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May 2015 **ISSN 1548-0628**

www.audioxpress.com

audioxpress (US ISSN 1548-0628) is published monthly, at $50 per year for the US, at $65 per year for Canada, and at $75 per year Foreign/ROW, by Segment LLC, Hugo Van haecke, publisher, at 111 Founders Plaza, Suite 904, East Hartford, CT 06108, US Periodical Postage paid at East Hartford, CT, and additional offices.

Head Office:
Segment LLC
111 Founders Plaza, Suite 904
East Hartford, CT 06108, US
Phone: 860-289-0800
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Glass Audio Special—Still Loving Tubes

Here we are in the era of the Apple Watch, wearable technologies, and the Internet of Things (IoT), and still loving and writing about tubes. The Glass Audio Special edition is a tradition we have kept since the former title was merged with other magazines to become *audioXpress*. Not that *audioXpress* doesn't publish tube-related articles every month, but we still feel this is one of the highlights for our readers and it certainly is among our authors due to the amount of articles we receive—increasingly addressing applications outside of the typical high-end living-room stereos.

Personally, my relationship with tubes originates from my earlier contacts with a wonderful Kaiser German UKW radio, which was part of the furniture at my aunt's country house. As a child, while spending summer holidays at her house or a simple weekend visit, this allowed me to listen to FM radio for the first time. The Kaiser radio had this absolutely fascinating sound quality and for me the experience was even more meaningful because at my parent's home, I would typically listen to music on a small transistor radio (with a single ear piece of terrible quality if I wanted to listen to radio at night).

Years later, I convinced my parents to buy our first hi-fi system, a TEAC component package, complemented by a pair of Cabasse tower speakers. Even though the sound quality was dramatically better than anything else we had until then, I would go back to my aunt's place and immediately turn on the Kaiser radio and listen to any type of music, while staring for hours at the warm glowing light emanating from the inside. To make the experience even more unforgettable, not only did that tube radio sounded like nothing else I had heard but it also had five special keys—Klangregister—named 3/D, Jazz, Piano, Sprache (spoken-word) and Orch. These keys would generate completely different experiences when selected. Of course, with the rock music I listened to during those magical FM shows, the 3/D effect was my absolute favorite.

A few years later, I had a similar revelation when I plugged in my bass guitar to a rented Vox AC30 guitar amplifier. For the first time, I felt confident with the sound I was generating, which contributed to my decision to pursue my musical ambitions with a band. Yet, none of the (solid-state) bass amplifiers I subsequently owned provided me with the same rewarding fullness and warmth I'd experienced with that vintage tube amplifier.

Those early experiences and later professional duties, from selling hi-fi to professional audio recording equipment and designing a few FM radio studios and transmission chains, led me to dive into the fascinating world of multiband processing and understand the "FM radio" sound. That also gave me the chance to better understand the many reasons why vacuum tubes are so intrinsically related to music history. From the tone of guitar amplifiers and processing devices used in recording studios, to high-end hi-fi equipment, I learned to admire the nonlinearities of tubes, the saturation and harmonic distortion on the output transformers, and why they have "magical" sonic signatures.

Years later, following a short visit to my aunt's place, she gave me that Kaiser radio. I had recently moved with my wife to our new home and we still have it, contained in a specially designed cabinet custom-made for the living room.

Those fond memories of musical discoveries and my fascination for sound henceforth are directly related and it sounds particularly fit to remember it in tribute to my aunt who sadly passed away this last February.

João Martins
Editor-in-Chief

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COLUMNISTS

Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—which is now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, a monthly publication. Although he has been involved with publishing throughout his career, he still works as an engineering consultant for a number of loudspeaker manufacturers.

Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject while Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike specializes in materials and fabrication techniques to enhance speaker performance.

Steve Tatarunis has been active in the loudspeaker industry since the late 1970s. His areas of interest include product development and test engineering. He is currently a support engineer at Listen, in Boston, MA, where he provides front-line technical support to the SoundCheck test system's global user base.

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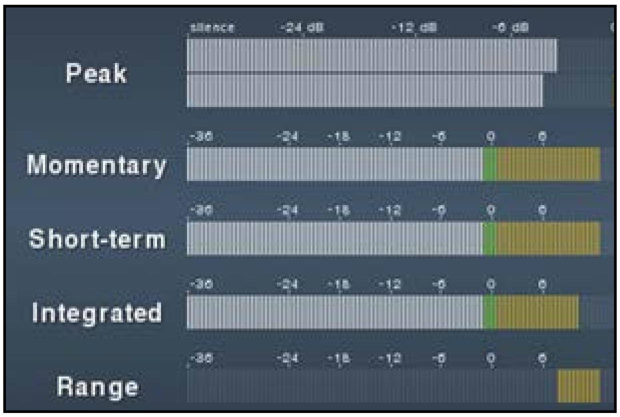

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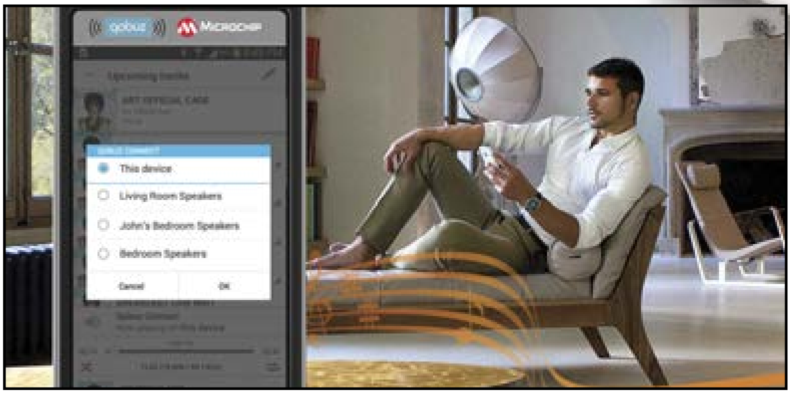

