also integrated into Morel's operation) and launched Dynaudio in Denmark. This perhaps solves the mystery as to why Morel and Dynaudio have a common design philosophy.

Morel's Beginnings

Mordechai called the former Danish-style speaker company, Morel. After the split, relations were good among the former partners. At the beginning, Dynaudio even supplied some components for a couple of Morel's loudspeaker models. In 1982, Morel invented the renowned double magnet system for its high-power 3" diameter range of drivers.

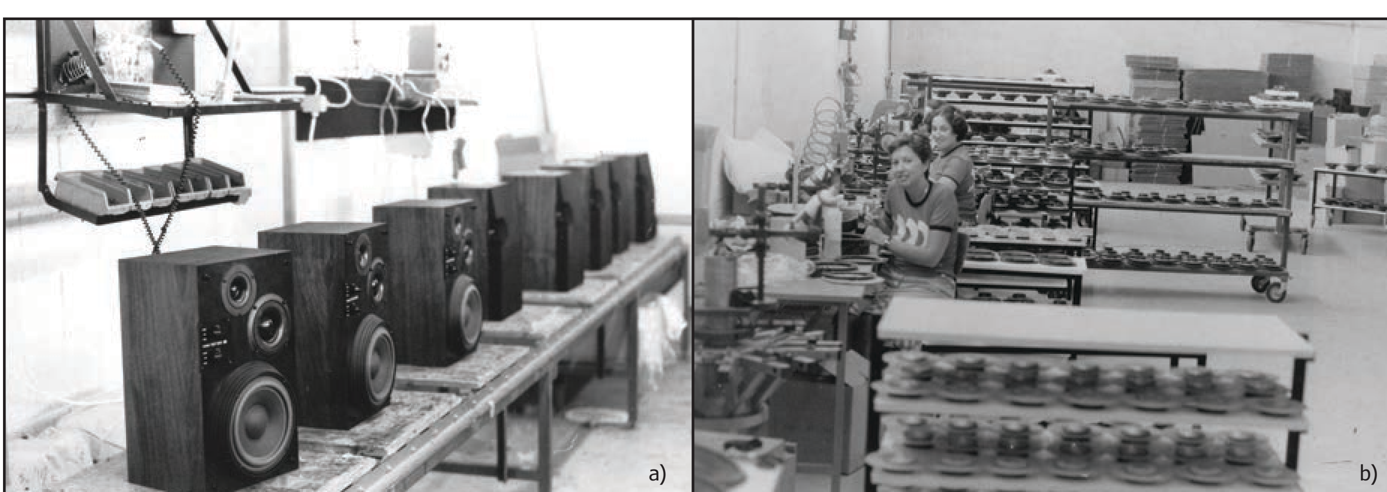
In the mid 1980s, Morel briefly opened an additional factory in the UK. Mordechai did this to deal with the trade laws and practices of the European Community (EC), which hindered any export into Europe. In the mid 1990s, Israel became a member of the EC trade agreements, the trade laws were changed, and the Israeli and the UK facilities were unified into the one manufacturing facility in Israel.



Meir Mordechai tests speakers at the Morel lab in 1980. (Photo courtesy of Morel)

Almost 40 years later, Meir Mordechai still watches over Morel. His son Oren (now 40) manages the factory. Nir Paz manages the company's global sales and U.S. operation. Morel remains successful partially because the company has continued its legacy of innovation with a range of products that incorporate small but powerful and creative motor systems inside a rather large voice coil for a given speaker size.

Internationally, Morel is best known for some very over-the-top drivers in the aftermarket car industry. Its Ultimo Titanium 10" subwoofer has won the prestigious European Imaging and Sound Association (EISA) 2014-2015 Best Product award.



Morel's speaker assembly line (a) and production floor (b) are pictured in 1980. (Photo courtesy of Morel)

Meir Mordechai continues to oversee the company that he started almost 40 years ago. (Photo courtesy of Morel)

Meir Mordechai is pictured with his son Oren, who now manages the factory. (Photo courtesy of Morel)

Morel still uses in-house voice coil winding. (Photo courtesy of Morel)

Morel's Fat Lady tower speakers are favored by audio aficionados. Morel's OEM driver strengths are that they are made-in Israel with the technology, service, and communication that come with it. The drivers are also aggressively priced, which make them unprecedented when compared to similar-grade drivers from Europe. As for communications, almost all Israelis are fluent in English.

Morel's work-cell approach to fabrication lends itself to customization on all drivers, even with small batch runs. And, the company can provide OEMs and ODMs complete turnkey solutions for drivers, integrated electronics, and enclosures. OEM and ODM customers that use Morel's drivers include MACROM, Xtant, DLS, Phoenix Gold, Eggelestone Works, Rega, Ruark, Quested, HHB, Leon Speakers, CAT, CAV, NIro Nachamichi, and others. Morel also has a long-established sales and warehouse facility in New York.

The Factory

Morel is a boutique operation with more than 40 employees. Separate engineering areas, the acoustics lab, and the production floor are well organized and operate to a high standard. On a recent visit, special speaker construction and quality techniques were used throughout the production facility.

Morel makes most of its parts, using in-house voice coil winding. The company also locally sources items (e.g., the deep-drawn copper caps that attach to the pole pieces to null eddy currents and linearize phase response and the unique one-piece carbon fiber cones) from other high-tech companies.

Morel's "DNA"

Loudspeakers have a signature sound as well as a signature construction. The European approach to ultra-low distortion linearity is evident in the design and construction of Morel speakers.

Morel's motto is "Sound that captures musical magic." Morel translates that existential thought into tangible technological developments and precise time-domain performance, optimizing frequency response and reducing distortion while remaining linear at large signal levels. Time-domain performance comprises impulse response rise time and settling time, as well as phase/group delay linearity and latency.

The fidelity and integrity of what goes into building the loudspeakers impact the outcome. Morel's high standards guide the selection and design of new materials and geometries for cones, tweeter diaphragms, magnetic circuits, and everything in between. The unique design techniques found in the Ultimate subwoofer, the Titanium former woofers, and the Elite and Supreme series differentiate Morel's speakers from commodity speakers in several ways.

Morel loudspeakers offer superior linear phase/group delay characteristics. These characteristics are achieved through eddy current canceling topology that reduces inductive rise and lowers total harmonic distortion (THD) and intermodulation distortion (IMD). Copper caps (some with extended skirts deep into the gap) null eddy current modulation and IMD even at higher voice coil excursions.

Linear excursion Thiele-Small (T-S) parameters are stable due to the voice coil position. "Over-motored" designs deliver superior electrodynamic damping, achieved through high BL in the motor using strong unique magnetic-structure topology and large-diameter voice coils (e.g., a 3" diameter voice coil in a 6.5" speaker and a 5" diameter voice coil in a 10" woofer).

The open-cast aluminum chassis with below spider venting is anechoic (no reflection from the frame of the backwave comes back through the diaphragm) and it offers increased heat damping for reduced power compression. Also, Morel developed its own composite cone assemblies that combine art and science to suppress diaphragm (cone) breakup and minimize standing waves and reflections, using simulations, treatments, secondary operations, and esoteric materials.

The proliferation of flat-screen displays and TVs has started an OEM race to create a matching thin-depth home theater audio system. It seems that many OEMs have traded true quality hi-fi sound performance for size. The fact is that big deep speakers are a prerequisite for a high-performance sound system, especially as we approach the low end. Morel raised to the challenge with a new Slim woofer line that will appeal to OEM manufactures seeking a slim design with quality hi-fi sound performance.

Getting There

The Morel factory is in Ness Zona, just 20 minutes outside Tel Aviv, which is the business capital of Israel. From the US, you can fly from New York (JFK), but we flew non-stop on Norwegian Air from Copenhagen Denmark to Tel Aviv, which took about four hours. In addition to the usual international hotels that line the beaches, there are many charming boutique hotels in Tel Aviv.

Morel's Future

The near-term future looks very exciting for Morel, which is launching new driver product lines, including the Titanium series (Titanium bobbins), super shallow speakers, and a beryllium tweeter. Morel is also expected to launch its first high-end soundbar at the 2015 Integrated Systems Europe Show (ISE), which will be February 10-12, in Amsterdam Rai, NL.

The TiCW1258Ft is Morel's latest Titanium subwoofer, equipped with a 5.1" voice coil. (Photo courtesy of Morel)

The TSM634 is Morel's newest Titanium midrange loudspeaker, equipped with 3" voice coil. (Photo courtesy of Morel)

Morel now offers a new power slim woofer line for OEM manufactures seeking a thin design with quality hi-fi sound performance. The SMW642 is a 6" woofer, equipped with a 4-Ω, 2" voice coil. (Photo courtesy of Morel)

Resources

V. Dickason, "Morel Acoustics," *Voice Coil*, October 2009.

Morel, Ltd., www.morelhifi.com.

Fresh From the Bench

Universal Audio Apollo Twin

Small Things Make a Big Difference

By
Oliver A. Masciarotte
(United States)

One-knob boxes have become a common sight in musical instrument (MI) and semi–professional circles. Usually, these are simplistic products built to appeal to a less than discriminating consumer. Universal Audio, long known for its high-quality outboard gear, has adopted the “Big Knob”–on–a–sled form factor, while providing the new Apollo Twin family with features and sonics that are missing from the competition. This is not just another pretty (inter)face.

The rugged and compact Universal Audio Apollo Twin cabinet is dominated by a large, centrally located knob. The knob is augmented by a range of clearly labeled soft switches and visual mode annunciators. Two models—the Apollo Twin SOLO and the Apollo Twin DUO—are available. The name indicates the number of Analog Devices’ fourth-generation SHARC processors—the ADSP-21469KBCZ4-02—that are available for on–board processing.

Apollo Twin Hardware

The Apollo Twin family consists of two input and six output desktop audio interfaces with up to 24-bit/192-kHz audio conversion. As with all Apollo hardware, real-time accelerated processing is available for tracking through UAD Powered Plug-Ins with very low latency. In a bold move, Universal Audio has chosen to only use Thunderbolt as its host bus attachment, for exceptional throughput and satisfyingly solid performance on modern desktop and portable Macs.

The two inputs can be switched between the line and the microphone, with Unison technology offering true hardware emulation for the digitally controlled microphone preamps. Two line outputs and two digitally controlled, independently addressable analog monitor outputs are also included, as is a front panel Hi-Z 0.25” unbalanced input and an independently addressable headphone output. Up to eight channels of additional digital input are available via a standard ADAT F05 connector. The plastic optical fiber connection also supports S/PDIF optical with sample rate conversion when needed for operation at 4x sample rates. UAD Powered Plug-Ins, in AU, AAX64, VST, and RTAS are included for use in our favorite digital audio workstation (DAW) and the 64-bit device drivers support Peripheral Component Interconnect Express (PCIe) audio protocols.

DSP is used for all UAD Powered Plug-Ins. Universal Audio's Technical Marketing Manager Gannon Kashiwa explained that, "All our plug-ins run on our DSPs. That includes plug-ins instantiated in our DAW. The DAW sends the audio to us (the Apollo hardware) for processing, but…(the DAW thinks) we're native to their platform."

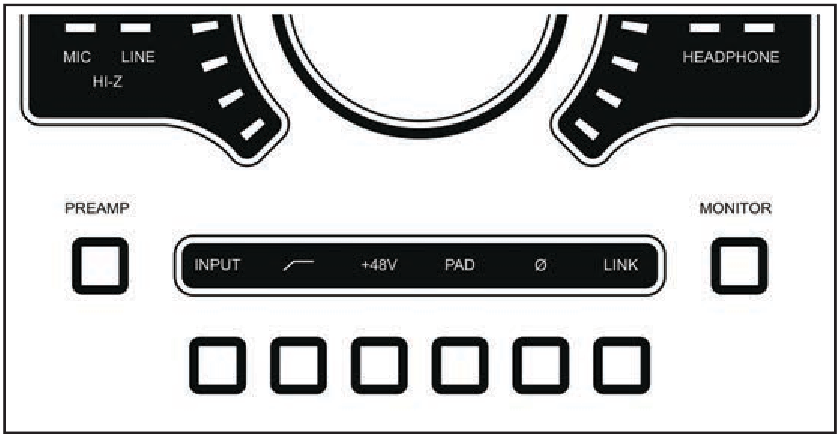
Another unique feature of the Apollo line, the Unison technology, replicates the behavior of physical microphone preamplifiers. "The impedance and gain staging are the physical components that are automatically changed when you insert a Unison-enabled preamp plug-in," Kashiwa said. This means that a microphone "sees" the actual load that corresponds to the hardware model that the plug-in represents. Other parameters—including a high-pass filter, a pad, and a polarity invert—can also be controlled by either the plug-in interface or on the Apollo hardware itself.

The centrally placed "Level Knob & Switch" (aka the Big Knob) controls multiple functions. The knob's function is selected with either of the flanking Preamp or Monitor buttons. When Monitor is selected, pressing the knob mutes or unmutes the monitor outputs. When Preamp is selected, pressing the knob alternates between Channel 1 and Channel 2. For both roles, twisting the knob provides gain adjustment.

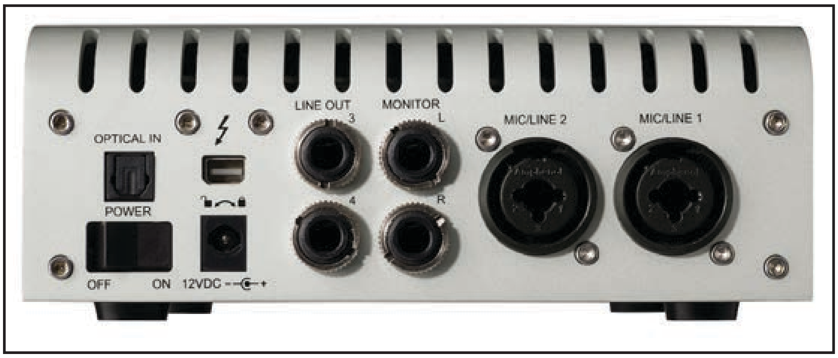
Below the Big Knob are six latching soft buttons. The Input Select alternates between the microphone and the line inputs, with the current selection displayed to the left of the Big Knob. The Hi-Z input is selected automatically whenever a mono 0.25" plug is inserted into the front panel's instrument jack. That input is hard wired to show up on Channel 1. The low-cut filter is a second-order high-pass filter, with a 75-Hz resonant frequency. The third button engages phantom power for the selected input, while the pad provides 20 dB of attenuation only for the microphone preamps. The Link button groups Input 1 and Input 2 so both channels are simultaneously adjusted as a stereo pair.

Apollo Twin Software

Apollo Twins are supplied with a software application, the Console, and a utility, the UAD Meter and Control Panel. Both software pieces are accessible from the dock once installation is complete. Software installation is simple and



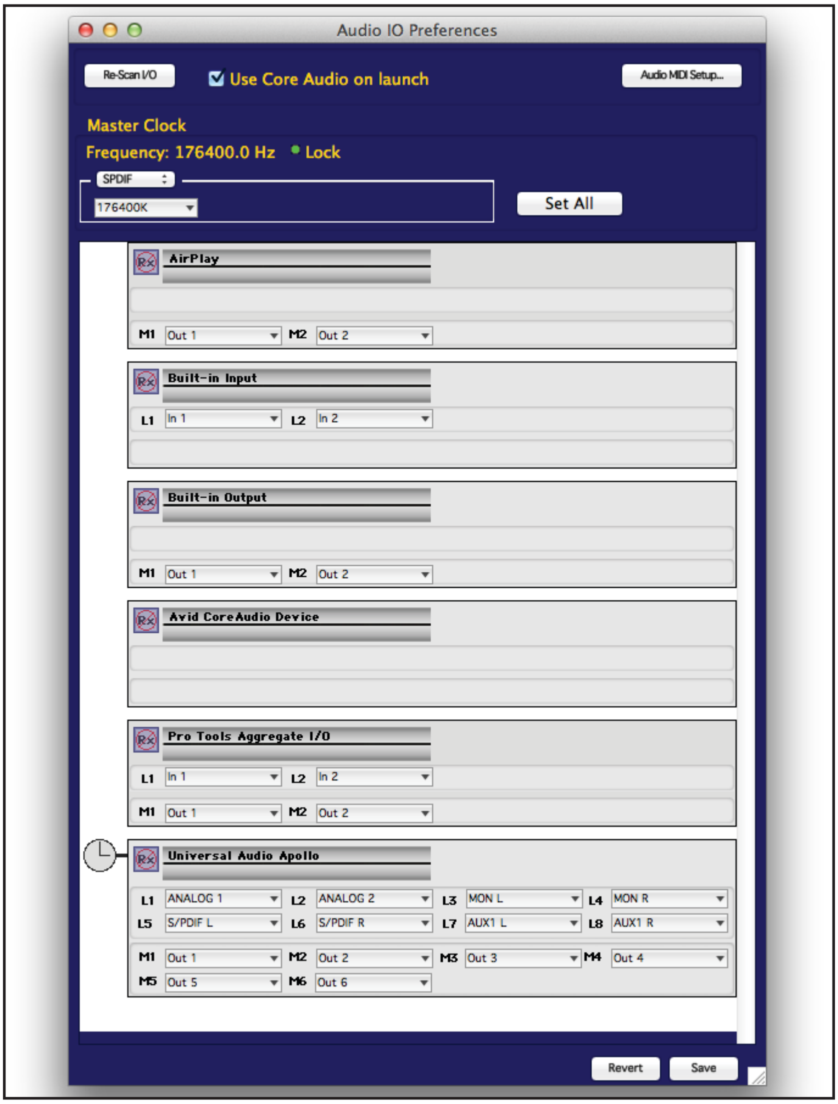
The Universal Audio Apollo Twin provides easy-to-use Preamp and Monitor buttons as well as six Preamp Option buttons that enables users to choose Input Select, Low-Cut Filter, 48 V, Pad, Polarity, and Link.



Apollo Twin offers two microphone inputs and two analog line inputs, one Hi-Z instrument input, two analog line outputs (independent mix buses) and two analog monitor outputs, headphone output, TOSLINK optical (ADAT or S/PDIF, selectable) and one Thunderbolt port (Thunderbolt 2). It also has a bayonet-style locking connector to prevent accidental disconnection of the included wall wart power supply.

Apollo Twin includes a real-time Analog Classics Bundle that contains UAD Powered Plug-Ins to track and mix analog emulations, including guitar and bass amp emulations from Softube, as well as Universal Audio's 610 Tube Preamp and EQ plug-in.

A valuable feature of the Apollo Twin family is its ability to add separate processing to foldback while printing a take dry and with effects.



Apollo Twin provides several I/O options, plus plug-ins linking to your DAW environment.

straightforward. The modal Control Panel monitors hardware resources, visually confirms that the UAD system is properly operating, and enables users to configure the global UAD Powered Plug-ins system parameters.