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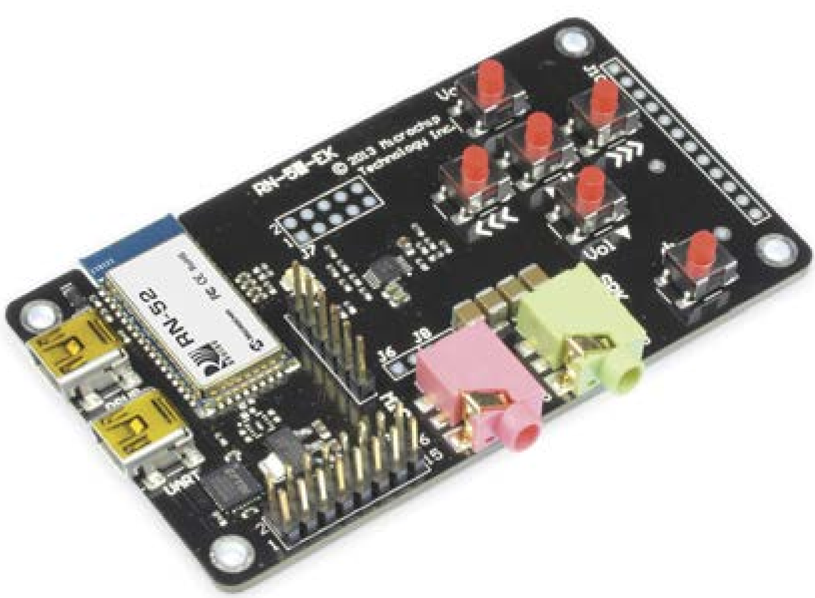
Wireless Audio (Part 8)

Microchip Wireless Audio Platform Portfolio

Concluding this article series on the development platforms currently available for wireless audio, we look at Microchip Technology and its extensive portfolio of solutions. Being one of the largest electronic companies, Microchip expanded its global presence through acquisitions and also became a leading brand for wireless multimedia ecosystems. Microchip offers an extensive portfolio of Bluetooth platforms, Kleernet technology for low-latency applications, and the JukeBlox Wi-Fi/Ethernet platform.

By
João Martins
(Editor-in-Chief)



Prior to the SMSC's acquisition, Microchip already had an extensive portfolio of wireless audio solutions such as this RN52 Bluetooth Audio Evaluation Board with dual-channel audio output/input in analog and digital formats, a built-in amplifier, and support for codecs such as aptX, AAC, and others.

Microchip Technology provides multiple solutions for wireless audio development with an expanded and constantly evolving portfolio, resulting from the company's recent acquisitions.

Microchip's wireless audio solutions include the JukeBlox Wi-Fi connected audio platform, originally developed by BridgeCo. The JukeBlox platform offers support for popular music applications along with standard Ethernet and Wi-Fi interfaces on home networks or the Internet. For Wi-Fi audio applications, Microchip's portfolio naturally includes a range of solutions, including USB transceivers and bridges.

In addition to Wi-Fi solutions, Microchip offers support for Bluetooth-streaming audio based on PIC32 MCUs. These microcontroller-based systems offer the ability to tailor features and functions to any application. Advanced functions include support for multiple phone break-in modes, or additional functions (e.g., graphical displays, audio equalizer functions, or support for USB streaming playback).

Standard Microchip Bluetooth solutions feature support for streaming sub-band coding (SBC) and advanced audio coding (AAC) over Advanced Audio Distribution Profiles (A2DP). The company currently supports most low-cost applications in the consumer markets, such as Bluetooth speakers, consumer music-player docks, noise-canceling headsets, and soundbars with its PIC32MX1/2 range of 32-bit microcontrollers. Apart from offering larger Flash and RAM options and I2S for digital audio and a feature-rich peripheral set, these MCUs are coupled with comprehensive software and tools from Microchip, including support for designs in digital audio and general-purpose embedded control.

Microchip's new MPLAB Harmony software development framework also supports these devices with readily available software packages including Bluetooth audio development suites, audio equalizer filter libraries, various decoders including AAC, MP3, SBC, sample-rate conversion libraries, USB stacks, and graphics and touch libraries.

Microchip's PIC32 Bluetooth Audio Development Kit provides a comprehensive solution to develop Bluetooth A2DP audio streaming solutions and applications. The board is coupled with a Bluetooth HCI Radio Daughter Card that demonstrates a low-cost Bluetooth implementation and an Audio DAC Daughter Card that demonstrates a high-quality 16/24-bit, 32–192 kHz audio conversion/amplification for line or headphones. The kit ships with demo code that enables wireless streaming digital audio from any Bluetooth-enabled smartphone or portable music player or over USB.

The PIC32 Bluetooth Audio Development Kit provides a comprehensive solution to develop Bluetooth A2DP audio streaming solutions and applications. The board is coupled with two daughter cards: the Bluetooth HCI Radio Daughter Card that demonstrates a low-cost Bluetooth implementation and the Audio DAC Daughter Card that demonstrates a high-quality 16/24-bit, 32-192 kHz audio conversion/amplification for line or headphones.

The Kleernet Alternative

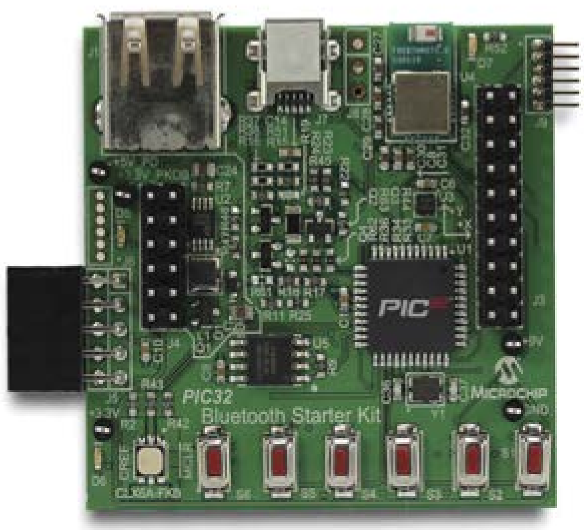
Microchip differs from other companies already discussed in this article series, because, apart from Bluetooth and Wi-Fi solutions, it also features another important wireless audio alternative with its Kleer radio frequency (RF) technology.

SMSC acquired Kleer Semiconductor (originally from Cupertino, CA) in 2010 after the company had already established important relationships with major consumer electronic brands. In 2012, SMSC was acquired by Microchip, which currently licenses the technology rebranded as Kleernet.

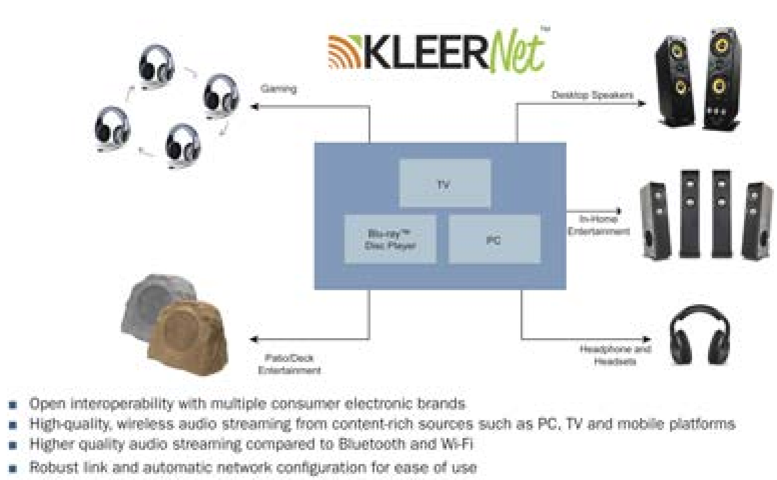
The Kleernet platform enables a high-quality, uncompressed, low-latency Bluetooth alternative for wireless distribution of digital content to headphones, speakers, and other audio devices. It supports triple-band transmission on 2.4, 5.2, and 5.8 GHz. Current platforms are able to simultaneously stream a signal to four wireless audio receivers.

Kleernet is a complete hardware and software solution enabling Kleernet-compatible products to recognize and pair with other Kleernet products. The platform is a robust solution for home-theater applications, soundbars, headsets, gaming, remote controllers, and even televisions.

The PIC32MX270F256D Bluetooth Audio PIM is designed to demonstrate the capabilities of the PIC32MX1*xx*/2*xx* family of devices using the PIC32 Bluetooth Audio Development Kit.



Microchip's PIC32 Bluetooth Starter Kit is a low-cost Bluetooth development platform featuring the PIC32MX270F256D MCU. The starter kit offers an Android App, Demo code, and Bluetooth Serial Port Profile stack for free.



Microchip's KleerNet technology allows for low-latency wireless audio connectivity in multiple RF bands and frequencies within each RF band (2.4, 5.2, and 5.8 GHz), making it well-suited to effectively manage gaming and VoIP headsets, wireless headphones and microphones, wireless rear and standalone speakers, and wireless soundbars and subwoofers within a short range (50′ maximum).

After some earlier support from Sony and Sennheiser, who used Kleer RF technology for wireless headsets targeted mainly at home-theater applications, Kleer was also adopted by Hewlett-Packard (HP) in its Wireless Audio adapter, which was complemented by a dedicated USB transmitter and directly supported by selected HP laptops. HP introduced its Kleernet solutions promoting the fact that they would be directly compatible with products from other companies such as Monster, Astro Gaming, and Boston Acoustics, which also adopted the technology.

Boston Acoustics made a clever implementation of Kleernet for multichannel home-theater applications. The company uses the RF technology to connect multiple satellite speakers since Kleernet works better in single rooms and relatively short distances—at maximum 50′ (15 m), without walls.

Monster's implementation for its Streamcast HD connectivity range was also the first wireless audio solution from the brand and was sold as a transmitter and receiver kit for $189.95. Since the idea was to send four separate channels (streams) of music to up to 12 rooms (zones), which are then controlled using a smartphone or tablet StreamCast App. However, several additional receivers were necessary (at $99.95 each). Later, Monster started to offer Bluetooth aptX Streamcast HD alternatives.

Monster also sold its ClarityHD Model One speaker with a built-in Kleer receiver, but the system cannot compete with existing Wi-Fi-based solutions. Clearly, Kleernet is a much better choice for wireless headphones and typical home-theater applications.

Kleernet Today

Kleernet has now reappeared in a radical new approach from HearNotes, a new company that introduced the WireFree Earbuds concept at the 2015 International Consumer Electronics Show (CES).

HearNotes opted for Kleer wireless audio technology to differentiate its new product in the highly competitive headphone industry. In its promotional material, HearNotes states that Kleer technology "transmits uncompressed wireless audio that is superior to Bluetooth. The HearNotes WireFree earbuds are able to receive audio at a range of 25′ to 50′ from the transmitter and offer four-plus hours of playtime. Also, there's no need for clumsy Bluetooth 'pairing' procedures, and there are no latency issues or signal interference."

As illustrated by existing implementations, the drawback with implementing Kleernet has to do with the need to provide both the transmitter and the receiver as an integrated hardware solution, contrary to the wide interoperability of standard Bluetooth or Wi-Fi available everywhere.

The HearNotes WireFree package includes a pair of WireFree earbuds with an integrated receiver and a hardware transmitter that plugs into any 3.5-mm headphone jack, along with a charging pad that inductively charges both the earbuds and transmitter (and also serves as a protective carrying case).

Kleernet enables streaming non-compressed

Hewlett-Packard adopted Kleernet wireless technology in its laptops, which is complemented with this Wireless Audio adapter and a USB transmitter.

digital audio for up to four receivers from the same transmitter. The Kleer radio protocol has also been extensively tested to avoid interference from wireless local area networks (WLANs), Bluetooth, digital enhanced cordless telecommunications (DECT), and so forth, requiring only 3 MHz in the 2.4-GHz band for CD-quality audio. This means it can use channels that are between the bands that WLANs use. It can also use channels at the edge of the band that are not used by WLANs. Tests have also shown that there is no loss of audio with Kleernet, even if there are three WLANs using all available nearby channels.

According to HearNotes, a Kleer-based earbud will also be smaller and lighter than one using Bluetooth technology. This is because the Bluetooth radio needs more power and has to use a larger, heavier battery. "A Kleer-based wireless earbud can operate for 10 hours with an 80-mAhr coin cell type battery. A Bluetooth radio will operate for less than 1 hour with this size coin cell," HearNotes explains.