The Console is the main software interface for Apollo hardware. It is used to remotely control the hardware's mixing and monitoring functions, including the sample rate, the clock source, and the reference levels. Console replaces the DAW's software monitoring feature. The Apollo hardware's extremely low latency, less than 2 ms, in conjunction with its ability to provide reverb and EQ to the headphone mix, makes for comfortable foldback. The Console's analog-style mixer enables real-time, DSP–accelerated processing using UAD Powered Plug-Ins to be configured and operated. UAD Powered Plug-Ins can be inserted into all Console inputs and/or auxiliary returns while monitoring and/or tracking. All processed or unprocessed mix buses, including the monitor, the auxiliary, the headphone, and the cue buses can be optionally routed to a DAW for recording.

Console can be simultaneously used with a DAW for front-end processing and monitoring functionality. Complete setups can be saved as presets for easy recall of the entire configuration. A Console Recall plug-in, in the previously mentioned four formats, is included to conveniently store Console configurations within DAW project files. Console has two pre/post stereo auxiliary buses, with independent send levels per input, for grouped signal processing, which conserves DSP resources. It can also be used when routing to alternate hardware outputs. Using Console, the headphone mix bus can be mirrored to any hardware output. The straightforward layout and clear labeling makes it easy to use.

The Apollo family includes permanent licenses for the Real-time Analog Classics package, which consist of a UA 610-B tube Preamp & EQ, the Softube Amplifier Room Essentials, the 1176SE/LN Classic Limiting Amplifiers (Legacy version), the Pultec Pro Equalizers (Legacy version), the Teletronix LA-2A Classic Leveling Amplifier (Legacy version), the CS-1

About the Author

Oliver. A. Masciarotte is a graduate of the Lowell Institute of MIT. Oliver has been a principal of Seneschal, a consultancy to the professional audio and rich media industries, for more than 25 years. Oliver is also the author of more than 100 trade articles and he wrote *To Serve & Groove*, a book about file-based music playback for the home.

No Limits for Thunderbolt

Making its debut at the 2009 Intel Developer Forum alongside the release of the Nehalem microarchitecture and Core i5, the then-new "Light Peak" technology was demonstrated to me on prototype "mad scientist" hardware. At the time, the engineering team was able to run parallel 1080p streams plus Ethernet and storage protocols over a 30-m glass fiber. That technology was later rebranded as Thunderbolt and, to reduce costs during commercialization, an electrical copper PHY was adopted. Today's Thunderbolt offers 10-Gbps bidirectional throughput on separate data and display channels, enabling you to move 1 TB of data in less than 5 mins. Up to six devices can be daisy chained.

Ready for "4K" UHDTV video, the next generation Thunderbolt 2 runs at 20 Gbps. With DisplayPort 1.2 support, Thunderbolt 2 enables video streaming to a 4K monitor. Thunderbolt's advantages are legion—an ultra-high speed throughput, support for copper and fiber, plus backward compatibility with PCI Express, DisplayPort, FireWire, HDMI, DVI, and Gigabit Ethernet.

Old School FireWire reliably extends only 3 m. In 2013, Thunderbolt went back to its roots with several vendors introducing optical cables. Corning, which is best known for its Gorilla Glass, manufactures 5.5-, 10-, and 30-m Thunderbolt Optical Cables by Corning. Corning has also announced a new 60-m length. These widely available interconnects, ready for Thunderbolt 2, employ active circuitry in the connectors to transform electrical signaling to modulated light and back again. Very lightweight, thin, tough, and highly pliable optical cables are perfect for moving bulky data around a facility. Because the multi-mode glass fiber is optimized for the specific application, jitter and signal integrity are not issues, even over relatively long distances.

In addition to Corning's generous contribution to this review, the folks at Belkin supplied a Thunderbolt Express Dock, their answer to every modern Mac-equipped studio's dreams. This slim aluminum sled sports a Thunderbolt connection to your computer, another Thunderbolt port to daisy chain up to five additional devices, a Gigabit Ethernet port, a FireWire 800 port, a 3.5-mm headphone jack, a 3.5-mm analog audio input jack, and three powered (500 mA) USB 3.0 ports. It also includes a 40" Thunderbolt cable. In short, every possible spigot you could require, located next to the host or remotely via a Thunderbolt Optical Cable by Corning.

Thunderbolt Optical Cable by Corning is now available in a 60-m length.

Precision Channel, and the RealVerb Pro.

Each input and auxiliary return can have four plug-in instances. The only constraint is that the signal flow is always serial. The insert processing can be monitored "wet," while the recording can be wet or dry. Sub-2 ms round-trip latency through four serial non-upsampled UAD plug-ins at 88.2 kHz means foldback with effects and real-time performance are not a problem.

When asked about DSP expansion, Kashiwa mentioned that, "…you can add PCIe cards via a Thunderbolt PCIe chassis, an existing FireWire Satellite via the Apple Thunderbolt–to–FireWire Adapter or… one of the latest UAD-2 Satellite Thunderbolt external units. (Note: This was announced while the review was in the works—see "New UAD-2 Satellite Thunderbolt DSP Accelerators" sidebar.) Any of those options give you more DSP for mixing (in your DAW). These DSPs are not used for real-time processing. It's possible to have four UAD devices in a single system so, for example, you could max it out at three OCTOs and a Twin for a total of 26 DSP chips," which is a lot of DSP horsepower.

As the Apollo family is meant to be an entry level piece, they are not very expandable when it comes to I/O. Kashiwa suggests that, "you can add up to eight (I/O channels) with ADAT, and that's about it. One cool combo we like is the UA 4-780d, a four-channel 'Tone-Blending' mic preamp with dynamics, in combination with the Twin. (That combination provides)…some great tube front end and four additional line inputs that get converted and passed through."

Apollo Twin Components

Conversion components and clocking are what makes a converter sound the way it does.

New UAD-2 Satellite Thunderbolt DSP Accelerators

By João Martins

At some point, the market believed external DSP accelerators would be surpassed by native (multiple-core) central processing unit (CPU) resources on PC and Mac workstations. This is why most manufacturers decided to discontinue that strategy. Universal Audio was the only company that remained in this domain and proved how audio recording hardware and software could continue to exponentially be improved by add-on DSP resources due to the evolution in faster communication interfaces (e.g., Intel and Apple's Thunderbolt technology).

Recently, Universal Audio announced a new line of UAD-2 Satellite Thunderbolt DSP Accelerators as a sleek, powerful way for Thunderbolt-equipped Mac users to "supercharge" their systems and run larger mixes filled with rich, DSP-intensive plug-ins. DSP accelerators combined with newly optimized ultra-low-latency plug-ins is proving to be the future of studio outboard with all the flexibility and convenience of software integration.

The new desktop-friendly fan-free units provide full access to UAD Powered Plug-Ins, including exclusive plug-in emulations from Studer, Ampex, Lexicon, Neve, Manley, SSL, Roland, EMT, Empirical Labs, MXR, and more. Available in QUAD or OCTO models with a choice of four or eight SHARC processors, UAD-2 Satellite Thunderbolt DSP Accelerators can also be integrated alongside UAD-2 PCIe cards and Thunderbolt-enabled Apollo interfaces, including Apollo Twin, Apollo, and Apollo 16—for truly scalable mixing power.

Universal Audio's new line of UAD-2 Satellite Thunderbolt DSP Accelerators shipped in early October.

The new UAD-2 Satellite Thunderbolt's key features include support for UAD Powered Plug-Ins via Thunderbolt or a Thunderbolt 2 connection on Macs for improved performance at higher sample rates, and reduced plug-in latency vs. FireWire, with 8 (OCTO) SHARC processors providing a massive DSP boost for running large professional mixes.

Universal Audio
www.uaudio.com

Adam Tucker at Signaturetone Recording (a) emphasized the simplicity of the Apollo Twin's setup and routing (b). The combination of small form factor and Corning's optical Thunderbolt connectivity make for easy, optimal placement.

Apollo Twin uses the "premium," differential input monolithic AKM AK5388EQ for analog-to-digital conversion. Asahi Kasei Microdevices specifies the part as having a short group delay, with an anti–aliasing filter that "…features a modified FIR architecture (that) minimizes group delay while maintaining excellent linear phase response." Group delay produces frequency–dependent delay or what the unwashed masses think of as "time smear."

Finite impulse response (FIR) filters are an architecture not easily implemented in the analog world. Unlike active analog EQ, where an RC or frequency selective network is placed in the feedback loop of a differential amplifier, a FIR filter is a feedforward design. No feedback means no ringing, a fast settling time, and a linear phase response.

As to clocking, the engineers at Universal Audio wouldn't reveal too much about their solution but they did allude to their use of JetPLL. According to TC Applied Technologies, JetPLL is a patented solution for audio band jitter attenuation, with a hybrid design that incorporates noise shaping to "virtually remove audio band jitter."

Digital-to-analog conversion on the monitor and line outs are performed by Cirrus Logic's "flagship" CS4398. According to Cirrus, this part includes "half dB step-size volume control…(and) selectable fast and slow roll off digital interpolation filters followed by an oversampled multibit Delta-Sigma modulator, which includes mismatch shaping technology that eliminates distortion due to capacitor mismatch. Following this stage is a multi-element switched capacitor stage and low-pass filter with differential analog outputs."

It's a pity Universal Audio didn't provide software control for the anti-imaging filter topologies.

For the headphone output, Universal Audio chose the quotidian, differential output AKM AK4480EF. Though the Twin's headphone out is no match for more costly converters, it provides good quality playback right where it's needed most, and helps keep the total cost down.

In Use

The Apollo Twin is one of those solidly built boxes that reliably does what you expect, without the annoying implementation and plethora of questionable features found in many MI offerings. The sound quality is excellent for the price, and the UAD Powered Plug-Ins are peerless. It seems like a trivial thing, but all the engineers who logged some time with the box appreciated the monitor mute switch built into the Big Knob. Another small but significant feature is the power connector. The included wall wart power supply has a bayonet-style locking connector to prevent accidental disconnection. When asked to weigh in on the design of the power supply unit (PSU), Gannon Kashiwa opined that an external power supply was "absolutely" required due to space considerations. "The inside of Apollo Twin is packed and heat is a huge consideration. The DC power from the external supply is conditioned inside the box and gets regulated and cleaned to provide a stable, noise-free source."

Another aspect of the Apollo Twin that met with approval was the product's uncomplicated nature. Engineers Adam Tucker at Signaturetone Recording and August Ogren at Humans Win! appreciated the simplicity of the setup and the routing, which enabled everyone to obtain excellent sounds and quickly move forward with the recording sessions. With another nod to the design team's expertise, the preamps overload surprisingly gracefully, bringing fatness rather than harshness to more energetic musical moments. In all, an outstanding entry level interface, with a deep, innovative Thunderbolt implementation, and excellent tracking and mixing capabilities.

After spending a good bit of time with the product, only one flaw came to light. Although the Plug-ins tab in the supplied UAD Control Panel indicates which plug-ins are currently licensed and available for use, the Console's plug-in selection menu doesn't. Also, that the Console's plug-in menu does not allow grouping of available plug-ins by type, making it painful to select a particular class of effect or virtual gear. Universal Audio is aware of these shortcomings and, hopefully, will address them in a future software release.

Sources

AKM AK5388EQ and AKM AK4480EF ADCs
Asahi Kasei Microdevices, Corp. | www.akm.com

Thunderbolt optical cables
Corning, Inc. | www.corning.com

CS4398 Stereo DAC
Cirrus Logic Inc. | www.cirrus.com

Apollo Twin
Universal Audio | www.uaudio.com

Voice Processing for Mobile Audio

Today, people take their smartphones with them everywhere they go, and they expect to be able to communicate with those on the other end of the line regardless of their environment. This article explores the co-processors typically used in smartphones to filter out wind noise and crowd babble, enhance voice intelligibility, and cancel acoustic echo.

By
Mike Klasco and Steve Tatarunis
(United States)

Mobile audio products are, in many ways, even more challenging to design than audio gear. The engineering considerations are substantially different with portable consumer electronics (e.g., smartphones, tablets, laptops, Bluetooth hands-free over-the-ear, and Bluetooth fitness and wearable audio products) presenting unique performance challenges.

Special considerations in mobile audio design include physical robustness, environmental resistance (e.g., moisture, dust, dirt, etc.), low power consumption and miniaturization while still achieving the main goal—great sound.

For voice communication products, sound clarity is the highest priority. The goal is to provide clearly intelligible speech in noisy environments. With the uplink (i.e., what you say) and the downlink (i.e., what you hear from the other side), users expect to communicate without intrusive noise, which is not an easy task in the noisy world we live.

Yet, there is an expectation for clear phone communication while walking down a busy street in NYC, driving a car with the window open, at a windy beach, or in the din of a crowded restaurant **(see Photo 1)**. All these examples present challenging environments for even the most sophisticated state-of-the-art signal-processing algorithms.

This month, we examine voice processing for mobile audio products (e.g., smartphones, tablets, laptops, Bluetooth, and wearable gear). This processing may take place in a smartphone's voice co-processor, in a Bluetooth's earphone, in the DSP core section of the Bluetooth IC, in a tablet's ARM processor, or in a smart-codec's DSP core. This type of signal processing can also be found in boardroom conference phones and so forth.

Voice processing improves voice quality and speech recognition performance by reducing noise, using advanced processing features such as adaptive noise filtering, voice intelligibility processing, and echo cancellation. Smartphones typically use voice co-processors to handle wind noise, crowd babble, voice intelligibility enhancement, and acoustic echo cancellation (AEC).

Co-Processors

The IBM PC AT (from the 1980s) had an empty socket for the Intel 8087 chip, a floating-point math co-processor designed to boost its Intel 8086 CPU's performance when running calculation-heavy CAD or graphical applications (see **Photo 2**). A co-processor's function is to assist the primary processor with specific tasks—increasing performance or freeing up processing power that can be used for something else.

Dedicated graphics chips or graphics processing units (GPUs) started as co-processors, taking the heavy demands of visual processing away from the central processing unit (CPU). Also, sound cards can be considered a class of co-processors that add higher audio quality, multichannel audio playback, and real-time audio DSP effects. Now, most smartphones use dedicated integrated circuit (IC) voice co-processors that are a type of application specific integrated circuit (ASIC).

In the most sophisticated adaptive processors, intelligent speech tracking and adaptive processing analyze ongoing speech patterns and environmental conditions in real time. The result is a self-adjusting signal-dependent algorithm that emphasizes voice quality—whether in a quiet room or the noisiest vehicle using techniques such as automatic level control and automatic volume control.

Noise Suppression

Crowd noise, audience babble, and similarly variable and unpredictable complex noise can significantly degrade speech intelligibility but only the most sophisticated voice processors can effectively separate and filter this type of random noise from the speech signal.

Mobile audio products tackle wind noise suppression through acoustics and signal processing. Acoustic suppression of wind noise is achieved through the use of mesh wind screens, which is the same materials used on professional stage microphones. These materials are designed to be acoustically transparent while blocking the wind's force before it can overload the microphone's output signal. Once the microphone diaphragm is hit by the wind pressure, the signal processing can only do so much to help. Many mesh materials also have high international or ingress protection (IP) marking ratings, which means they act as barriers that prevent dust and moisture from entering the mobile device (see **Photo 3**).

Photo 1: Attempting to talk on a cell phone while walking on a crowded city street is one of the many challenging environmental hurdles smartphone audio design engineers have tried to overcome. (Photo courtesy of Wikipedia)

Once past the wind screen's acoustic filtering, any remaining noise component of the wind turbulence can be attenuated through signal processing. Wind is not a constant noise. Between the wind's changing speed and direction, the user's unpredictable path, and the device's aerodynamics, we receive "non-stationary" noise. This means the noise's pitch, amplitude, envelope shape, and duration may all vary in a random fashion. Contrast this with stationary noise (e.g., a motor at constant speed and load and the user is not moving relative to the noise source). This situational noise is more easily suppressed with less computationally intensive signal processing.

Another type of noise that adaptive noise filtering designs must contend with comes from heating ventilation air conditioning (HVAC) systems. While air ventilation noise may not be that noticeable to those using a boardroom conference phone, for those on the receiving side of the call this type of noise can make the conversation fatiguing and unpleasant.

Photo 2: The IBM PC AT is an early example of consumer electronic co-processor use. (Photo courtesy of Wikipedia)

Photo 3: The BlueAnt's Pump HD Sportbuds use sophisticated wind noise suppression. The IP 67 rating means they have excellent resistance to dust, dirt, and water—to the point of being immersible. (Photo courtesy of BlueAnt Wireless)

Our hearing uses both ears to help us "tune out" ambient noise, but when the noise and the voice communication emanate from a single speaker source, our natural psychoacoustic processing does not work.

In the case of a party or similar environment where there are many simultaneous conversations (e.g., crowd babble), we experience the "cocktail party" effect. This lets us use our binaural hearing to zoom in on a single conversation that is not necessarily louder than the next. Dolby Labs has introduced a conference phone that actually enables up to three people to talk through separate loudspeakers spaced in a circle, which reproduces the "cocktail party" effect (see **Photo 4**).

Acoustic Echo Cancellation

Echo suppression and echo cancellation are signal processing methods used to improve voice quality either by preventing echo from being created or removing it after it is already present. Consider a full-duplex speakerphone operation. When you talk, your voice will be reproduced at the far location through a speaker. If the microphone gain at the far location is too high (above unity gain) then your voice from the loudspeaker will be picked up by the microphone and sent back to you, resulting in an echo.

There is always some degree of latency in voice communications systems, so the echo is further delayed, which can make conversation impossible. When conditions are right (wrong, actually!) the microphone will pick up your echoed voice and send it back to the other location and the feedback begins.

Echo suppressors work by detecting a voice signal traveling in one direction on a circuit, and then inserting a great deal of loss in the other direction. Usually the echo suppressor at the circuit's far-end adds this loss when it detects a voice coming from the circuit's near-end. This added loss prevents the speaker from hearing his own voice.

Echo cancellation involves first recognizing the originally transmitted signal that re-appears, with some delay, in the transmitted or received signal. Once the echo is recognized, it can be removed by subtracting it from the transmitted or received signal. This technique is generally digitally implemented using a digital signal processor or software. It can also be implemented in analog circuits. AEC enables natural full-duplex conversations—two singers at distant locations can sing a duet without interference created by the send/receive signal paths.

Photo 4: Dolby Labs developed a new speakerphone that is actually a circular device with a touchscreen. The microphones help distinguish the direction from which people are talking. A person calling can also sense movement (e.g., when the person speaking stands up or walks across the room). (Photo courtesy of Dolby Laboratories, Inc.)