Now as part of Microchip's portfolio, Kleernet has leveraged seven generations of wireless audio technology and provides reference modules in multiple form factors to simplify system integration and product development. The same transceiver module can be used as either the source (transmitter) or speaker (receiver), depending on the software configuration. A software development kit (SDK) and graphical user interface (GUI) reference package for PCs is also available, enabling simple software implementation and system development for PC audio streaming applications.

For Kleernet implementation, Microchip currently offers an open ecosystem, including the DWAM83 digital wireless audio module, based on Microchip's DARR83 baseband processor chip. The solution enables flexible product design with support for the three RF bands and features dual PCB antennas supporting Tx and Rx diversity, an integrated microcontroller, digital audio interfaces, and an automatic Sample Rate Converter.

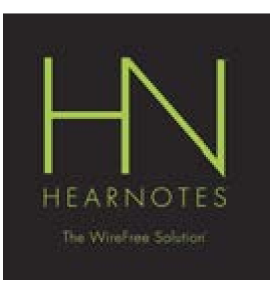
There's also Microchip's DWHS84 wireless audio reference design, based on the DARR84 audio processor chipset, specific for headset and headphone application design, with a complete software stack.

The DWPCIe83 USB module is a digital wireless audio module with a PCI Express interface based on the Microchip DARR83. This USB module can be used to build an uncompressed wireless digital audio transceiver operating in the three bands. The DARR83 chip provides the basic functions of audio processing and buffering, a data link layer, and a physical layer.

Finally, the DWUSB83 USB dongle is a digital wireless audio module with a USB 2.0 interface based on the Microchip DARR83, used to build an uncompressed wireless digital audio transceiver.

The SMSC Acquisition

An article describing the history of Microchip's acquisitions during the last five years would take



HearNotes is a recent example of a new company leveraging the Kleernet technology for its WireFree earbuds, one of the clear audio innovations at the 2015 International Consumer Electronics Show (CES).

more space than this column allows. However, it is clear Microchip has been expanding its portfolio and market reach with a clear interest in wireless technologies.

Just recently, Microchip acquired ISSC Technologies Corp., a leading company on low-power Bluetooth and advanced wireless solutions for the Internet Of Things (IoT) market. ISSC is headquartered in Hsinchu, Taiwan (Hsinchu Science Park) with customer service and research activities in Shenzhen, China, and Torrance, CA.

Yet, its most important and strategic acquisition on the wireless audio segment was undoubtedly Standard Microsystems Corp. (SMSC), which Microchip announced in 2012. Founded in 1971 and based in Hauppauge, NY, SMSC was a leading semiconductor company, strong in areas such as embedded controllers, USB, Ethernet, flash-memory card reader products, among many other connectivity, networking, and control solutions for various applications. And, SMSC had already acquired Kleer Semiconductor in 2010.

In 2011, almost a year before the Microchip acquisition, SMSC acquired BridgeCo, a leading company in networked audio technologies. That acquisition brought the JukeBlox technology to Microchip's portfolio and placed it decisively on the map with regard to wireless audio development.

Originally headquartered in Zurich, Switzerland (where its main engineering resources are still located), BridgeCo expanded to Los Angeles, CA, in 2005 and actively developed Digital Media Player silicon and middleware technologies, docking stations, and Firewire/1394 audio interface systems. In 2006, there were more than 250,000 Firewire devices, using BridgeCo's BeBoB platform, in the market.

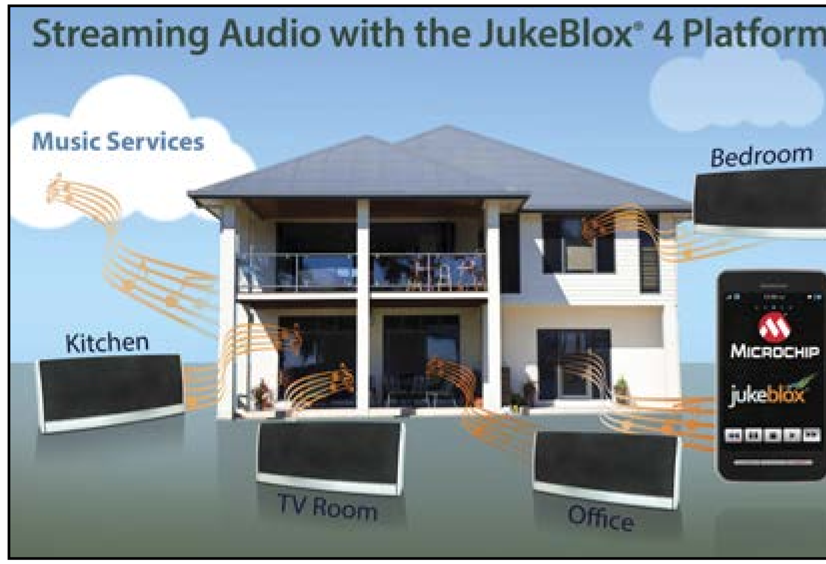
Underlying all BridgeCo solutions are proprietary, integrated semiconductor and software technologies with support for networking and signal-processing standards and protocols, including IEEE 802.11/Wi-Fi, IEEE 1394, and UPnP.

Since 2006, BridgeCo offered the industry's only system-on-a-chip (SoC) and SDK for networked audio. BridgeCo was the first to support wireless streaming not only with local storage but also from cloud-based music libraries and Internet radio.

BridgeCo's line of ARM-based multi-core network media processors integrated a system controller, real-time audio processing engine, DRM security processor, network I/O processor and a Wi-Fi MAC/Baseband to enable high-bandwidth wireless audio streaming.

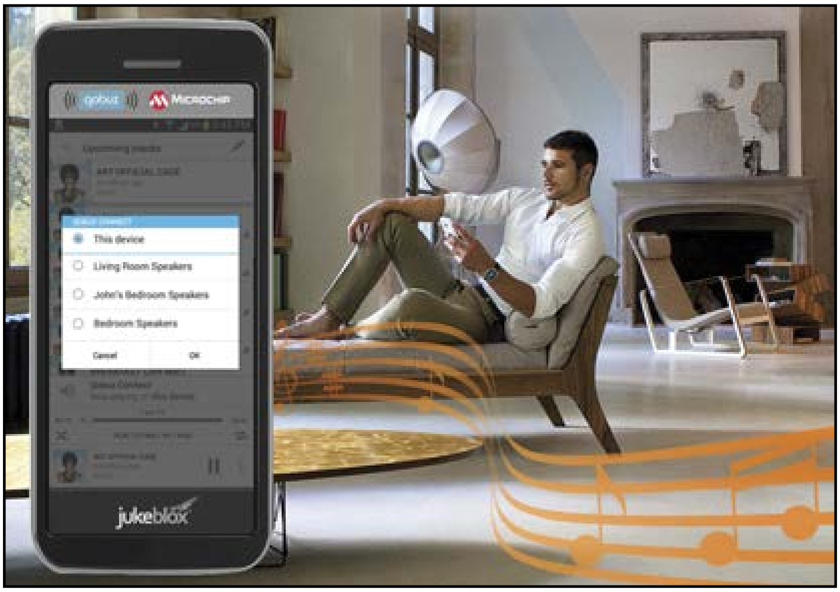
Specifically for streaming audio, BridgeCo developed the JukeBlox platform, including its

Following SMSC's acquisition by Microchip, the company announced a new JukeBlox Software Development Kit (SDK) and the CX875 Wi-Fi Network Media Module, which provided the easiest and most cost-effective solution for creating AirPlay-compatible wireless audio streaming systems.



The JukeBlox platform was an early solution in the streaming-audio market and has been adopted by more audio brands than any other platform available for Wi-Fi audio connectivity. Engineering support is still provided by BridgeCo's engineering team, out of Switzerland.

Demonstrating the upgrade flexibility of its JukeBlox platform, Microchip released recently the JukeBlox 4 (fourth generation) with support for Spotify Connect direct streaming, together with several key improvements.



Audio devices powered by Microchip's JukeBlox platform, such as wireless speakers and AV receivers, are able to support lossless audio services such as French online music service Qobuz Hi-Fi (available in Europe) with support for both iOS and Android Mobile Devices.

network audio processor with integrated Wi-Fi, and software that allowed access to music libraries and playlists on or through smartphones, tablets, Macs, and PCs and to directly stream it to any enabled home-audio product.

Its JukeBlox Wi-Fi platform, combined wired or wireless audio streaming via multiple standards, provided USB functionality, a suite of codecs, protocols, and DRM standards, apart from AM/FM radio and clock/alarm functionalities. It also enabled music streaming to virtually all home-audio equipment, including home-theater systems, AV receivers, radios, wireless speakers, and portable music player docking stations. The JukeBlox solution was also designed to support multitasking capabilities with the latest operating systems, allowing music to wirelessly stream throughout the home, while keeping smartphones and tablets free for other activities.

JukeBlox was adopted by some of the largest consumer electronics brands in the world including Pioneer, Philips, Denon, Marantz, Bowers & Wilkins, and Harman (JBL and Harman Kardon). And, SMSC was very aware of the market potential in wireless audio applications. Music streaming from smartphones and tablets, to Wi-Fi home networks, was booming and BridgeCo played an important role in driving the adoption of those ecosystems with interoperable hardware and software solutions. With the acquisition of BridgeCo, SMSC (and consequently Microchip) became an even larger partner to the whole audio industry.

BridgeCo's technology was developed to ensure interoperability across multiple operating systems and existing or emerging standards, creating a clear alternative to the Sonos ecosystem, and was crucial to Apple's AirPlay development, which received important contributions from BridgeCo's development team.

BridgeCo was not only chosen by Apple as a development partner for AirPlay but also allowed the company to pursue implementation services for other audio companies interested in having AirPlay compatible solutions. BridgeCo's acquisition by SMSC and SMSC's subsequent acquisition by Microchip Technology is undoubtedly an important moment in wireless audio history.

Steve Sanghi, Microchip's President and CEO, said that SMSC's "smart mixed-signal connectivity solutions aimed at embedded applications are an ideal complement to Microchip's embedded control business," and specifically highlighted its contribution to expand Microchip's business in "the automotive, industrial, computing, consumer and wireless audio markets."

JukeBlox Technology

Recently upgraded to its fourth generation, JukeBlox SoC is the foundation for the main wireless audio development platform currently offered by Microchip.

The Jukeblox SDK delivers multi-speaker

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streaming (whole-home audio), direct support for popular streaming music services (cloud music services), cross-platform interoperability (iOS/Android/Windows 8), Apple AirPlay, DLNA protocols, and support for a range of audio codecs. Direct streaming frees the mobile devices 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. It also offers simplified connection to a Wi-Fi audio device with the simplicity of Bluetooth and simplified network setup.

Leveraging all the advantages of Wi-Fi to simultaneously stream audio to multiple speakers for whole-home audio applications and benefiting from the evolution of Wi-Fi standards, especially in terms of extended coverage, the JukeBlox platform also supports lossless full-resolution quality audio.