Sources

eS700 Earsmart series
Audience, Inc. | www.audience.com

CS48LV12 and CS48LV13 Voice processors
Cirrus Logic | www.cirrus.com

AudioSmart software
Conexant | www.conexant.com

FM-36 Voice processor
Fortemedia, Inc. | www.fortemedia.com

Fluence active noise cancelling technology
Qualcomm Technologies, Inc. | www.qualcomm.com

Photo 5: Cirrus Logic's voice co-processor ASIC provides the voice processing features and performance needed for today's high-end smartphone, tablet, and other voice communication devices. (Photo courtesy of Cirrus Logic)

Multi-Microphones

Multi-microphone schemes adaptively monitor, detect, and process incoming voice, echo, and noise signal power levels to provide noise reduction and echo cancellation to achieve the clear full-duplex communication and audio capture.

Depending on the algorithm in use and its flexibility, two or more microphones can be used in a spaced array to create a beam (i.e., beamforming), even though each microphone may also be omnidirectional. If you look closely at your smartphone, you'll probably see at least two microphone openings (on the iPhone 5, the openings are on the top and the bottom). Some multi-microphone schemes also use a third microphone to sense wind noise (wind acts differently over spaced microphones than audio signals so the processor can discern and filter the wind). The microphone array creates a pickup pattern and a wide talking angle enables greater user freedom in handheld calls.

Who Creates This Technology?

The IC voice co-processors used in the leading smartphones are from Audience's eS700 Earsmart series and Cirrus Logic's CS48LV12 and CS48LV13. These are compact footprint, relatively low power-consumption ICs. These processors and variants have found use in a wide range of other mobile audio applications.

In some cases, the "guts" of these ASIC designs and/or the software algorithms have been integrated into larger "mixed signal" ICs (see **Photo 5**). Cirrus Logic has recently absorbed Acoustic Tech and Wolfson, both serious players in voice signal processing, which is why these name are not separately discussed.

Fortemedia also offers voice processors such as its flagship FM-36. Its software and hardware solutions are used in many mobile audio platforms (e.g., tablets and laptops).

Conexant's AudioSmart software is based on its proprietary Smart Source Pickup (SSP) noise suppression algorithm. SSP uses spatial representation of target speech and noise sources to reduce interference. Conexant claims that one of AudioSmart's advantages is that speech and noise can arrive at any angle relative to the microphones. SSP even removes noise when it comes from the same direction as the desired speech. SSP is also effective at reducing keystroke noise distractions during a conference. Conexant's approach uses software, where the platform (e.g., a tablet or PC) already has an ARM processor.

Qualcomm, the king of the smartphone chipset vendors offers Fluence for its own voice processing solution. A full range of active noise cancellation (ANC) voice processing techniques are offered. The Qualcomm Fluence is built into selected Snapdragon chipsets as an option.

Voice processing is the unsung hero in mobile audio devices. It filters out all types of noise, enabling natural full-duplex communication in speakerphones, suppressing echo and acoustic feedback, enabling clear speech conversations in noisy places, and improving voice command accuracy in speech recognition systems.

The vendors of these technologies operate in a fiercely competitive market, which drives continuous product design improvements. Further innovative R&D efforts will no doubt lead to better, faster, less expensive solutions that will benefit mobile device users in years to come. ax

Measurement Microphone Suppliers

A measurement microphone is a calibrated microphone designed for use with systems that quantify sound (e.g., sound level meters). Measurement microphones are built to be the most accurate and reliable microphones available. Because of the high quality demanded of these microphone types, manufacturers must use extremely high standards in the engineering of these devices. This article explores a dozen of the best-known measurement microphone suppliers.

By
Richard Honeycutt
(United States)

ACO Pacific

ACO Pacific, Inc., founded about 32 years ago, is an internationally known manufacturer and supplier of measurement microphones and microphone systems, industrial and community sound (noise) level alarm and monitoring systems, calibrators, sound intensity probes, acoustical/EMI isolators, and noise generators. ACO Pacific's company headquarters are located in Belmont, CA. ACO Pacific's microphones are available for measuring sound pressure levels (SPLs) from 2 to 175 dB, and frequencies from 3 Hz to 120 kHz. The line includes both free-field and pressure microphones. ACO Pacific capsules utilize stainless steel, nickel, or titanium diaphragms, and have a long-term stability greater than 250 yr/dB at 20°C. Each microphone is individually calibrated.

Audix Microphones

Audix Microphones was founded in 1984, primarily to produce microphones for recording and live performance. However, Audix Microphones also manufactures one relatively inexpensive measurement microphone. This microphone operates from 18-V-to-52-V phantom power and is prepolarized, using a polyester diaphragm. It's dynamic range extends from 28 dBA to 140 dB SPL. Its frequency response is stated as 20 Hz to 25 kHz. The Audix Microphones measurement microphone is not individually calibrated.

Behringer

Behringer Manufacturing Co., founded in 1989, produces many microphones for live sound and recording, but it only produces one test microphone model. However, that relatively inexpensive model is widely used due to the Behringer brand's acceptance. The Behringer measurement microphone has a polyester diaphragm and operates from 15-V-to-48-V phantom power, providing an advertised frequency response of 20 Hz to 20 kHz. The measurement microphone's self-noise and maximum sound

Photo 1: The Brüel and Kjaer model 4160 Laboratory Standard microphone is one of the most precise and stable microphones available. (Photo courtesy of Brüel & Kjaer)

pressure level (SPL) are not specified. Individual calibration is only available from third-party labs.

Brüel and Kjaer

The Danish company Brüel and Kjaer (B&K) has manufactured acoustical measurement devices and systems since 1943. Its measurement microphones—introduced in the 1950s—are used as primary standards in laboratories worldwide **(see Photo 1)**.

B&K measurement microphones use tensioned metal diaphragms. The company also makes externally polarized and prepolarized models, with frequency limits as low as 0.07 Hz and as high as 100,000 Hz. Electrical noise in some units is as low as –2.5 dB, using a preamplifier incorporating a frequency-compensation network. Maximum SPL limits can be as high as 184 dB.

Some models require an external preamplifier, while others do not. Phantom-powered and constant current powered (CCP) models are also available. All B&K measurement microphones are individually calibrated, and some support Transducer Electronic Data Sheets (TEDS). B&K's catalog includes American National Standards Institute (ANSI) Type 0, Type 1, and Type 2 microphones. The purchaser can choose among free-field, pressure, and random-incidence microphones.

Earthworks Audio

David Blackmer, founder of dBX, established Earthworks Audio as a loudspeaker manufacturer in the late 1980s. When measuring his loudspeakers, Blackmer concluded that human hearing involved frequencies up to at least 200 kHz. So when he began developing measurement microphones, he designed them with higher upper frequency limits (up to 50 kHz) than were usual for audio test microphones. The low-frequency limit extends to 3 Hz.

In addition to microphones for audio recording, four models of measurement microphones are available, permitting measurements from 22 dBA to 142 dB SPL (see **Photo 2**). All use polyester diaphragms. Each microphone is individually calibrated and electronic calibration files are available. Most models operate from 48-V phantom power, although there is one battery-operated model.

G.R.A.S. Sound & Vibration A/S

Danish acoustician/engineer Gunnar Rasmussen was a pioneer in acoustical measurement system design for more than 60 years. In 1994, he founded G.R.A.S., which manufactures measurement microphones and related acoustical equipment. G.R.A.S. is now owned and managed by his two

Photo 2: The Earthworks M50 is an ANSI Type 1 measurement microphone with a frequency response of 3 Hz to 50 kHz. (Photo courtesy of Earthworks Audio)

sons Per Rasmussen and Peter Wulf-Andersen. G.R.A.S manufactures both externally polarized and prepolarized microphones, the latter being used mainly with CCP preamplifiers.

Models are available for measurement as low as –2 dB SPL, and as high as 193 dB SPL. The low-frequency limit can be as low as 0.5 Hz; the high-frequency limits as high as 140 kHz. All G.R.A.S. measurement microphones utilize tensioned stainless steel diaphragms and are individually calibrated. All G.R.A.S. microphones are designed for measurement use.

iSEMcon

iSEMcon is a privately owned company with locations in Viernheim, Germany, and northwest Ohio, in the US. Partners Wolfgang Frank and Win Otto each have more than 20 years experience in design and application of capacitive sensors. Frank also has 13 years experience in measurement microphone design and manufacturing.

iSEMcon measurement microphones use prepolarized capsules with polyester diaphragms. The range of SPLs is 30 to 140 dB, and the frequency range is 10 Hz to 20 kHz. Some models are terminated in an RCA jack, and they are designed to be powered through the signal-output connector. Other models are terminated in an XLR or mini-XLR connector, and are phantom-powered.

A third group of microphones are CCP, and are terminated in an SMA, an SMB, or an BNC connector (see **Photo 3**). Individual calibration is included for some models. For other models, there is an extra charge for individual calibration. One model includes both free-field and diffuse-field individual calibration. Another model is rated International Electrotechnical Commission (IEC) Class 1. Some models support TEDS.

Photo 3: The iSEMcon EMM-7101-CHTB microphone can be phantom powered or constant current powered (CCP). (Photo courtesy of iSEMcon)

Josephson Engineering

Josephson Engineering was established in 1988 by David Josephson. The company makes a wide range of condenser microphones for studio, stage and field sound pickup, and audio measurement.

There are three models of Josephson measurement microphones that are differentiated by sensitivity, maximum SPL, and dynamic range, permitting measurements from 19 to 170 dB SPL. The frequency range extends from 20 Hz to 20 kHz, or higher with different capsules. The less expensive models use polyester diaphragms, while the higher grade unit has a traditional nickel diaphragm and quartz insulation. Most are free-field microphones, although diffuse/random-incidence capsules are available. The microphones are individually calibrated. The company also sells Microtech Gefell measurement microphones in North America.

Microtech Gefell

Microtech Gefell was established in Gefell (located in former East Germany) by Georg Neumann in 1943 following the bombing of his factory in Berlin. Throughout the Cold War, the Gefell factory supplied studio and measurement microphones to the Eastern Bloc countries. After the fall of the Berlin Wall in 1989, Microtech Gefell resumed selling microphones to the worldwide market. Microtech Gefell measurement microphones are similar in construction and performance to Brüel and Kjaer units, with frequency limits as low as 0.5 Hz and as high as 100 kHz, self-noise as low as 11 dB, and maximum SPL up to 172 dB.

In addition to measurement microphones and accessories, Microtech Gefell manufactures microphones for studio and broadcasting use. Free-field, pressure, and random-incidence microphones are available, and each microphone is supplied with an individual calibration curve. All Microtech Gefell measurement microphone diaphragms are made of tensioned nickel.

NTi Audio

The history of NTi Audio AG began in 1977 when the Test and Measurement Division of Neutrik AG introduced the AudioTracer chart recorder. In 2000, NTi Audio was formed through a management buyout of this division from Neutrik.

Headquartered in Schaan, Liechtenstein, NTi Audio manufactures handheld analyzers and end-of-line manufacturing test instruments as well as measurement microphones. NTi Audio's measurement microphones are 48-V phantom powered. They are also prepolarized. The microphone line covers SPLs from 12 dBA to 163 dB and frequencies from 5 Hz to 20 kHz. Most of the microphones have tensioned metal diaphragms, and have a long-term stability greater than 250 yr/dB. All models are free-field types, and pressure and diffuse-field correction factors are included. Each microphone is individually calibrated, and some models include a built-in electronic datasheet.

Parts Express

In 1986, Jeff Stahl founded Parts Express as a distributor of replacement parts for electronic equipment. By the year 2000, the company had become a major source of parts for small loudspeaker manufacturers and amateur loudspeaker builders. This led to the development of three low-cost measurement microphones that are marketed under the Dayton brand. These microphones share SPL limits of 57 to 127 dB and frequency-response limits of 18 Hz to 20 kHz. All three microphones have polyester diaphragms. Two models operate from 15-V-to-48-V phantom power. The third microphone is designed to be used with iOS, Android, or Microsoft mobile devices, to which a microphone can be connected using the TRRS plug for power and signal connection.

PCB Piezotronics

PCB Piezotronics began offering piezoelectric integrated circuit (IC) transducers in 1967. The company added capacitor measurement microphones with its acquisition of Larsen Davis in 1999. PCB Piezotronics manufactures externally polarized and prepolarized measurement microphones capable of measuring SPLs as low as 35 dBA and as high as 164 dB, in a frequency range from 0.25 Hz to 100 kHz (±3 dB limits). Some units support CCP. All PCB Piezotronics capacitor measurement microphones are individually calibrated and use tensioned metal diaphragms. Some support TEDS.

The Rion Co.

The Rion Co. was founded in 1944 and is headquartered in Tokyo. Rion manufactures and sells hearing instruments, medical equipment, sound and vibration measuring instruments, and particle counters. Rion's capacitor measurement microphones include both free-field and pressure types. The microphones cover a dynamic range of 2 dBA to

Photo 4: The Studio Six Digital iPrecision microphone is the only ANSI Type 1-certified microphone that is also an iOS direct-connect Lightning accessory. (Photo courtesy of Studio Six Digital)

164 dBA and a frequency range of 5 Hz to 100 kHz. All the measurement microphones use tensioned titanium diaphragms and are individually calibrated. Externally polarized and prepolarized models are available. Some preamplifiers are designed for CCP.

SoundFirst

SoundFirst is operated in Longmont, CO, by Ray Rayburn. In addition to selling measurement microphones produced by several other manufacturers, the company also makes two titanium-diaphragm measurement microphones, utilizing ACO Pacific. The design provides long-term stability in excess of 250 yr/dB. Both models are phantom-powered. One model is ANSI Type 1 rated and has a dynamic range of 10 to 138 dBA. Each microphone is individually calibrated.

Studio Six Digital

Studio Six Digital is focused on providing apps for iOS mobile devices, including two measurement microphones designed to work with these apps. The iPrecision Type 1 microphone (see **Photo 4**) has a tensioned metal diaphragm, is prepolarized, and is suitable for measuring from 20 to 125 dBA in two ranges.

The iPrecision Type 1 microphone offers a frequency range of 15 Hz to 20 kHz. Individual calibration is available at an extra cost. The output connector is an iOS-device-compatible cable. Long-term stability is greater than 250 yr/dB. The microphone also includes a power input port designed for the included 5-V/2.5-A power supply. This power is fed to the iOS device so the entire system can be run indefinitely on AC power. The Type 2 microphone has a polyester diaphragm. It offers a dynamic range of 28 to 120 dBA and a frequency range of 20 Hz to 20 kHz. The Type 2 microphone's individual calibration curve is stored in nonvolatile memory. This microphone's connector is also iOS-device-compatible.

Author's Note: The Loudspeaker Industry Sourcebook, *published by Segment, lists 60 measurement microphone suppliers, plus 30 microphone capsule suppliers. Some of these companies sell OEM microphones to other companies, not to end users. We have chosen to introduce a few of the best-known companies.*

Measurement Microphone Checklist

In selecting a measurement microphone, several specifics must be considered. These were discussed in last month's Sound Control article. For ease of use, I have created a user checklist.

Does the microphone need to be American National Standards Institute (ANSI) Type 0, Type 1, or Type 2?

- Type 0: laboratory standard, non-stringent environmental requirements
- Type 1: precision, accurate in field and lab
- Type 2: general-purpose, when high frequencies do not dominate

(Actually, the ANSI types are specified for sound level meters, not microphones, but sometimes they are used as indicators of microphone response flatness.)

Does the microphone need to be International Electrotechnical Commission (IEC) 651 standard Class 0, Class 1, Class 2, or Class 3?

- Class 0: laboratory standard
- Class 1: mainly for lab use, or in controlled acoustical conditions in the field
- Class 2: general field applications
- Class 3: for field noise surveys

What diameter is needed?

(This information is only a rough guide.)

- 0.125″ or 0.25″—most omnidirectional, least sensitive, and highest noise
- 0.5″—most commonly used
- 1″—most directional, most sensitive, and lowest noise

What distortion limit is needed?

- Smaller microphones tend to have higher maximum sound pressure levels (SPLs).
- If the preamplifier is separate, its distortion limit must be considered also.
- Some measurements can be made with up to 10% distortion, and the 10% limit may be 6 to 10 dB higher than the 3% limit.
- Unfortunately, some lower-priced measurement microphones list a maximum SPL on the datasheet, but do not give the % distortion at this level, leaving the user to wonder whether this is the 3% HD limit, 10% limit, or some other value.

Is a low electrical noise level critical?

- Larger microphones tend to have lower noise levels. For instance, the noise level of a 0.125″ microphone may be 45 dB higher than the noise level of a 1″ microphone.
- If the preamplifier is separate, its noise must be considered. Either thermal noise in the capsule or electrical preamp noise can be dominant.
- Equivalent input noise (EIN) is usually given with A-weighting, which is appropriate for sound-level measurements that are taken in A-weighted decibles (dBA), but not for room acoustics of loudspeaker equalization. As long as the EIN is at least 5 dB below the measured sound level, the noise's effect is not significant.

Will the microphone be used as one of a pair for sound-intensity measurement?

- Phase-matched pairs should be used for sound intensity measurements. These are available from some manufacturers.

Is a specific powering arrangement necessary?

- Some measurement microphones require a separate preamplifier that provides power to the capsule.
- Some microphones use phantom power, which can be anywhere from 12 to 48 V, and can be supplied by the test instrument or an external phantom power adapter.
- Some microphones use constant-current powering, derived from this type of measurement system. Such microphones accept power from an external constant-current source (nominally 4 mA, but it must be between 2 and 20 mA) and modulate the output voltage around a quiescent value such as 12 VDC. Constant current powered (CCP) microphones have low output impedances, and therefore provide better immunity to ambient electrical noise than do microphones having higher output impedances. CCP microphones are all prepolarized (electret), since the normal 200-V polarization is not available. Numerous trademarks apply to CCP microphones including: ICP, IEPE, Deltatron, AcoTron, Isotron, and Piezotron. (These trademarks are the property of their respective trademark holders.) CCP microphones typically use some variant of a BNC connector.
- Some microphones made for computer or mobile device use require power applied through the unbalanced signal cable—typically 3 to 10 V.

Is a metal diaphragm needed to provide stability for high operating or storage temperatures?

- Plastic diaphragms may be electrically or physically damaged by exposure to high temperatures. Metal diaphragms are more stable (often nickel or titanium are used).

What frequency range is required?

- The low-frequency limit can be determined by the capsule or the preamplifier, and is often 10 to 20 Hz for measurement microphones used in audio applications. However, the low-frequency –3 dB point for many high-quality test microphones is 1 to 3 Hz, and is as low as 0.01 Hz in some microphones.
- The high-frequency limit (at –2 dB point) is often specified as 20 kHz in microphones for audio applications. In less expensive microphones, response flatness may be questionable in the upper octave. Some microphones have an upper limit as high as 140 kHz.