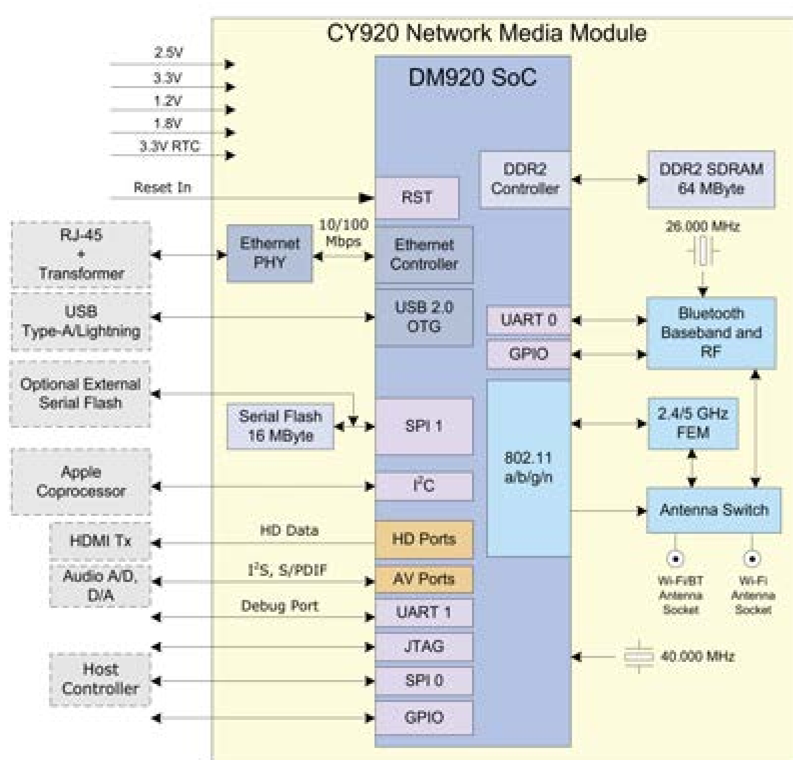
According to Microchip, more than 8 million audio products have been shipped based on these network audio processors. Those solutions are also firmware upgradable, as demonstrated in 2014 with the introduction of Spotify Connect and Qobuz Connect lossless support to existing products based on Microchip's CX870 Wi-Fi modules and DM860 Ethernet processor.

The recent JukeBlox 4 SDK update, in combination with the CY920 Wi-Fi and Bluetooth Network Media Module, expanded support for multizone/multiroom functionalities. The fourth-generation JukeBlox SDK now offers next-generation dual-band Wi-Fi technology, simultaneously supports play of a single audio stream on multiple speakers with tight audio synchronization device discovery (reducing network bandwidth use), dynamic linking/pairing and shared control, as well as more integrated streaming music services.

The certified CY920 Wi-Fi and Bluetooth Network Media Module is based on Microchip's new, low-cost DM920 Wi-Fi Network Media Processor, which integrates 2.4 GHz and 5 GHz 802.11a/b/g/n Wi-Fi, high-speed USB 2.0 and Ethernet connectivity. The DM920 Wi-Fi Network Media Processor also features integrated dual 300-MHz DSP cores that can reduce or eliminate the need for costly standalone DSP chips.

The integrated DSPs utilize a GUI-based development tool with built-in algorithms with integrated Bluetooth software and radio without the need of an external host processor, and cost-saving features to lower the bill of materials on new products. An easy-to-use, PC-based GUI simplifies the use of a pre-developed suite of standard speaker-tuning DSP algorithms, including a 15-band equalizer, multiband dynamic range compression, equalizer presets, and various filter types.

As Microchip can rightfully claim, thanks to BridgeCo's earlier lead, the JukeBlox platform was an early pioneer in the streaming-audio market and has been adopted by more audio brands than any other platform available for Wi-Fi audio connectivity.



Microchip offers two families of modules specifically designed for Wi-Fi speakers and AV receivers. The CY920 Module is the most recent one and supports 802.11 a/b/g/n Wi-Fi, 2.4 and 5 GHz, diversity antennas and optional Bluetooth. The DM920 SOC is complemented with integrated dual DSP cores.

Signal Dynamic and Loudness

This is the sixth article in our multi-part series that takes readers on a continuing quest for realistic recorded sound. The series, which began in the December 2014 issue with mono, has been more-or-less linear—in time and in technological advancement. But from here, the path branches into different approaches to realism at similar times. Also, some of these methods overlap, but I will sort them out in due course, so please stay tuned.

By
Ron Tipton
(United States)

Realism, or our impression of it, depends on our listening environment: the size, the shape, and the acoustics of our listening room or home theater. Free and low-cost software is available to help you design a "good" listening room. And, other software can help you analyze and optimize a room you already have. There are systems that claim to automatically optimize the sound to your room, but do they work? And, if so, are they worth the cost? For now, we will focus on other characteristics.

Loudness

A live performance is by definition realistic, the yard-stick against which a recording of it is measured. Is playing the recording also realistic? In some cases, I think it is not. A musical track's dynamic compression (the "loudness wars") is an interesting topic in realistic reproduction.

If you think recorded (and live) music has gotten louder through the years, you are correct because it has. Dynamic compression, using either hardware or software, has increased the number of high-amplitude peaks but the dynamic range has not necessarily changed much. This is graphically illustrated in **Figure 1**.

I'm a Neil Diamond fan and I have many of his albums—from 1970s LPs to a 2014 CD (*Melody Road*)—so I mostly used his songs to create **Figure 1** and **Table 1**. To show there wasn't much change in compression even in the 1970s, I included an Artie Shaw number recorded in 1937. According to Hugh Robjohns in *Sound On Sound*, "the 'volume' of pop, rock, and other music recorded and released on commercial CDs has risen steadily since the late 1980s, with a corresponding reduction in dynamics and, in many cases, a trend towards a more aggressive and fatiguing sound character…"

Positive Peak Values

Figure 1's data curves show the amplitude distribution of positive peak values. The 1937 and the 1971 curves are nearly coincident while the 2014 curve exhibit a greater number of high peaks. To generate **Figure 1**, I wrote a computer program named dynamics.exe (in the C language) for the PowerC compiler. The exe file, source code and the needed library files are included in dynamics.zip in the Supplementary Material found on the *audioXpress* website (see Resources). I began using the PowerC compiler in about 1990 and it's still available for less than $20.

To use, enter the name of a mono .wav file and the program computes the maximum positive peak, average positive value, and the peak distribution using 10 amplitude "bins." The results are displayed on the monitor screen and can be optionally printed. It's an MS-DOS program, but I have tested it in Windows 2000 and the 32-bit versions of XP and 7. (The program won't run in the 64-bit versions.) The screen is properly formatted with response entry boxes accomplished with the inclusion of the ansi.sys file. Instructions on where to put this file for the different Windows versions are included in dynamics.zip. The input file must be mono but it's easy to select and save a single channel of a stereo or multichannel file with almost any audio editor.

Loudness Meter Plug-Ins

I evaluated several VST Loudness Meter plug-ins (see the Loudness.pdf in the Supplementary Material) to compare their readings with my results from dynamics.exe. **Table 1** shows the results using the free MLoudness Analyzer (which I think is an excellent analyzer compared to the others I tested, especially because it's free) for the same .wav files used for **Figure 1**. The numbers in the Integrated Loudness column correlate nicely with the dynamics program. The true peak and the range are also displayed.

True Peak shows the true peak level which may not be the same as the peak shown on an audio monitor because the analog-to-digital conversion and filtering may miss an actual peak value. The plug-in simulates the conversion process with a higher sampling rate to catch a true peak. This is important because if the true peak is higher than full-scale (0 dB) clipping distortion is produced. This is especially important when upward dynamic expansion is done because it increases the signal's amplitude, think of the process as "stretching" the signal that is above the set threshold by the expansion ratio. Thus, it is necessary to attenuate the amplitude by 3 to 6 dB before expansion and then renormalize to no more than 0 dB, which I will discuss in more detail later.

Range is a measure of the signal's dynamics but it's unclear how it is measured because the reported numbers appear low compared to the maximum positive peak divided by the average positive value as calculated by dynamics.exe. In general, with this analyzer, minimal dynamic compression is indicated by an integrated value below 5 to 6 and a range above 7.

Dynamic Compression

More dynamic compression is needed for live concerts (especially in a large outdoor venue) than is needed in your listening room or home theater because these locations are quieter. Why then are recordings of live outdoor concerts marketed with the same compressed dynamics? Perhaps one reason is to retain a "true" concert rendering. However, I find this degree of compression tiring and maybe even boring when listening in an environment where such compression isn't needed. Case in point: the live Neil Diamond "Hot August Night" concert at the Greek Theater in Los Angeles, CA, in 1972. I love the music but not so much the presentation: All the tracks sound the same. I have included a graph named HotAugust.pdf in the

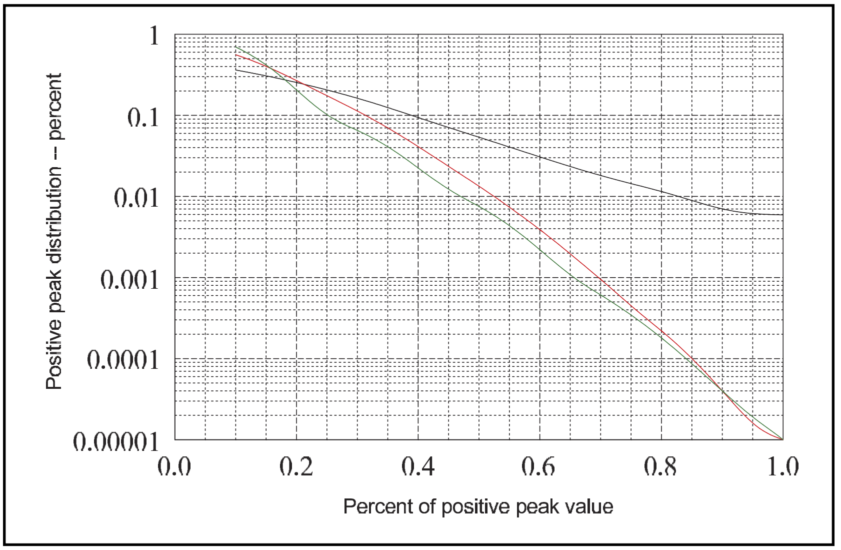


Figure 1: This graph compares the recorded music loudness from 1937 to 2014. The black curve is from the 2014 *Melody Road* CD, the red curve is from the 1971 LP *I Am…I Said*, and the green curve is Artie Shaw's "That's Why the Lady is a Tramp" recorded in 1937. (See the References in Table 1 for the complete discography.)

.wav File Name	Reference	Integrated Loudness	Range	True Peak
"Melody Road"	A	10.6 dB	4.7	4.2
"If You Go Away"	B	8.4 dB	8.5	1.4
"Stones"	B	8 dB	4.9	1.4
"I Am…I Said"	B	6.9 dB	9.1	1.4
"That's Why the Lady Is a Tramp"	C	3.3 dB	9.1	–0.18
"I'll Be With You in Apple Blossom Time"	C	2.8 dB	8	0

Table 1: The loudness measurements for various songs were recorded using the MLoudness Analyzer VST plug-in. The song from Reference A came from the Neil Diamond's *Melody Road* CD, Capitol Records, B002176002 (2014). The songs from Reference B came from Neil Diamond's *Stones* LP, MCM Records, Stereo 93106 (1971). The songs from Reference C came from *The Complete Artie Shaw and the Rhythmakers,* recorded in 1937 and re-released on the Swingdom Records LP, Mono SW-7005-4.

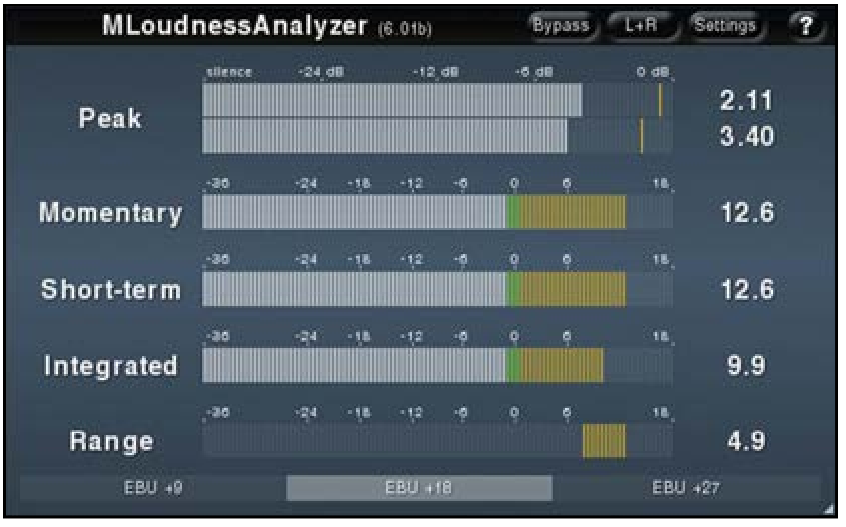


Photo 1: The free MLoudness Analyzer VST plug-in performed as well or better than any of the "not-free" plug-ins I evaluated. My evaluation of eight plug-ins is included in Loudness.pdf in the Supplementary Materials.