Is the flattest frequency response specified for pressure, free-field, or random-incidence use?

- Pressure-response microphones are used if the microphone is flush-mounted or is part of the wall of a closed cavity.
- A free-field microphone should be used if the sound is coming only from one direction, as it compensates for pressure buildup at the diaphragm occurring at high frequencies under those conditions. A random-incidence microphone has a flat frequency response for sound in a diffuse field—arriving from all directions. When sound in a free field is being measured, a free-field microphone should be pointed at the sound source. If a pressure microphone is used in a free field, it should be pointed at right angles to the direction of the sound, and the high-frequency limit will then be lower than specified for pressure use. If a random-incidence microphone is used in a free field, it should be oriented at 70° to 80° from the direction of sound propagation (see **Figure 1**.)
- Random-incidence correctors are available for some free-field microphones.
- Most lower-priced measurement microphones are of the free-field variety.

Does the microphone need to have a specific connector?

- Many of the better measurement microphones screw mount directly to a preamplifier with the cable terminated in a 7-pin LEMO connector.
- Microphones designed for phantom powering usually connect via a 3-pin balanced XLR connector.
- Microphones designed for use with computer sound cards may terminate in a 3.5-mm TRS plug or an RCA plug.
- Microphones designed for mobile devices may terminate in a 3.5-mm TRRS plug or an Apple 30-pin connector.

Is the microphone supplied with an individual calibration curve?

- For precision measurements or for measurements that must withstand critical scrutiny by clients, courts, or governmental bodies, an individual calibration curve traceable to the National Institute of Standards and Technology (NIST) is essential.
- For general acoustical measurements, the response curve on a printed datasheet from a trusted manufacturer may suffice. Not all manufacturers can be trusted to print accurate curves, so if the response curve is printed in a heavy line and looks ruler-flat, it may not be reliable. **Figure 2** shows a printed response curve and an individual calibration curve.
- The Institute of Electrical and Electronics Engineers (IEEE) standard 1451 specifies a method of including calibration and identification information in a transducer such as a measurement microphone. This Transducer Electronic Data Sheet (TEDS) can be included in an EEPROM within the microphone or it can be supplied as a data file accessible from the measuring instrument with which the microphone is used. Many newer measurement microphones support TEDS.

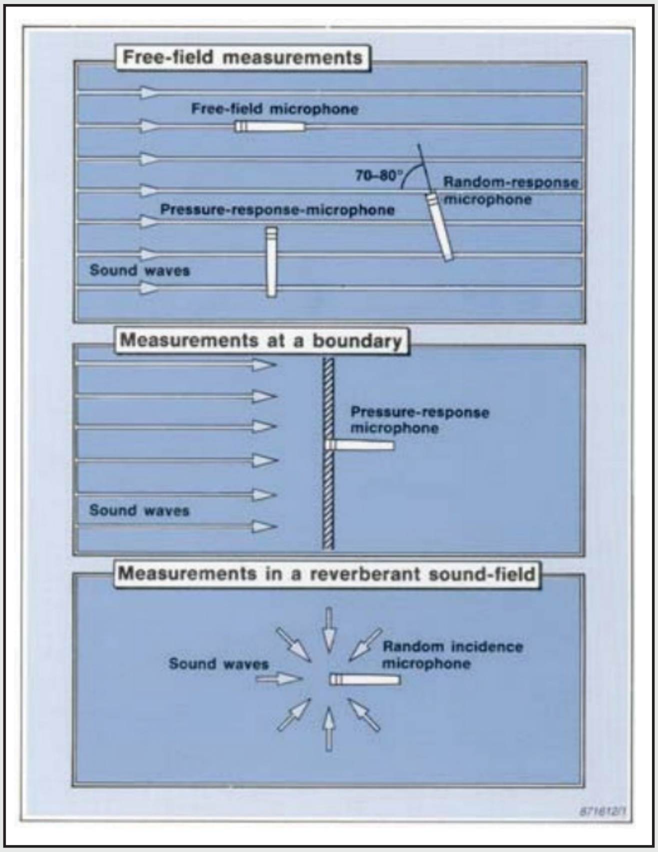


Figure 1: Measurement microphones can be calibrated for free-field, pressure, or random-incidence response. (Image courtesy of Brüel & Kjaer)

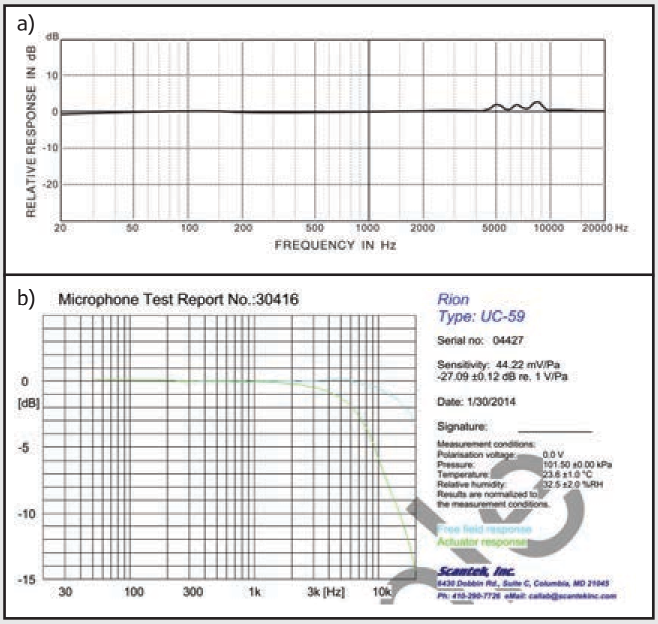


Figure 2a: This frequency-response curve from a printed datasheet tends to hide small variations in response. b: This individual calibration curve shows precise information about the microphone. (Images courtesy of Scantek, Inc.)

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Libratone CEO Marches to the Beat of His Own Drums

An Interview with Gregg Stein

Gregg Stein, Libratone's CEO, uses his drumming knowledge to hone his management style.

By
Shannon Becker
(United States)

Libratone teamed up with Berliner, Lizzy Courage, to design the first special edition speakers—five Libratone Lives and one Libratone Lounge—using her decoupage art. Courage's World of Comics speaker design is shown here.

SHANNON BECKER: Tell us a little about your background.

GREGG STEIN: First and foremost I am a musician. I started playing the drums at a very young age, ended up graduating from Berklee College of Music (Boston, MA), and ultimately began performing as a professional musician. While still in college, I was fortunate enough to work in a variety of sales and marketing roles at Universal Music Group and Sony Music. After Berklee, I joined the oldest continuously owned family-run business in the United States, the Avedis Zildjian Company. While I continued to work (and still do) as a drummer, I began to develop my craft as a marketer.

After several years, I went back to school to study entrepreneurship and global management at Harvard University. Upon graduation, I decided to apply my newly found marketing management skills to the burgeoning and fast-paced world of technology. At that time, I joined Numark Industries (now called InMusic Brands) where I played a key role in driving the business development and overall growth with brands such as Numark, Alesis, Akai Professional, MixMeister, and ION Audio. After many years of contributing to year-over-year (YOY) revenue growth, I joined Behringer and The MUSIC

Group, a leader in value driven professional audio products, to start a consumer electronics division from scratch. Following a successful launch of the brand in the consumer electronics market, I took on the Chief Strategist role at The Loop Loft, a leader in "real musician" based musical loop collections for drums, percussion, guitar, bass, saxophone, and more. After much success at The Loop Loft, I realized I was ready for my next challenge: to help kick start and position the Libratone brand in the Americas. Now, I am Libratone's CEO for the Americas.

SHANNON: How did you become involved with Libratone (www.libratone.com)?

GREGG: A good friend and former colleague of mine mentioned the brand to me. At the time, I had never heard of Libratone. But then I saw the website, and I fell in love. The Scandinavian design, the hi-fi audio quality, and the overall experience absolutely blew my mind. There are so many audio companies out there but very few that focus on the detail and the overall audio experience. I've now been with the company for a little more than two years. I originally joined the company as a consultant before taking on the CEO role. Since then, we have relocated our US company headquarters to Framingham, MA, with distribution at some of the most elite retailers ranging from Nordstrom to Design Within Reach to Crutchfield to Best Buy.

SHANNON: Tell us about the company. Founded in 2009, in the age of streaming, what is the goal when it comes to designing speakers for the 21st century?

GREGG: In 2009, Libratone's founders began a journey to become one of the first wireless audio companies to consider the aesthetics of loudspeakers—to move speakers out of the corner of the room and into the center of the room (and even outside the room for those consumers on the move).

Designed with Scandinavian tradition, Libratone develops high-performing sound refined through organic wool for a warmer, brighter listening experience. We have dared to stand out in a market, which to a large extent has been commoditized. Using unconventional approaches to design, materials, and colors, as well as through sales channels, we provide a strong alternative to existing brands in the market.

We are excited to see the adoption of wireless audio streaming. We support streaming music in AirPlay/DLNA environments, and we support direct

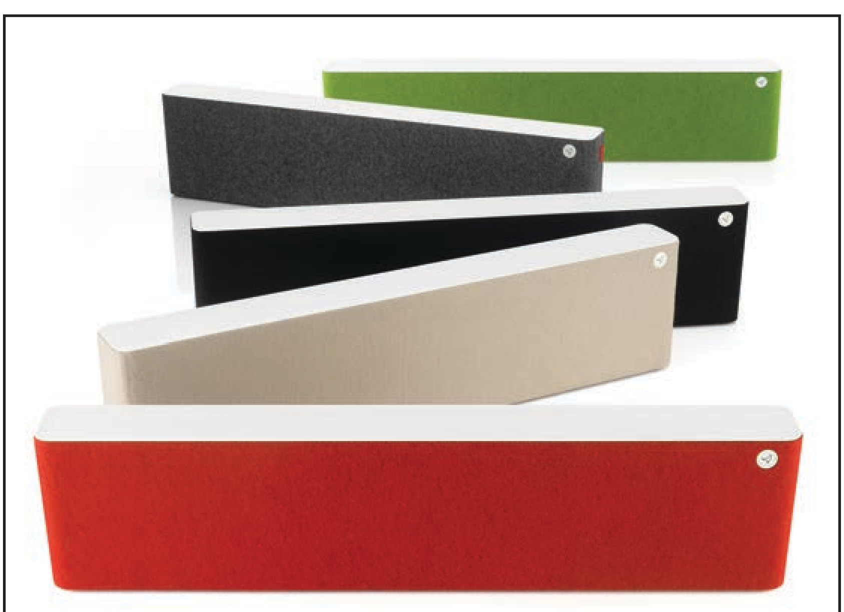
The Libratone Live can be moved anywhere in the house.

The Libratone Zipp is a portable, battery-driven speaker designed to provide high-end sound wherever you are.

streaming from the cloud through Spotify Connect, Qplay, HTC Connect, and several other streaming services that are under development. We have also just announced the addition of Bluetooth 4.0 with AptX and NFC hardware to our latest speaker line, available in the US starting December 1, 2014.

In recent times, we are seeing the proliferation of many new wireless standards. We are excited for AirPlay, DLNA, and Bluetooth to become ubiquitous in all Libratone products, but we also have a keen eye on the evolution of cloud-streaming protocols and multi-room capabilities.

The Libratone Loop is a small, versatile wireless speaker with big stereo sound that offers multiple placement options.



The Libratone Lounge provides room-filling sound and TV connectivity without the need for multiple speakers.

The best part is we are just getting warmed up. We aim to reinvent speakers for the 21st century by delivering on our promise of wireless freedom and innovation. Our speakers are meant to seamlessly integrate into your home decor while providing a listening experience unlike any other.

SHANNON: You have been an accomplished professional musician for more than 30 years. Do you think your knowledge of drums and percussion have an impact on the decisions you make at Libratone?

GREGG: The way that I think when I am drumming is analogous to the way I run a business. Being a musician for more than 30 years continues to teach me how to find the right rhythm, tempo, and counterpoint of a business—people, passion, and creativity.

Good music and great companies come from people. My role as a drummer and also as a business leader is to lay down that "foot-thumping groove" for those around me to dance. From behind the drum kit, I want to make everyone dance. In the boardroom, dancing could be translated to generating an emotional brand connection and genuine excitement for the company.

I'd like to think that my leadership style is largely linked back to my passion for the tribal nature of drumming. I am very customer and employee centric. I listen to those around me (especially our customers) and always look for ways to help those around me achieve greatness.

I bring a genuine passion to my work whether on stage or in the boardroom. I like to approach business decisions with creative, almost musical, solutions—as if I am on stage promoting the Libratone brand.

SHANNON: What are some of the challenges that you face in this highly competitive audio market?

GREGG: While the wireless audio category continues to become more competitive, there are few manufacturers that are truly innovating in this market space. Our opportunity is to push the boundaries of design, technology and user experience innovation. We see so much commoditization in this category, and it is our responsibility to "lead" by creating a differentiated and quality user experience for our customers. We are constantly looking for new ways to free our customer's music and sound; this is what we focus on every day.

SHANNON: Tell us what makes Libratone different.

GREGG: Design. Technology. Sound.

At Libratone, we create all-embracing, high-performing sound refined through organic wool for a warmer, brighter listening experience. Not only is our design unique, but so is our technology. We have developed FullRoom technology, which disperses sound in all directions, like an acoustic instrument. Moreover, our Danish lead sound engineering provides listeners with the confidence that their speakers are presenting the best and most honest musical experience. Our products are designed in Denmark to provide a sleek and simplistic look and feel. We look at our speakers not only as high quality sound systems, but also as design elements for your home. To me, Danish design is the simple, clean, and pure intersection between form and function.

SHANNON: Describe Libratone's first speaker?

GREGG: The Libratone LIVE, a flexible, simple and stylish wireless sound system designed to soundtrack your life, was the first speaker. To this day, it remains an iconic and award-winning speaker design. In fact, it can be found on permanent display at the Art Institute of Chicago and even in a recent issue of *WIRED Design Life.*

SHANNON: What have you learned about design?

GREGG: Design is everything. You want your product to not only look unique and appealing but to also provide a functional benefit. Our primary focus is to first make sure our speakers deliver an unparalleled audio experience with its design, inside and out, supporting those functions. The interior audio design is as crucial as the exterior, customizable experience. The stylish look to our speakers allows customers to integrate the sound systems seamlessly into their homes, leaving behind the days of black box speakers and tangled wires so all that remains is the purity and freedom of your sound.

SHANNON: What do you feel is Libratone's most successful product to date?

GREGG: We have seen tremendous success from the Libratone ZIPP. I can talk about the ZIPP all day but every time people hear it, they simply cannot believe it. It's 360° rich, warm hi-fi sound coupled with its battery life, interchangeable Italian wool covers, and Italian leather strap makes even the most skeptical person a believer.

The Libratone DIVA is a combined TV soundbar and wireless music system, which is easy to control with a TV remote via IR Learning.

SHANNON: What's next for Libratone?

GREGG: We are in the process of launching a new soundbar and all-in-one music sound system called the Libratone DIVA. We are also adding Bluetooth to our Zipp and Loop.

Build Your Own Audio Mixer (Part 2)

Add Circuits to Create a More Complex Unit

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

By
George Adamidis
(Greece)

In the second part of this article series, we will take the basic audio mixer and add circuits for audio equalization (EQ). We will add tone-control circuits for attenuating or boosting a range of frequencies (e.g., bass, midrange, and treble) on each audio channel, and a general graphic equalizer to perform general EQ at the output. We will also include a volume unit (VU) meter and a headphone monitor.

Photo 2. This prototype has 12 inputs and uses three-band tone control modules at each input.

Tone control enables listeners to adjust sound to their liking. It also enables them to compensate for recording deficiencies, hearing impairments, room acoustics, or shortcomings with playback equipment.

Our mixing console is a modular design so we can build as many input channels as we wish (see **Photo 1**). We can also upgrade the design to use some additional modules for equalization (tone control). **Photo 2** shows the upgraded design, which uses an additional module for tone control at each input (see **Figure 1**). Obviously, this is not the only possible arrangement. Different arrangements may use tone control modules only at specific inputs, instead of at all of them. You might also use one tone control module at an output (e.g., the output of the summing amplifier).

Usually, the additional module required for tone control is a two-band or a three-band tone control circuit. **Figure 2** and **Figure 3** provide two tone-control circuit design examples.

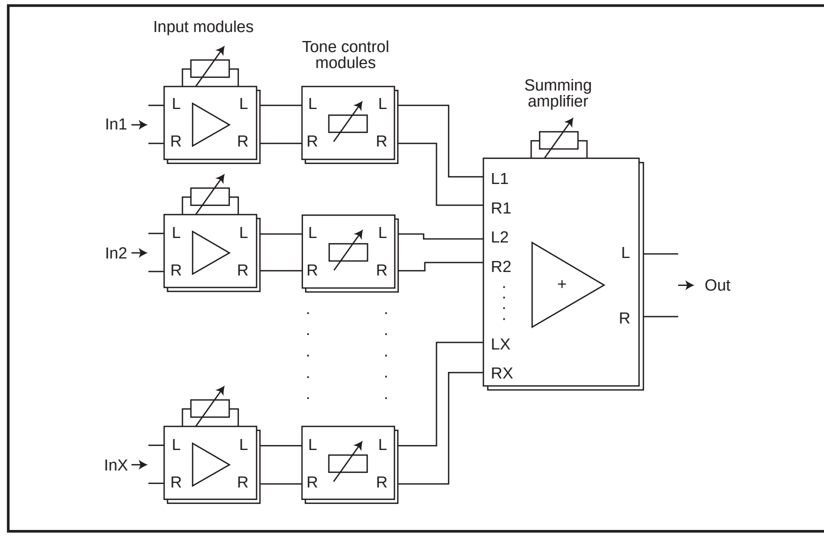


Figure 1: The audio mixer uses tone-control modules.

The circuit shown in **Figure 2** is a two-band tone control. It is able to attenuate or boost bass and treble (two frequency bands). The circuit is based on design formulas found in the datasheet for Texas Instruments's LM833-N dual-audio op-amp (see Resources). R1 and R2 potentiometers provide independent control of bass and treble frequencies, respectively. Both the bass and the treble can be boosted or cut. With both controls at their mid positions, the circuit provides a relatively flat frequency response (see **Figure 4**).

The two-band tone control circuit uses one low-pass and one high-pass filter at each audio channel (see **Photo 3**). Both filters are almost flat top at their pass-band, and any filter's flat-top gain can be independently varied between –20 and 20 dB. The cut-off frequencies (–3 dB) for the low- and the high-pass filters are preset at 340 Hz and at 10 kHz, respectively. If you want to alter them, refer to the design formulas.

The electronic schematic is quite simple. The circuit can be easily built on a universal board (i.e., a prototyping matrix board). However, you must use low-tolerance components to ensure the same response for both right (R) and left (L) audio channels. The circuit operates normally when connected to a low-impedance voltage source (e.g., the line-input module of our mixer). Due to DC-coupling, any DC present at the input may cause some problems. So use DC-blocking capacitors, if necessary.

The circuit shown in **Figure 3** is based on Texas Instruments's popular low–noise NE5532 op-amp and has three separate adjustable filters. The first one is an adjustable-gain low-pass filter. The second one is a band–pass adjustable filter. The third one is an adjustable high-pass filter. Each filter enables adjustments for bass, midrange, and high frequencies, respectively. The cut-off frequencies for the bass and high-frequency filters are about 200 Hz and 2 kHz, respectively (see **Figure 5**). The band-pass filter's central frequency is about 1 kHz. The normal gain, G, (when potentiometers are at midrange) is given by G = 20 × LOG (R2/R1). By using equal resistor values for R2 and R1, the normal gain becomes equal to 0 dB.

In **Figure 3**, only the right-hand audio channel circuit block is presented due to limited space. The same circuit block, should be also used for the left audio channel.

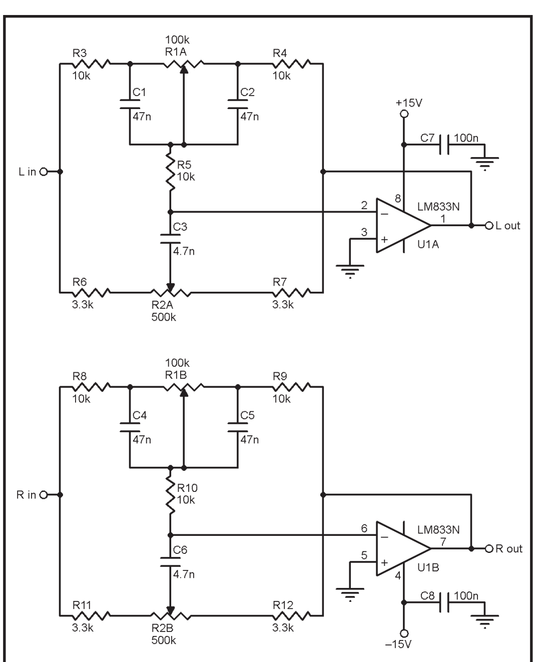


Figure 2: A two-band tone control circuit is shown.

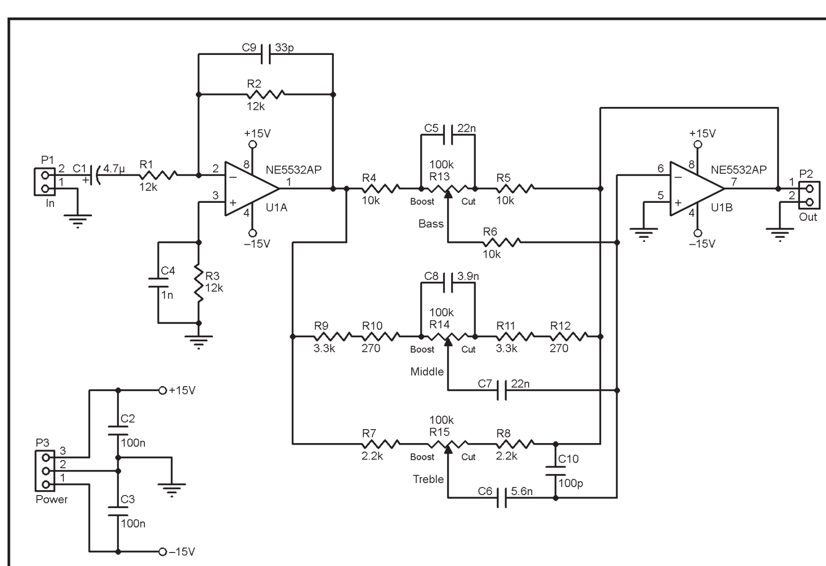


Figure 3: A three-band tone control circuit is shown.

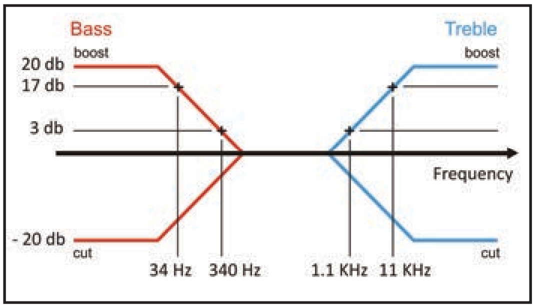


Figure 4: The two-band tone control circuit frequency response (Bode plot) is shown.

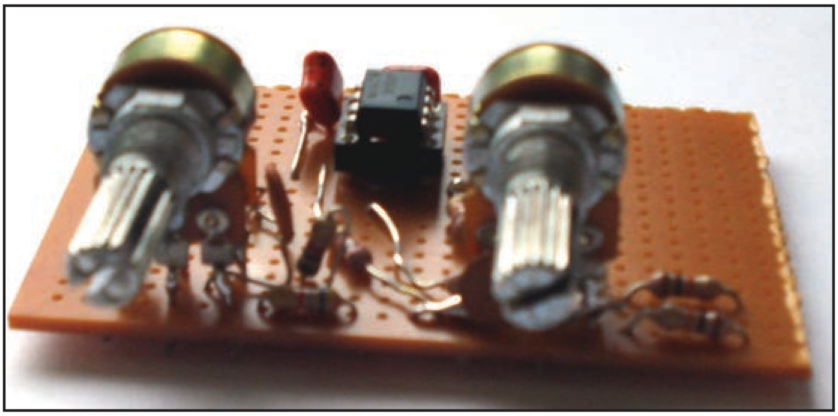


Photo 3: I built a two-band control module on a prototyping matrix-board.

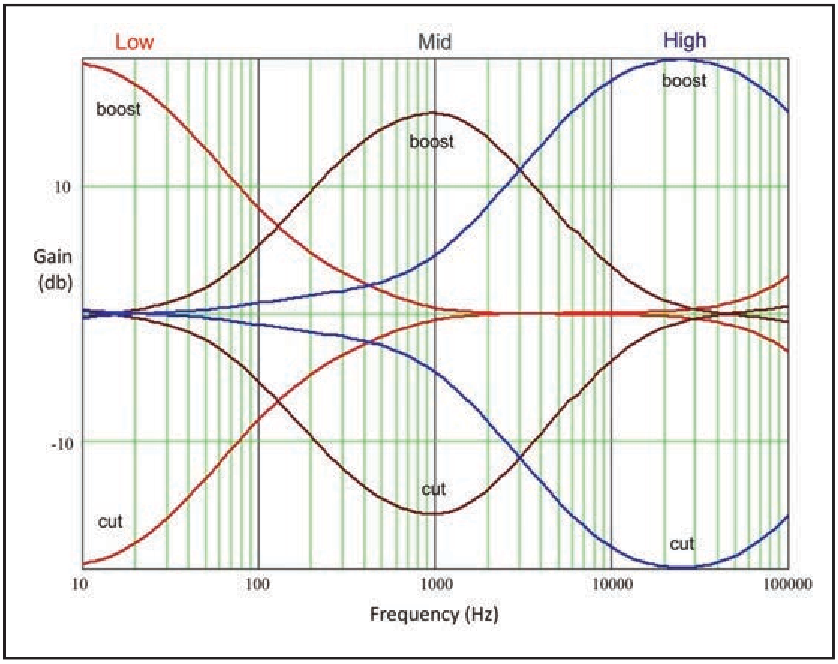


Figure 5: The three-band tone control circuit frequency response has a 1-kHz central frequency of the band-pass filter. (The simulation was done using Electronics Workbench software.)

Adding an Equalizer

At this point, our mixer uses relatively simple filters for limited adjustments. Graphic and parametric equalizers have much more flexibility in tailoring an audio signal's frequency content than a simple tone-control module. An audio equalizer is actually a bank of many adjustable filters. Using the modular concept, we can use one equalizer at every input of our mixer. However, since an equalizer is a complex and expensive circuit, it is more practical to use it only once, at our mixer's output (see **Figure 6**).

Figure 7 shows a reference design for a graphic audio equalizer. The specific circuit is based on Philips Semiconductors's Application Note 142, which is available from NXP Semiconductors's website (see Resources). The circuit itself has great performance and uses the NE5532, a top-performance op-amp.

The graphic equalizer consists of an input buffer (IC1-a), a variable-boost/ cut active filter (IC2-a), and an output summing amplifier (IC1-b). The IC1-a circuit is designed for unity gain and is mainly used for impedance-matching between the input source and the equalizer filters. The filter is a variable-bandpass or notching device, depending on the setting of the control potentiometer R2.

Any number of equalizer filter stages can be used within the range of about 20 Hz to 20 kHz (see **Figure 8**). However, the more stages you have, the easier it is to boost or cut a particular frequency without affecting the response at adjacent frequencies.

All the filter stages use the same R-C feedback-network configuration, to provide a maximum of about 15 dB of boost or cut at F_O, the center frequency. Different values for C1 and C2 are used in each stage, for setting the value of F_O.

Table 1 provides a list for the values of R1, R3, C1, and C2 for a seven-band graphic equalizer. Values for several arrangements can be found in Philips Semiconductors Application Note 142. C1 is about 10 times as large as C2 and the values for R1 and R3 are related to the value of R2, approximately by a factor of 10. The center frequencies of the list have been selected so that C1 and C2 are standard, off-the-shelf values. The equalizer uses linear slide potentiometers for R2.

The value of R6 depends on the number of filter stages used. It ensures that the gain across the equalizer is the same when all controls (R2s) are in the FLAT or 0 dB position. The nominal value of R6 is 100 kΩ divided by N, where N is the number of stages used. Note that only one audio channel is shown in the circuit schematic. To build a stereo version of the audio graphic equalizer, you'll need two of those circuits.

Frequently Asked Questions

What type of connectors can I use in my audio mixer?

You can use whatever type you like. However, the most common connectors used for line inputs and outputs are the RCA-type connectors. For microphone inputs and headphone outputs, the 6.35-mm (0.25") audio jacks connectors, both mono and stereo versions, are widely used instead of RCA. Audio jacks are also known as phone connectors. For any balanced input (e.g., for the balanced microphone preamplifier) you may use an XLR connector. XLR connector plugs and sockets are mostly used in professional audio.

Linear or logarithmic potentiometers?

You may use linear potentiometers for tone control. For volume control, you must use logarithmic potentiometers. This is because the human ear's amplitude response is approximately logarithmic. A logarithmic potentiometer ensures that on a volume control marked 0 to 10, a setting of 5 sounds subjectively half as loud as a setting of 10.

What exactly is the precision rectifier used on the volume unit (VU) meter?

When we talk about rectifiers, the first thing that comes to mind are power supplies. However, rectifiers are also used in high-precision volume unit (VU) meters to convert audio signal levels to DC. A common rectifying circuit will be completely useless for a high-precision VU meter. The reason is simply that the signal we would like to rectify will be less than the voltage needed to turn on a diode. Even small-signal germanium diodes require about 0.3 V to turn on. That may not seem like much, but when working with signals in the millivolt range, we must find a way to handle the problem.

The problem is solved by using a precision rectifier. The precision rectifier (aka a super diode) is a configuration obtained with one or more operational amplifiers to have a circuit behave like an ideal diode and rectifier. There are two standard methods for designing a precision rectifier: We can amplify the AC signal and then rectify it, or we can do both at once with a single operational amplifier. The circuit in **Figure 9** uses the latter method.

Is there any reason for using ±15 V voltage supply?

Actually, there is a simple reason for this. Using as high a supply voltage as possible will prevent saturation on DC connected amplifiers.

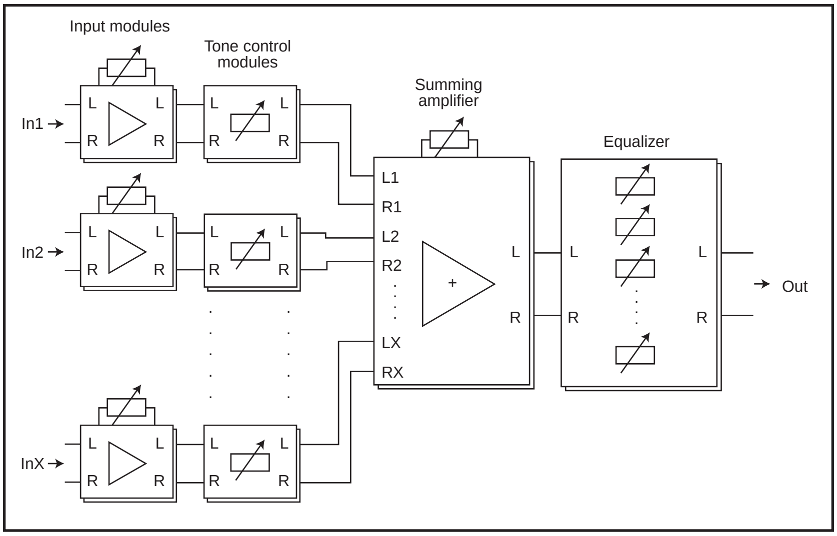


Figure 6: Adding an equalizer at the mixer's output is a practical application.

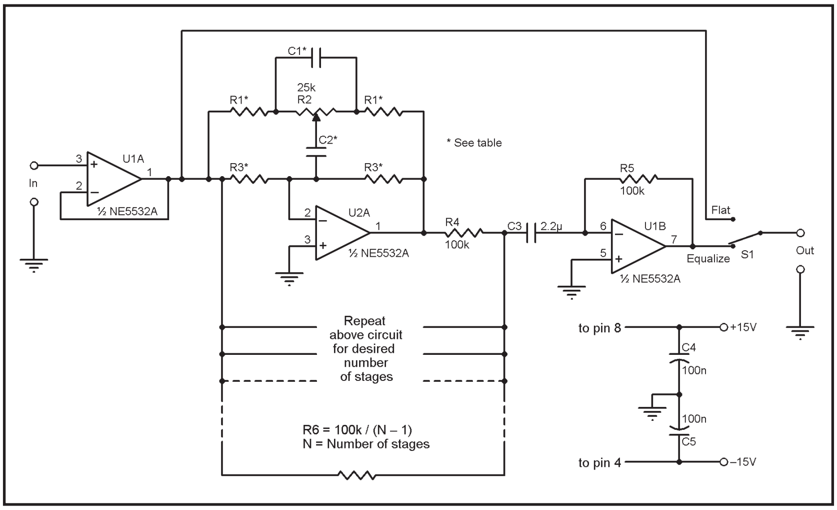


Figure 7: This reference design for a graphic audio equalizer uses the NE5532 op-amp.

R1 = 2.4 kΩ, R3 = 240 kΩ, R2 = 25 kΩ, R6 = 16.7 kΩ		
F_o	C1	C2
60 Hz	0.47 µF	0.047 µf
158 Hz	0.15 µF	0.015 µF
425 Hz	0.056 µF	0.0056 µF
1,082 Hz	0.022 µF	0.0022 µF
2,382 Hz	0.01 µF	0.001 µF
6,000 Hz	0.0039 µF	390 pF
15,880 Hz	0.0015 µF	150 pF

Table 1: This is the seven-band audio equalizer's component values.

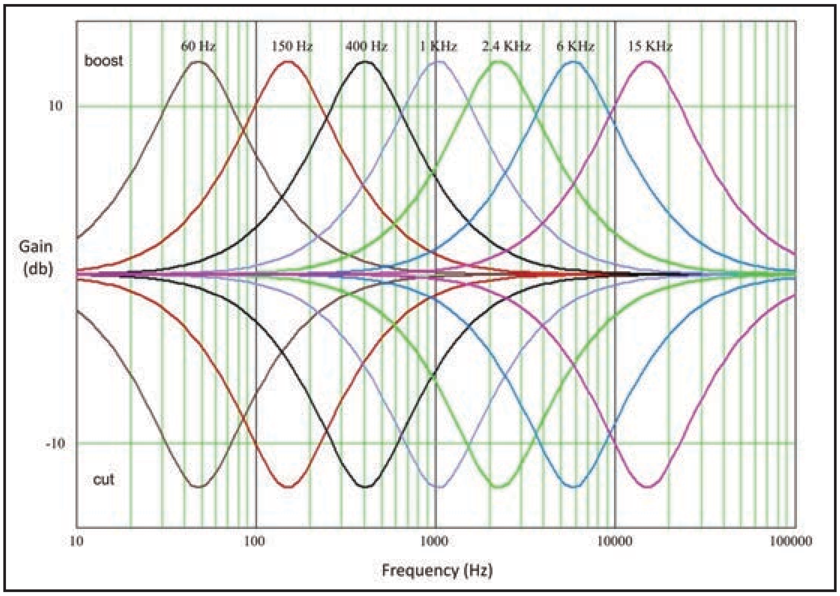


Figure 8: Any number of equalizer filter stages can be used. Here is a seven-band audio equalizer frequency response using Electronics Workbench software.

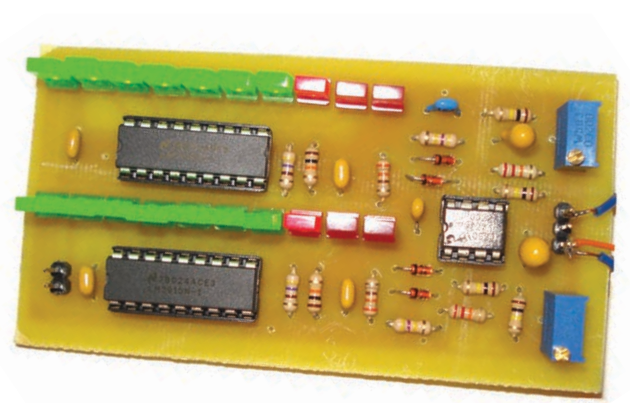


Photo 4. The bar graph LED volume unit (VU) meter module is based on LM3915 IC.

Adding a VU Meter

A volume unit (VU) meter is a device that is used to display a representation of the signal level. Think of the VU meter as a special kind of voltmeter that can be directly connected to any audio signal line of interest or similarly to a high-impedance voltmeter or oscilloscope input. The VU meter can measure the voltage at the specific line of interest while drawing minimal current (and hence minimal power) from the source. As long as the VU meter has enough input impedance, it will not affect the performance of the mixer circuits.

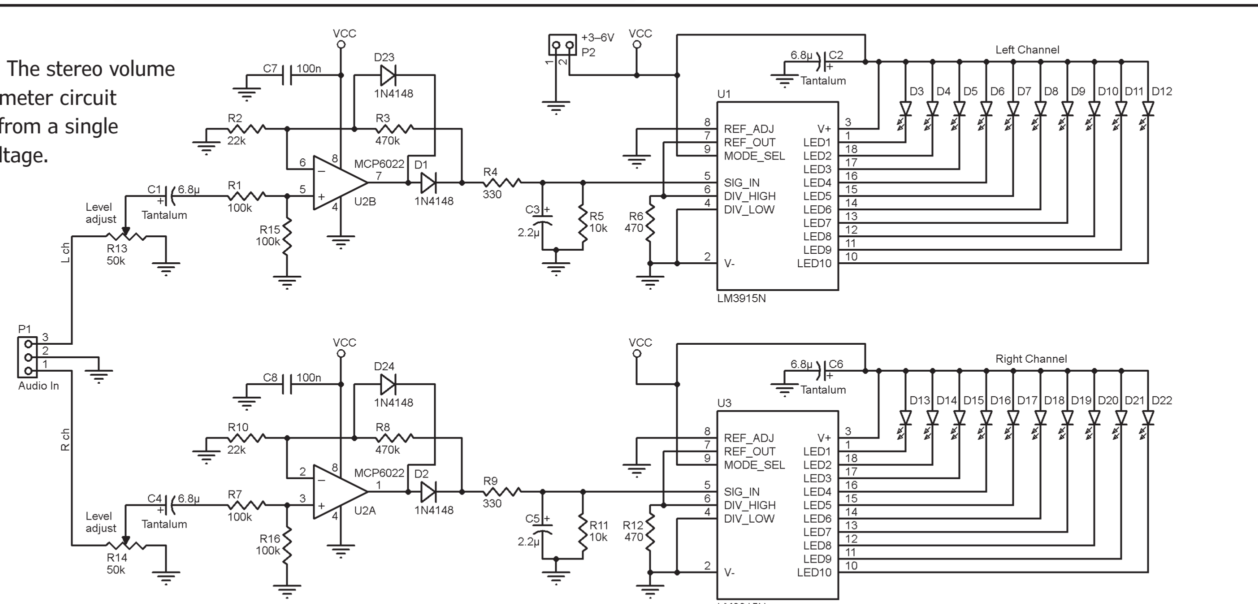
Most mixers have only one VU meter at their output lines, but some expensive mixing consoles offer independent signal-level indication for every input channel. Since we have a modular design, we can use as many VU meters as we want by simply directly connecting them to the signal lines of interest (e.g., at mixer's output or at any input-module's output).

Photo 4 shows an accurate LED-type stereo VU meter. The design is based on the well-known Texas Instruments LM3915 IC shown in **Figure 9**. The specific VU meter operates from a single supply voltage in the 3-to-6-V range. The maximum supply current at full scale is about 400 mA. It is not dependent on the power supply voltage.

The LM3915 senses analog voltage levels and drives 10 LEDs on a bar graph, providing a logarithmic 3 dB/step analog display. We use two LM3915 ICs—one for the right (R) and another one for the left (L) audio channel. We also use the MCP6022 dual op-amp IC as a precision half-wave rectifier and preamplifier.

The input impedance in each channel (R and L) is more than 33 kΩ. R13 and R14 are used for full-scale

Figure 9: The stereo volume unit (VU) meter circuit operates from a single supply voltage.



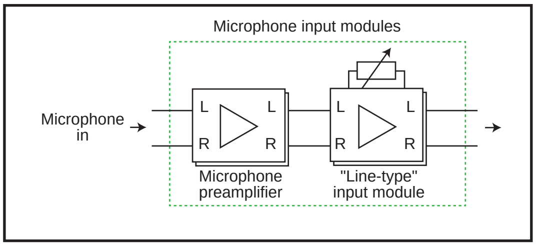


Figure 10: Converting a line-type audio input to a microphone-type one enables us to provide significant gain for a microphone source.

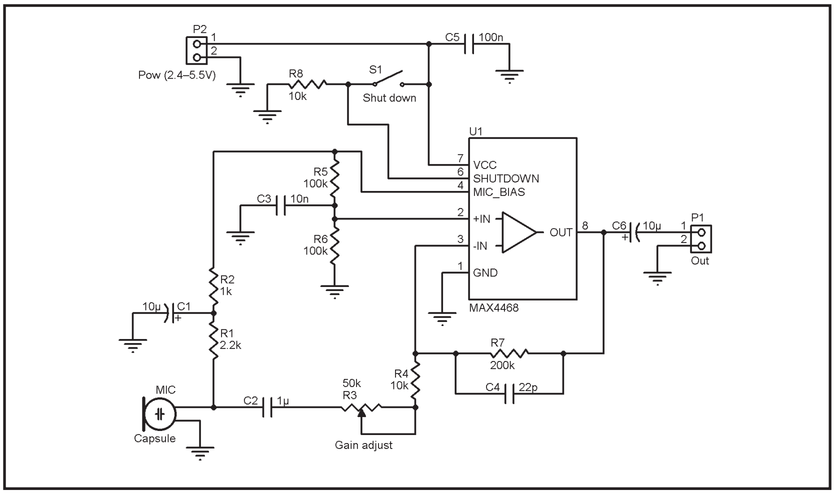


Figure 11: A microphone preamplifier is optimized to be used with an electret microphone capsule.

voltage level adjustment. The minimum full-scale voltage level can be as low as 150 mV. VU meters used at mixers outputs are usually adjusted for full-scale reading at about 1.4 V.

The C3, R4, R5, C5, R9, and R11 networks are simple integrators. They adjust the VU meter's rise and fall times for the L and R audio channel, respectively. The VU meter intentionally "slows" measurement, averaging out peaks and troughs of short duration. The "speed" of measurement is preset according to my preferences but it can easily be set to another value. By replacing C3, R4, R5, C5, R9, and R11 and after some trial and error, you will be able to find the ideal values according to your preferences. At one time, there were some standards for VU meter responses, but I think they mostly apply to the old passive electromechanical devices.

Adding Microphone Inputs

Up until now, the input stages were unable to provide significant gain for a microphone source because our mixer has only "line-type" audio inputs. We can overcome this limitation by adding a microphone preamplifier at any channel we want to convert from a "line-type" audio input to a "microphone-type" one. **Figure 10** shows the specific conversion method.

Figure 11 and **Figure 12** show two reference designs for microphone preamplifiers. The circuit shown in **Figure 11** is based on Maxim Integrated's MAX4468 IC. It is optimized to be used with an electret microphone capsule. The second circuit (**see Figure 12**) is a balanced microphone preamplifier based on Texas Instruments's NE5534 op-amp. It is an excellent choice for low impedance dynamic microphones.

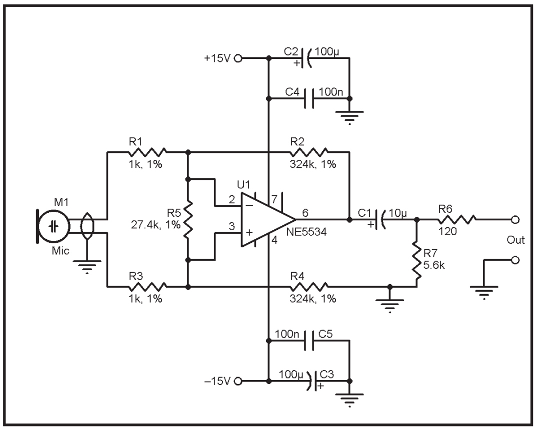


Figure 12: A balanced microphone preamplifier is used for low-impedance dynamic microphones.

Adding "Phono" Inputs

A "phono" input refers to a phonograph input. It is a special type of input that can accept signals from an analog turntable or a magnetic guitar-pickup (or other specific types of equipment). A phono input always uses a special circuit to boost the incoming signal. It also provides the Recording Industry Association of America (RIAA) equalization necessary to restore the original sound. If you still enjoy the vinyl sound or if you have a guitar that uses a RIAA magnetic pickup, you definitely need a phono input to be able to connect your classic turntable or your music instrument to the mixer. Again, you only need to convert one "line-type" audio input to a "phono-type" input. Simply add a phono

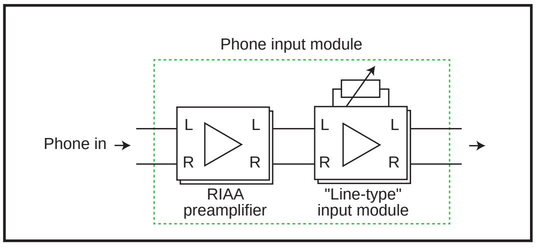


Figure 13: A line-type audio input can be converted to a phono-type one.

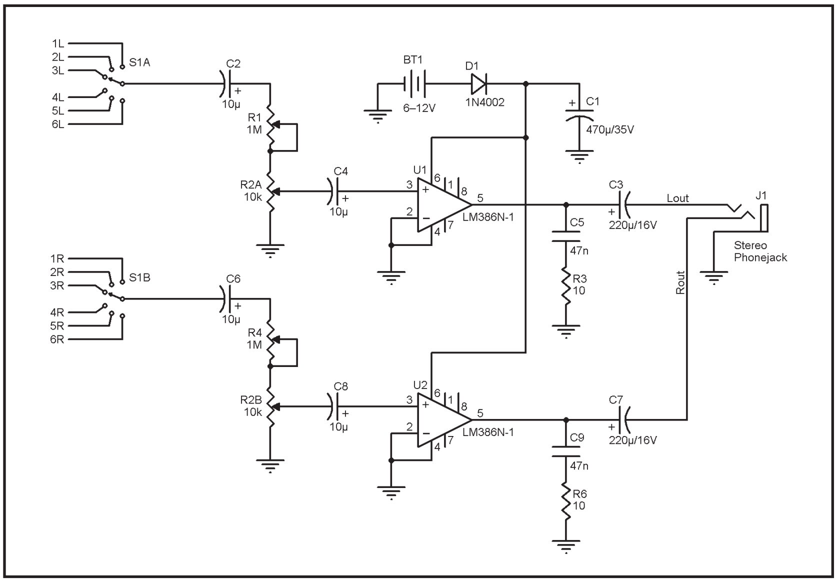


Figure 14: This stereo headphone monitor circuit is designed to be embedded in a homemade audio mixer

Photo 5: The headphone monitor module is shown without the rotary pickup selector switch.

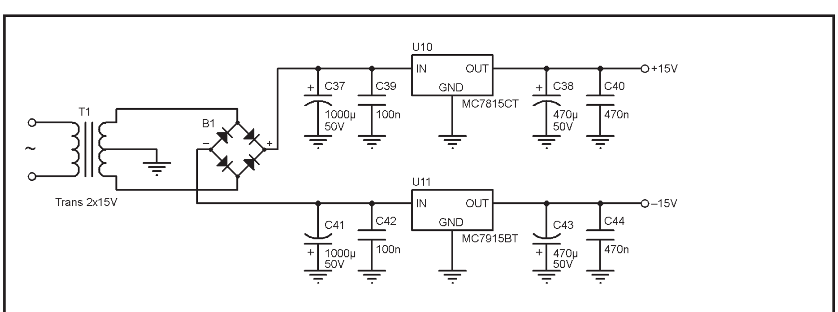


Figure 15: We use a simple linear power supply unit, which is based on 7815 and 7915 linear regulators.

preamplifier (see **Figure 13**). There are reference designs for RIAA preamplifiers in the LM833-N's datasheet (see Resources).

Adding a Headphone Monitor

To add headphone monitoring capability to your audio mixing console, use a headphone amplifier. With it you will be able to monitor the output of the mixer or any input. A headphone amplifier is just a small stereo power amplifier that provides a sufficient output to operate a pair of standard headphones.

Figure 14 shows a stereo headphone monitor circuit that is specially designed to be embedded in a homemade audio mixer. The circuit is equipped with a stereo volume-control potentiometer (R2) that enables the sound to be adjusted to a comfortable listening level. Due to its small size, it may be mounted inside an audio mixer by using the volume-potentiometer bush fixing (see **Photo 5**).

The circuit comprises two sections and a small number of components common to both sections. The first section is based on U1 and it is associated with the L audio channel. The second section is based on U2, which is responsible for the R audio channel.

Integrated circuits U1 and U2 are LM386 amplifiers. LM386s can provide up to 325 mW into an 8-Ω load. Standard headphones usually have greater impedance than this so the available output will be reduced. However, this does not matter because only a small output is sufficient to fully load the headphones. Since the components used for both audio sections are identical, only a single-channel description is needed.

For any input signal level, the preset potentiometer R1 must be adjusted so, when volume control (R2) is at maximum, there is minimal distortion combined with sufficient volume. R4, which is the preset potentiometer for the R audio channel, must also be adjusted so there is a correct balance (equality) in the volume between the left and the right channels.

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, radio frequency technology, smart antennas, antenna design, and electromagnetic compatibility.

The L channel's amplified signal appears at the output pin (pin 5) of U1 and AC-coupled to the left-hand headphone via C3. C5, which is connected in series with R3, is used to stabilize the amplifier and prevent any oscillation that might occur. The circuit can be powered from a 6-to-15-V power source via D1. D1 provides reserved-polarity protection.

S1 is a stereo, six-way rotary pickup selector switch that is used to select a specific input source (from six in total). To use the headphone monitor in your homemade audio mixer, you should connect inputs 1L, 2L,...6L and 1R, 2R,...6R, to the L and R outputs, respectively, of the modules you want to monitor. Think of the headphone monitor as a special signal probe that can be directly connected to any audio signal line of interest—similar to a high-impedance voltmeter or oscilloscope input. Using a six-way selector switch, you will be able to monitor up to six modules. For N inputs, you will need an N-way selector switch.

Power Supply

For powering the audio section, we use a simple linear power supply unit that is based on 7815 and 7915 linear regulators. Referring to the power supply electronic schematic, U10 and U11 are used to provide 15 V and –15 V. respectively (see **Figure 15**). A current of less than 1 A is enough to power up to 50 modules.

We use a separate 5-V power supply unit (PSU) for the VU meter (see **Photo 6**). The PSU used to power the VU meter is based on Texas Instruments' LM7525N-5 switch-type regulator (see **Figure 16**).

The End Result

This article's goal was to introduce a design procedure for building a high-quality modular audio mixing console. Modular refers to a design that can be split into smaller portions (modules) so when complete, they can be joined together to form one system. A modular audio mixer is formed by assembling some main modules that can be varied in number and/or disposition to suit individual needs. The examples presented here are only for reference. You can use the examples as they are or use different configurations. The circuits described here are also for reference. They have been built, tested, and exhibit good performance. However, there is always room for improvements.

You may notice that building an audio mixing console from modules, requires many wires. However, after deciding the number of modules you want to use and their specific arrangement, you could design a PCB, which will minimize the number of cable connections required.

Photo 6: The small power supply unit (PSU) is used to power the volume unit (VU) meter.

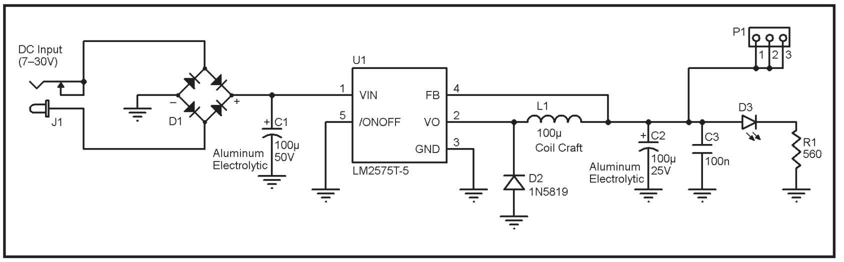


Figure 16: The power supply unit (PSU) used to power the volume unit (VU) meter is based on Texas Instruments's LM7525N-5 regulator.

Resources

Phillips Semiconductors, "Audio Circuits Using the NE5532/3/4," Application Note: AN142, October 1984, www.nxp.com/documents/application_note/AN142.pdf.

Texas Instruments, Inc., "LM833-N Dual Audio Operational Amplifier" datasheet, www.ti.com/lit/ds/symlink/lm833-n.pdf.

———, "AN-346 High-Performance Audio Applications of the LM833," Application Report, www.ti.com/lit/an/snoa586d/snoa586d.pdf.

Sources

MAX4468 IC
Maxim Integrated | www.maximintegrated.com

MCP6022 Dual op-amp IC
Microchip Technology, Inc. | www.microchip.com

Electronics Workbench Group
National Instruments Corp. | www.interactiv.com

LM833-N and NE5532 Op-amps, LM3915 IC, LM386 amplifier, and LM7525N-5 regulator
Texas Instruments, Inc. | www.ti.com

Correct Tube Amplification for Headphones Operation

Photo 1: The Elekit TU-8200DX vacuum tube headphone/integrated amplifier is a modern example of a headphone tube amplifier. The TU-8200DK comes as a kit that is aimed at the fairly experienced kit builder. (Photo courtesy of Elekit Audio, www.vkmusic.ca)

Due to their pleasant sound, single-ended Class-A-type tube amplifiers with low power output are used for operating headphones. Here, often wrong pre-conditions are assumed, because impedance matching is done instead of voltage regulation. Considering Ohm's law and the datasheet specifications of the headphone manufacturers, this article intends to demonstrate what is really necessary.

By
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(Germany)

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In electroacoustics, the term headphones means acoustic magnifier. It is most often used to play music through loudspeakers. If the music affects the eardrums, the belly, and the entire body, it is an enhanced sound experience. However, due to the volume, it can disturb those in nearby surroundings. In this case, headphone use does have its benefits. If you don't want to disturb anyone with loud music in the late evening hours but you still want to hear the smallest details of a music recording, headphones are the best solution.

Higher resistance dynamic headphones can be directly connected to low-resistance tube amplifiers. However, if the impedance goes down to 300 Ω or lower, it becomes critical. Most tube preamps with headphone output deliver a sufficient output voltage but too little power for very low-resistance headphones. In this case, a suitable transformer may elegantly provide a remedy.

The fundamental mistake when designing the transformer is the impedance matching, which is often misunderstood. Normally, graded transformer outputs from 32 to 600 Ω are required. The basis of everything is, and remains, Ohm's law. If a 4-W amplifier has a transformer output of 600 Ω, it will have a voltage of 49 V. This would hopelessly overload 600-Ω headphones. If the manufacturer specifies a maximum headphone load of 200 mW, this means it will have a voltage of 10.95 V at 600 Ω. Amplifier power of 4 W at 4 Ω equals a voltage of 4 V. This means that if a headphone with an impedance of 600 Ω is connected to the amplifier and operated up to its performance limit, the transformer output should deliver about 2.74 times the voltage that would otherwise be present at 4 Ω. This calculation analogously applies to all other headphone impedances. A minor mismatch is unproblematic. Apart from that, headphones produce harmful levels of more than 110 dB already at 100 to 200 mW and, depending on the type and the manufacturer, even more than 120 dB. Therefore, one should never reach the maximum headphone level. With any headphones, the benchmark is defined by the sound pressure that can be achieved at 1 V_{EFF}. This is similar to loudspeakers where the mean sound pressure is specified in decibels (dB) at a distance of 1 m and 1-W output power. Usually, headphones produce high volume levels at 20 to 50 mW. Furthermore, you will find sound level specifications at 100 mW, which is another benchmark. With that, many headphones can produce sound levels of 120 dB and more, which would damage your hearing!

Headphone Impedance

In the early days of electronics, headphones with an impedance of 2,000 Ω were used. This very high impedance was a concession to high-impedance tube technology. At that time, the low-current tubes were able to deliver high voltages, but only low currents.

Initially, the telephone systems were only equipped with tubes. A useful audibility over a dynamic headphone with an impedance of 600 Ω required 1 mW, which is equal to 775 mV_{EFF}. This gave rise to the reference level of 0 dBm, which in principle is an output power level related to 1 mW. Therefore, many headphones were manufactured with an impedance of 600 Ω, which was used for studio applications and home stereo systems.

In addition, there were headphones that featured small speakers with an 8-Ω impedance. However, their quality was rather poor. Today, modern headphones offer impedance ranges from 300 Ω down to 24 Ω. On the basis of low supply voltages delivered by batteries, portable MP3 players and similar devices require low headphone impedances—while there is little voltage, more current is available.

Assuming that most headphones already produce sound pressures far beyond 100 dB at 100 mW, the **Table 1** specifies the voltage and current requirements of typical minimum and maximum headphone impedances. The values are rounded and for the sake of comparison, the performance levels are specified at 1 V_{EFF}.

Using Class-A Amplifiers

Using **Table 1** as a base and assuming a voltage requirement increased by 3 dB for full level control, you would need approximately 11 V for 600 Ω, which corresponds to 200 mW. Amplification reserves are formed in case signal sources with low output levels are connected or CDs and records with low recording levels are played. If a single-ended Class-A power amplifier delivers 4 W at 4 Ω, which is equal to 4 V. In turn, 8 V at 16 Ω would exceed the requirement for a voltage increased by 3 dB with headphones that have an impedance of 300 Ω. If the transformer has 32-Ω tapping, then 11.3 V are available—making it possible to operate 600-Ω headphones. This is not about the actual power demand, but rather an amplifier's maximum voltage output. If you have a power output of 4 W at 4 Ω, it equals a current of 1 A.

As specified in **Table 1**, the flowing currents are calculated in the milliampere (mA) range, even if two or more headphones are simultaneously connected.

Headphone impedance	Voltage at 100 mW	Current at 100 mW	Output power at 1 V_{EFF}
24 Ω	1.55 V	64.5 mA	41.7 mW
32 Ω	1.79 V	55.9 mA	31.3 mW
50 Ω	2.24 V	44.7 mA	20 mW
80 Ω	2.83 V	35.4 mA	12.5 mW
150 Ω	3.87 V	25.8 mA	6.7 mW
240 Ω	4.9 V	20.4 mA	4.2 mW
300 Ω	5.48 V	18.3 mA	3.3 mW
600 Ω	7.75 V	12.9 mA	1.7 mW

Table 1: The voltage and current requirements of typical minimum and maximum headphone impedances are specified.

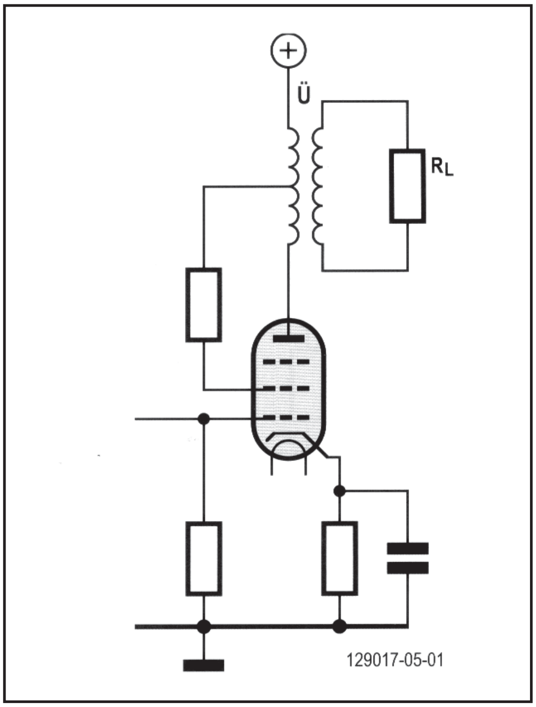


Figure 1: For headphone operation, the single-ended Class-A power amplifier's output transformer is terminated with a real load.

Impedance	R_L
4 Ω	22 Ω
8 Ω	33 Ω
16 Ω	47 Ω
32 Ω	68 Ω

Table 2: Reference values for the load current are provided.

If the power amplifier's power output is low when using headphones, the total harmonic distortion (THD) usually remains low. Therefore, normal single-ended Class-A power amplifiers can always be used for headphone operation. However, it is important that the amplifier is humfree and produces as little noise as possible, otherwise you will experience stress rather than enjoy the music.

Hum and noise directly entering at the ears (via headphones) are much more disturbing than hearing it through loudspeakers. For headphone operation it is also important that the single-ended Class-A power amplifier's output transformer is terminated with a real load (see **Figure 1**). Reference values for the load are provided in **Table 2**.

Resistors with a rating of 2 W each are used for R_L. This avoids the dangerous acceleration of the voltage inside the transformer if no load is connected. If the load resistor is connected to the highest tapping, a real termination of the secondary wiring is achieved, which benefits the whole system. In principle, it is not an issue whether a triode or a pentode is used as the end tube. Such an amplifier can be used for headphones and loudspeakers at high efficiency. If a pentode is used, a screen-grid tapping at the output transformer reduces the scratchy distortion part of the third harmonic as well as higher odd-distortion parts.

Variants

Now, there are two other variants when operating headphones with single-ended Class-A amplifiers. **Figure 1** shows the transformer being biased with direct current (DC). Therefore, a relatively big and expensive core with a corresponding winding is necessary for a good frequency response in the low-frequency range. To cleanly transmit wide frequency ranges, an elaborate winding technique (interleaved) is necessary. In case of low output power and high demands to the frequency response, the complexity rapidly increases. With increasing gain, triodes quickly tend to achieve high second-harmonic distortion values. When pentodes are used with a screen-grid tapping at the transformer, a protective resistor, and appropriate dimensioning, the overall distortion values remain low.

In case of negative screen-grid feedback caused by the transformer, the third harmonic decreases, which is characteristic for pentodes. This means the second harmonic predominantly remains, which results in a good sound with generally low distortion values. Using this amplifier principle, the negative feedback signal is typically directly tapped at the transformer's output winding. A more complex transformer will have a galvanically isolated negative feedback winding, and the output signal would be isolated from ground.

To avoid DC bias with all its disadvantages, you can use the circuit design shown **Figure 2**. In the anode circuit, the resistor R_A is used instead of the transformer. The disadvantage is that a lot of output power is turned into heat. The advantage is that the output transformer is coupled DC-free via the capacitor C. This enables the usage of a significantly smaller transformer core so the danger of running up without a load no longer exists.

Instead of the anode resistor, you could also use a plate choke. Plate chokes were used in many circuits during the early days of tube technology. This significantly reduces the power loss and the dynamic range is similar to the circuit shown in **Figure 1**. The big disadvantage is that the choke must feature good audio frequency characteristics. Also, the principle of its setup is similar to a DC-biased output transformer. Furthermore, the choke serves as an inductive source for the output transformer with its DC-free connection. So, nothing is saved or improved and the problem will only be eliminated with much effort.

Figure 3 shows a third option. Instead of decoupling at the anode, the tube is operated as a cathode follower. The disadvantage is that the tube makes no contribution to the overall amplification, as it is with the options shown in **Figure 1** and **Figure 2**. The big advantage is that the signal is available at low resistance at the transformer. Also, a plate choke here, as well as a DC-biased output transformer would be possible. However, you will still face the disadvantages described with the **Figure 2** design as well as higher costs.

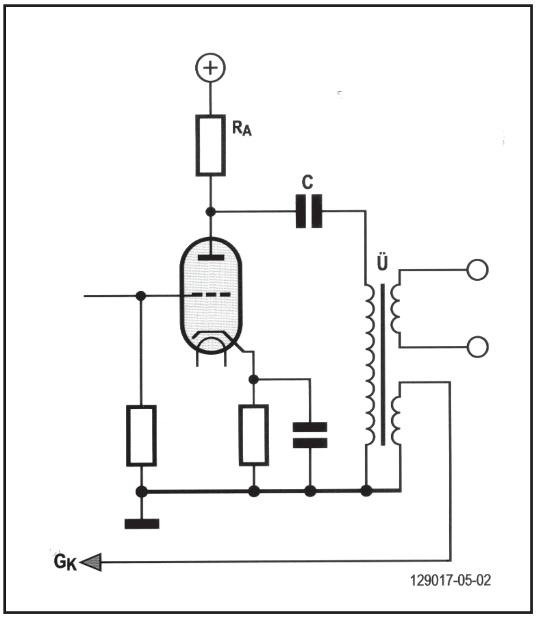


Figure 2: To avoid DC bias with all its disadvantages, use resistor R_A instead of a transformer.

If a galvanically isolated negative feedback winding is present, the transformer can be included in the negative feedback loop to have the output isolated from ground. A moderate negative feedback improves the harmonic distortion, the frequency response, and the output transformer. The transformer adapts the high output voltage at low current to the higher current at low voltage to the connected loads. A completely ground-free output over a voltage resistant transformer is safe. Output transformers are safety components that are 100% tested by the manufacturers for their voltage resistance. If a tested transformer is interconnected, it won't prove harmful even if the isolation of the conductors or the headphone itself is damaged.

If you are wearing headphones with defective isolation, the high operating voltage of the tube could lead to death in the event of an amplifier failure. High-quality headphones are designed to protect against destruction should an amplifier component become defective. A properly built and tested transformer protects against unwanted high direct and alternating voltages at the output that can be caused by a defective tube amplifier.

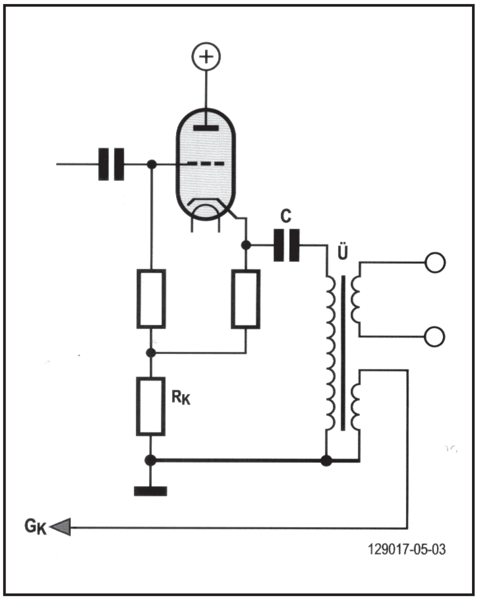


Figure 3: This circuit design operates the tube as a cathode follower instead of decoupling at the anode.

Sufficient Current Needed

In all three cases, the transformer converts the tube's high output voltage at low current into a low voltage with a higher current. In all three circuit designs, it is important that the tube is able to deliver sufficient current to power the transformer. Let's say that with the circuit shown in **Figure 3**, headphones with an impedance of 24 Ω is connected and the transformer features a transformation ratio of 8:1. For 100 mW, the headphones require 1.55 V at 64.5 mA. The transformation ratio of 8:1 requires a voltage of 12.4 V at 8 mA at the transformer's primary winding. According to the calculations, a tube with a load current of about 10 mA would be enough. However, this will not work because the internal resistance of the tube with pinout would be too high to power the transformer.

This is easy to understand when you build the corresponding circuit. The signal deformation and, therefore, the harmonic distortion will increase as the level increases. For this to work, you must use tubes with a sufficiently high operating current that also tolerate the transformer's inductance and its low copper resistance and that are able to magnetize the iron core.

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. He is now working on an updated followup.

Space-Charge Tubes

Last month, we discussed using the starved amplifier approach for operating triodes and pentodes at low voltages. However, there is another way to build tube preamplifiers for low-voltage operation: use tubes specifically designed to operate with low plate voltages. These tubes were designed for use in automobile radios.

Photo 1: Many of the low-voltage tubes are housed in 7-pin miniature packages.

By
Richard Honeycutt
(United States)

The original car radios used standard vacuum tubes. In fact, car radios were a primary market for 8-pin "loctal" tubes. But all these designs required a high B+ voltage—in the 180-to-200-V range. Since car radios were powered by a 6-V car battery, then later a 12-V battery, a vibrator had to be used to chop the DC into a square wave, which was then fed to a power transformer to provide the high B+ voltage. Vibrators were an added expense for radio manufacturers. They also produced mechanical and electrical noise and were not very reliable.

In the late 1950s and early 1960s, transistors were available, but they were expensive—especially radio frequency (RF) transistors. Therefore, the major tube manufacturers designed a series of vacuum tubes that would operate with plate voltages in the 12-V range. These were called "space-charge tubes," since they were designed to use the space charge that developed in the tube near the directly heated cathode for control-grid biasing. (Of course, this is just grid-leak biasing, but "space-charge tube" is a more marketable name than "grid-leak tube.") By using a single-ended germanium transistor for the power amplifier (car radios only produced a few watts of audio) with the low-voltage tubes used for the RF amplifier, oscillator, mixer, AGC, and audio preamp functions, manufacturers were able to eliminate the vibrator and the power transformer. The radios were only produced for a few years, but the number of tubes used was huge, and there are quite a few new old stock (NOS) low-voltage tubes still available.

Types of Space-Charge Tubes

Versions of space-charge tubes included diode/power tetrode combinations, diode/triode combos, pentodes, dual triodes, and pentagrid tubes. Most used a miniature 7-pin package (see **Photo 1**), although there was one—the 8056 high-frequency triode— that used a nuvistor package (see **Photo 2**).

The tetrodes, pentodes, and pentagrid tubes often used the grid nearest the cathode as an accelerator grid, which was biased at 12 V to speed the electrons from the space charge toward the control grid. Grid #2 (usually the screen grid) was then used as the control grid. In all these tubes, the

Model	Type	μ	g_m (μS)	Filament current (mA)
12DY8	Triode/tetrode	17	2,200 μS	350 mA
12EC8	Triode/pentode	25	4,700 μS	225 mA
12EL6	Dual diode/triode	55	1,200 μS	150 mA
12FK6	Dual diode/triode	7.4	1,200 μS	150 mA
12FM6	Dual diode/triode	10	1,300 μS	150 mA
12FR8	Diode/triode/pentode	10	1,200 μS	320 mA
12U7	Dual triode	20	1,600 μS	150 mA
8056	High-frequency triode	10	4,000 μS	135 mA (at 6.3 V)

Table 1: A variety of new old stock (NOS) space-charge tubes are still available.

Photo 2: A few low-voltage tubes used nuvistor packages.

plates of unused diodes can be grounded.

All these tubes were designed to operate with filament voltages between 10 and 15.9 V, which is the range of voltages produced by a "12-V" car battery. Since the triodes are probably of greatest interest for audio use, these are listed in **Table 1**. All these models are available as NOS tubes from Internet vendors.

Let's look at the 12EL6 as an example of these triodes. **Figure 1** shows the plate characteristic for this tube. Plotted on this is the load line for the circuit suggested on the datasheet as a Class-A amplifier. (It's almost impossible to read any data or this load line—more on this later.)

Figure 2 shows this amplifier's schematic. The circuit produces a voltage gain of 16. However, the maximum plate current is 12.6 μA (12.6 V/1 MΩ). A μ of 55 is specified for a plate current of 750 μA. This raises the question of whether we could improve the gain by using a higher quiescent plate current. Another advantage (for the designer) of higher I_{PQ} is that it would let us see enough of the plate characteristic near the load line to estimate distortion performance. Of course, increasing I_{PQ} with the 12.6-V B+ limit would require a smaller plate resistor, which would lower the gain.

If we use a constant current source instead of

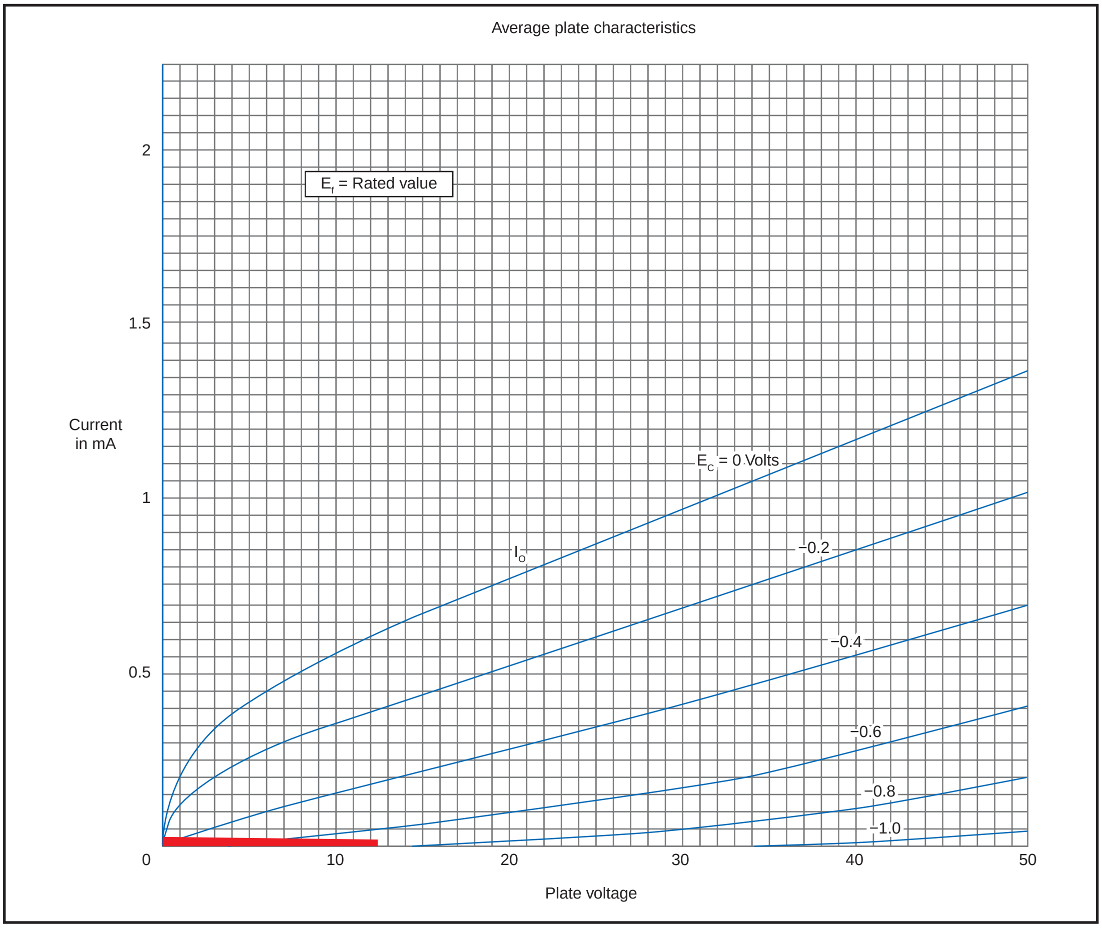


Figure 1: The load line of the example circuit given in the 12EL6 datasheet is biased such that the load line is almost off the graph!

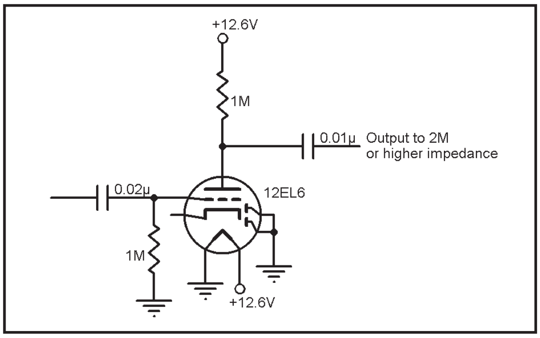


Figure 2: The 12EL6 datasheet gives this example circuit for an amplifier with a gain of 16.

the plate resistor, we could increase voltage gain and I_{PQ}. The easiest way to do this is to use a constant-current diode (CCD). If we're into experimenting for fun, we could obtain a 12EL6 and a 12-V wall wart rated greater than 150 mA. Then, carefully measure the low-voltage plate characteristics. Alternatively, we could use a 24-V wall wart, add a 12-V three-terminal regulator for the filament supply, and use the plate characteristic curve that we already have. Using a 1N5292 CCD (0.62 mA) and a 24-V B+, we have the load line shown in **Figure 3**. (Note that I have added an approximate plate characteristic curve for a control-grid bias of −0.1 V.)

You can see from **Figure 3** that the triode will be essentially cut off (indicated by the plate voltage being equal to the B+ voltage) at a control-grid voltage of −0.2 V. The graph indicates cutoff near a control-grid voltage of −0.17 V, but if we actually plotted the triode's performance in this range, there would be some curvature in the load line as the CCD became unable to deliver the full 0.62 mA at increasingly negative control-grid bias values. With this arrangement, the graph tells us that changing the signal voltage from −0.17 V to 0 V gives us a plate voltage change of 11.5 V—a voltage gain of 68. (Really, the math tells us that the gain equals the mu of 55. The discrepancy stems from the graph's inaccuracy near cutoff, and our inability to precisely read the graph.) A different CCD could have been used, providing a different plate current, but the 1N5292 is available in stock at my favorite supplier; whereas, many of the others are special-order items.

Control-Grid Bias

A control-grid bias of −0.1 V is a bit challenging with a 1-MΩ grid resistor, since the negative bias

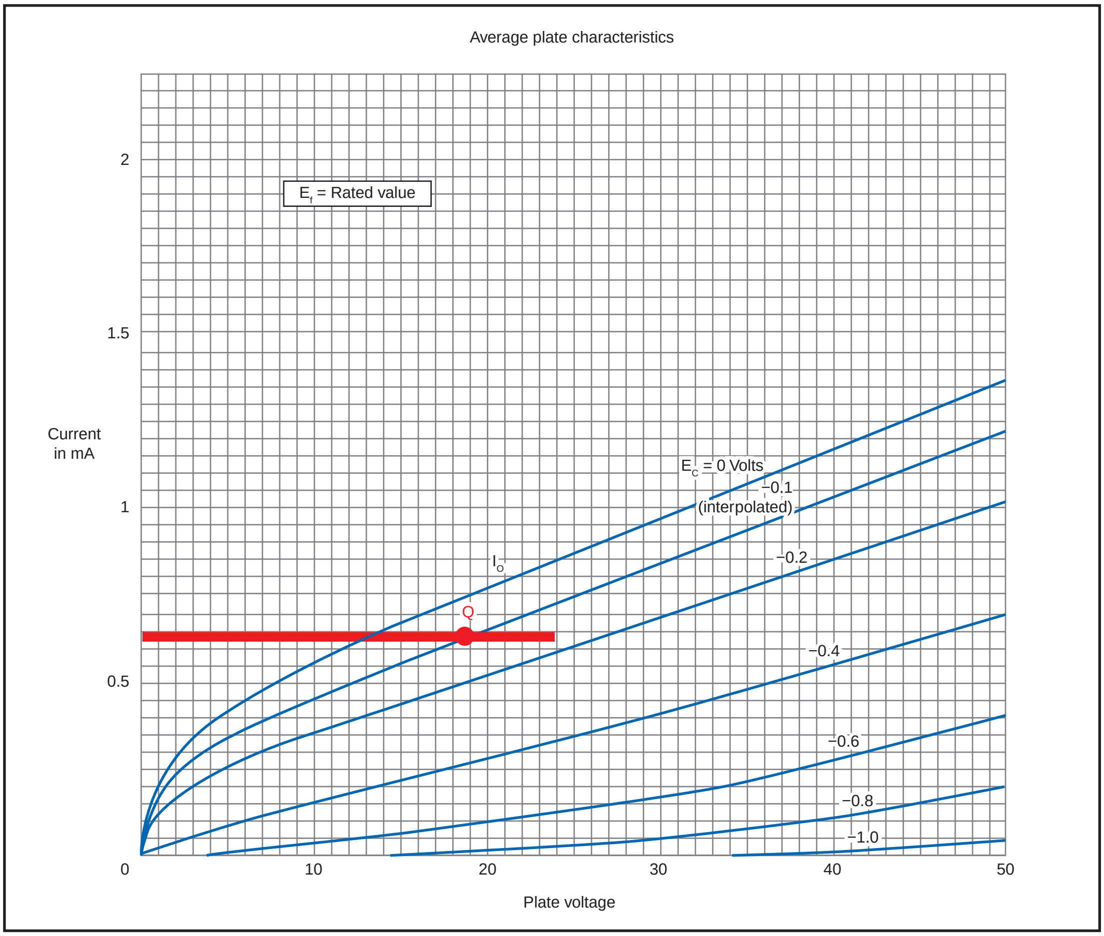


Figure 3: This is the load line for a 12EL6 using a 24 V B+, with a 0.62-mA constant-current diode (CCD) instead of a plate resistor.

created by "pitchfork-in-the-rain" current in that resistor may be as much as −0.6 V for a 12EL6. Unfortunately, the datasheet does not include curves that enable us to accurately calculate this grid-leak bias voltage. But we can experimentally bring the bias voltage to the desired value by connecting a very large resistor from the grid to a point with a known—but adjustable— positive voltage (see **Figure 4**). This will add a small positive voltage to the grid-leak voltage, and we can adjust for the sum to equal −0.1 V.

A side benefit of this approach is that the bias potentiometer gives us a way to adjust the distortion to our preference, and to compensate for variations in performance when the 12EL6 ages or we replace it. The input resistance of the stage will be the 1 MΩ and the 10 MΩ resistors in parallel—about 900 kΩ.

If you measure the control-grid bias voltage while adjusting it, use an electronic voltmeter, since most of those have an input impedance of 11 MΩ, which will only cause about an 8% error due to loading.

To have as much gain as possible, we need a very high impedance for the next stage. If we added another 12EL6 stage with the grid circuit shown in **Figure 4**, we'd have an AC plate load of 1 MΩ, giving us a gain of:

$$A_V = \frac{\mu r_L}{R_P + r_L} = \frac{55 \times 900{,}000}{52{,}631 + 900{,}000} = 51.96$$

if we use the 52,631-Ω plate resistance found as the inverse slope of the plate characteristic at the Q point. To do us any good, the added stage would have to be connected as a cathode follower (see **Figure 5**). Then we'd have a low enough output impedance to feed the next piece of equipment. In this case, the next piece of equipment would be a guitar amplifier, likely having an input impedance of about 500 kΩ, but perhaps less for a solid-state amplifier. The output impedance of our circuit should be lower than this by a factor of at least 10.

You may have suspected that this design was meant to be a guitar distortion "stomp box," and you'd be right. Many distortion boxes have attempted to imitate the sound of a tube amplifier, often resulting in large pedal boards of effects (see **Photo 3**).

To properly bias a cathode follower, we first have to examine the cathode resistor's effect on gain and output impedance. These are given by:

$$A_V = \frac{\mu R_k}{R_k + r_p} \text{ and } Z_O = \frac{r_p R_k}{(\mu + 1)R_k + r_p}$$

We often approximate a cathode follower's gain as unity. However, we see that this is true only if

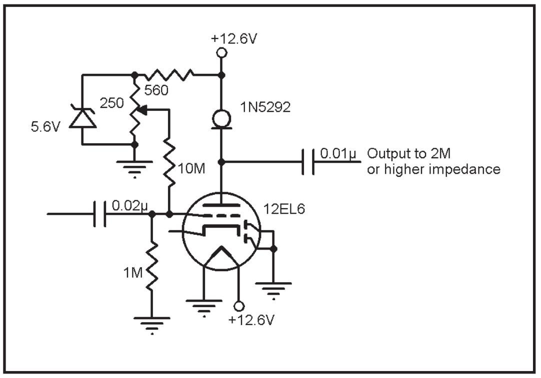


Figure 4: This amplifier stage using a 12EL6 has a gain above 50.

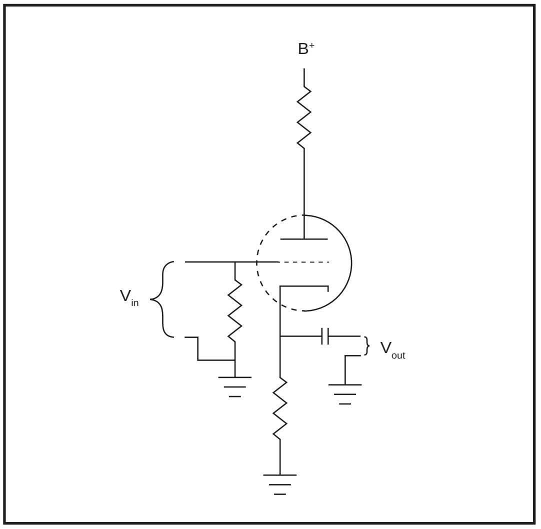


Figure 5: The cathode follower stage provides a low output impedance.

R_k is large compared to r_p. A cathode resistor this large would bias the control grid too far negative. If we adjust the bias potentiometer for −0.4 V on the control grid, using a 24-V plate voltage, we can see from **Figure 3** that the plate current will be about 0.33 mA. This would require a cathode resistor of 303 Ω (0.1 V/0.33 mA). The standard 5% tolerance value of 300 Ω is close enough.

Looking at the plate characteristic's inverse

Photo 3: Several of a guitarist's many effects pedals are intended to imitate tube distortion. (Photo courtesy of Michael Morel, Barcelona, Spain.)

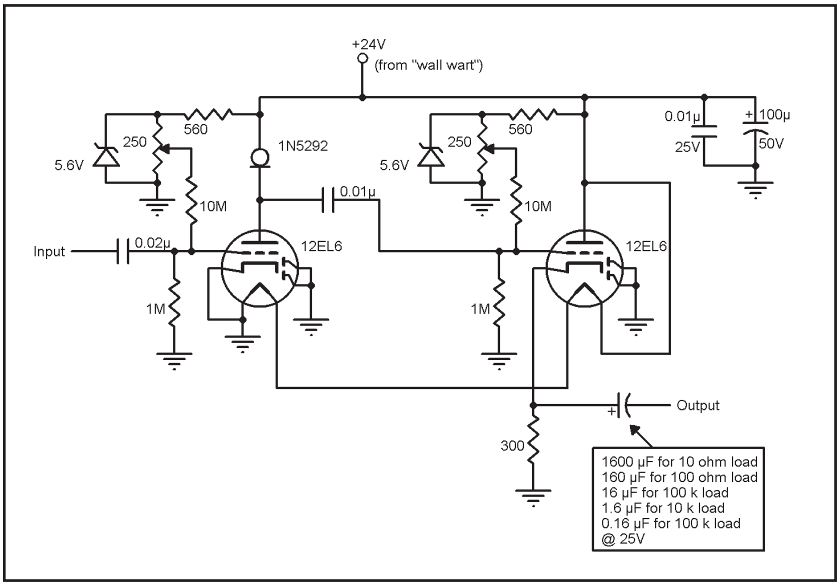


Figure 6: This is the final design for the distortion pedal.

slope, r_p of the cathode follower will still be about 52 kΩ. Thus, the voltage gain of the cathode follower stage is:

$$A_V = \frac{\mu R_k}{R_k + r_p} = \frac{55 \times 300}{300 + 52{,}000} = 0.315$$

giving us an overall gain of 16.4 (51.96 × 0.315). The output impedance is:

$$Z_O = \frac{r_p R_k}{(\mu + 1)R_k + r_p} = \frac{52{,}000 \times 300}{(55 + 1)52{,}000 + 300} = 5.35\ \Omega$$

This value is low enough to drive any practical load. In fact, it would even drive a standard 32-Ω headset or pair of earbuds.

Most of the distortion in this two-stage circuit will be generated in the first stage, since cathode followers are inherently pretty clean, due to the negative feedback resulting from the unbypassed cathode resistor. Since what we usually want is even-order distortion, and cathode followers mainly produce odd-order distortion, a clean cathode-follower stage is a good thing.

Although we can control the distortion by adjusting the bias potentiometer, if we set it for the design control-grid bias of –0.1 V, we can calculate the first stage's second-harmonic distortion, using an input of 170 $m_{VP\text{-}P}$ (60 mV_{RMS}):

$$\%2HD = \frac{\Delta V_{larger} - \Delta V_{smaller}}{2(\Delta V_{larger} + \Delta V_{smaller})} \times 100 = \frac{5.6 \quad 5}{2(5.6 + 5)} = 2.8$$

This is hardly enough for a distortion pedal, but the gain is high enough for the amplifier it feeds to add its own distortion, and 2.8% is enough to add a "tubey" sound even to a solid-state amplifier. Distortion will rapidly increase as input level is increased due to the cutoff occurring on larger negative excursions. Since the distortion products resulting from cutoff are different from grid-current limiting (due to large positive excursions), the waveform remains asymmetrical (mainly even-order) even through the onset of hard clipping.

Figure 6 shows the complete circuit. Notice that we are using a 24-V wall wart as our power supply and connected the filaments in series. If the wall wart is not voltage-regulated, we need to measure its output when the power-line is at 125 VAC. We can also measure the wall wart's voltage and the power-line voltage and apply whatever correction factor is needed to get a 125-V equivalent. For example, if your unregulated "24-V" wall wart's output measures 28.4 V with a 115 V line voltage, it would probably measure:

$$\frac{125}{115} \times 28.4 = 30.87$$

with the line voltage at 125 V. The 12EL6 filament is rated for a maximum of 15 V, although the datasheet has a note saying that up to 15.6 V is okay. This means two in series could have as much as 31.2 V applied.

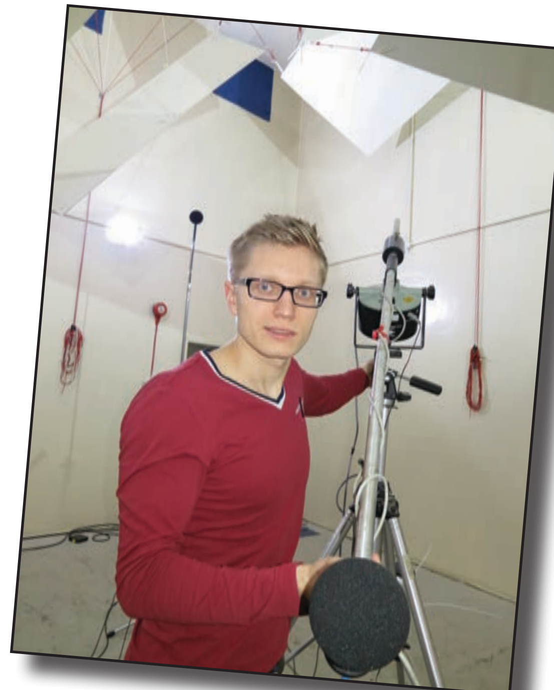
An unregulated wall wart's voltage will drop when current is being drawn, so the no-load voltage you measure will almost always be higher than the rating. If the no-load voltage is higher than 30 V, connect a 200-Ω, 2-W resistor to the output and measure the loaded voltage. As long as the loaded voltage is between 20 and 30 V, you are safe.

Note the two filter/decoupling capacitors from the 24-V line to ground. The 100-µF one is effective at lower frequencies and the 0.01-µF capacitor performs better at high frequencies. Thus, the combination acts pretty much like an ideal 100-µF capacitor.

This design is a "paper" design as it has not been built and tested, and thus may require some changes to produce a working project. Therefore, it is not presented as a DIY project but rather as an example of design principles using a space-charge triode.

Author's Note: I have recently become aware that Alan Kimmel did not develop the Mu Follower as stated in the "Controlled-Source Theory in Hollow-State Design" article (audioXpress, *October 2014). Chris Paul developed the circuit and gave it its name.*

Member Profile



Member Name:
Arthur Marker

Location:
Southampton, United Kingdom

Education:
MSc in Engineering Acoustics from the Institute of Sound and Vibration Research (ISVR), University of Southampton

Occupation:
Arthur is a PhD student at the ISVR, University of Southampton, in the UK.

Member Status:
Audio has been Arthur's main field of interest for many years. However, he said it can be challenging to keep up with current developments. He has found that *audioXpress* helps him get informed about the latest audio news as well as be aware of future trends.

Audio Organization Affiliations:
He is a member of the Audio Engineering Society (AES), the Acoustical Society of America (ASA), the Institute of Acoustics (IOA), the Young Acousticians Network (YAN) of the European Acoustics Association (EAA), and the Engineering Integrity Society (EIS).

Audio Interests:
Arthur said he is interested in electroacoustics, digital signal processing (DSP), echo cancellation (EC), and compensation strategies.

Most Recent Audio Purchase:
His most recent purchase was an external asynchronous DAC, based on an ESS chip.

Current Audio Projects:
Arthur's recent projects involve Acoustic Echo Cancellation (AEC) and a study of nonlinearities in microspeakers.

Dream System:
Arthur was very specific when describing his dream system. He said it would include Sonus Faber Aida loudspeakers, Dan D'Agostino Momentum pre- and power amplifiers, a Denon DP-100 turntable with a Lyra Connoisseur 4-2P SE phono preamplifier, and a Naim NDS network player.

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 (MDA), 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.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