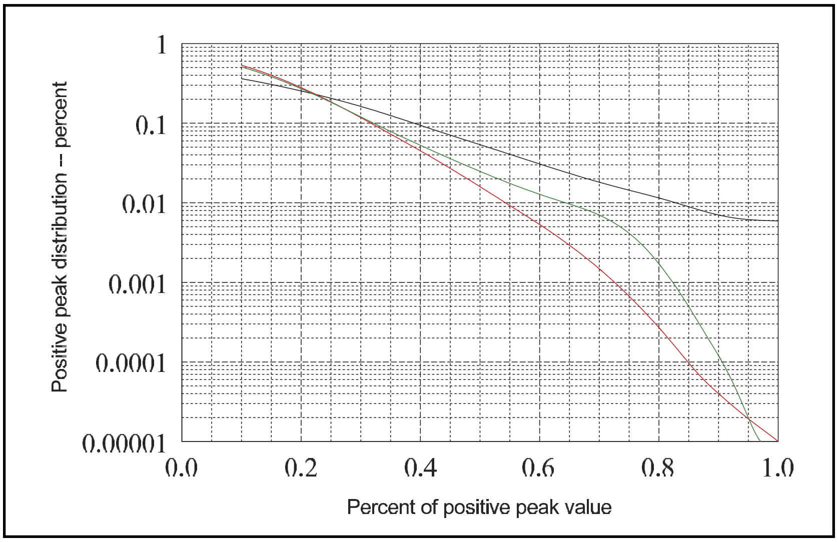


Figure 2: This graph compares two dynamic expansions of the *Melody Road* CD track to the original (black curve). The red curve was produced by the MLoudness Analyser plug-in and the green curve by the free FloorFish plug-in.

Table 2: These are the MLoudness Analyzer VST plug-in settings for dynamically expanding the "Melody Road" CD track. The track amplitude was attenuated −6 dB before expansion and then normalized to 0 dB after expansion.

Control Name	Setting
Threshold	−18 dB
Low gain	0 dB
High gain	0 dB
Ratio	2
Attack	0.32 mS
Release	19mS
Knee	Soft
Crossover	6 dB/octave
Lookahead	Off
Mode	Clean
Output	0 dB

Photo 2: The free FloorFish VST-plug-in made a noticeable dynamic improvement in the original "Melody Road" track. I looked at nine plug-ins said to do dynamic expansion, but I couldn't get all of them to work. My FloorFish settings are described in the Supplementary Materials as is my experience with the other plug-ins.

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.

Supplementary Material that illustrates the dynamics of three typically compressed tracks from the concert.

This begs the question: Can dynamic compression be "undone"? I've been experimenting with this, and to a satisfactory degree, it can. Doing a "perfect" job is nearly impossible without knowing the original compression parameters, which are seldom available, so trial-and-error is needed to do a musically pleasing job.

There are two types of dynamic expansion—downward and upward. In downward expansion, the signal is attenuated when it drops below the threshold and increases the perceived dynamics around the threshold. Upward expansion is the opposite: gain is added when the signal goes above the threshold so this increases the dynamics above the threshold. Some software does one or the other and some do both, depending on which one is selected.

By chaining an expansion plug-in with a loudness meter plug-in you can monitor real-time changes in the expansion settings. When the output looks and sounds better in the preview mode, you can run the expansion and then check the results with the dynamics program.

Figure 2 is a graph of the dynamics.exe print-out for the original and two dynamically expanded versions of the Melody Road track. The black curve corresponds to the original compressed file. The red curve is from an expansion with the Multi-Dynamics plug-in using the settings in **Table 2** and the green curve is from the free FloorFish plug-in (see **Photo 2**). The settings can be found in the Supplementary Materials, which also includes one-minute segments of the original track and the two dynamic expansions.

Next month, we will focus on classical binaural recordings.

Project Files

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

Resources

Hot August Night CD, MCA Records, 088-112330-2 (2000). Also available as a higher definition download from www.hdtracks.com.

H. Robjohns, "The Loudness Wars," *Sound on Sound*, February 2014, www.soundonsound.com.

Sources

MLoudness Analyzer VST plug-in
MeldaProduction | www.meldaproduction.com

PowerC compiler
Mix Software, Inc. | www.mixsoftware.com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True Bass Rides Again (Part 2)

The Power Question

In the first in a two-part series, I described my efforts to recreate the true bass sound I once achieved in a former home. In this article, we discuss how to construct and attain the power needed to complete that sound system.

By
Thomas Perazella
(United States)

There is no doubt that high levels of very clean amplifier power are available at reasonable costs in the current market. That has allowed the successful use of speakers with lower sensitivity. Even with these advances you can rapidly reach a point where the need for amplifier power becomes prohibitive. If you stay out of the area of driver compression, for every 3 dB of additional acoustical output, you need double the electrical power. So if one driver is 3 dB more sensitive than another, it can produce the same acoustic output as the other with only half the power.

Using multiple drivers can help in the sensitivity issue. However, there is some snake oil that frequently pops up in this area. The often-quoted figure of a 6-dB increase by paralleling two of the same drivers is somewhat misleading as it assumes that the amplifier can deliver twice the power into the resulting half impedance. There are few amplifiers that can double the available power for each halving of the load impedance. Even if an amplifier could achieve the doubling, the power would in reality double, resulting in only a 3 dB increase per actual watt. That is an improvement but not as much as appears at first glance.

This situation is aggravated by the common practice of using a sensitivity figure based on 2.83 V being delivered to the speaker instead of 1 W. The problem is that voltage level would represent 1 W for an 8-Ω load, 2 W for a 4-Ω load, and 4 W for a 2-Ω load. In this case, a 2-Ω driver would have an apparent sensitivity 6 dB higher than an 8-Ω driver, which is not true because it is actually receiving four times the power at that voltage. Beware of creative "specsmanship." The availability of good relatively low-cost amplifier power is a plus, but you should take care in identifying how much power you will need, given the real sensitivity of the selected drivers. Also remember that sensitivity is specified at a distance of 1 m. Most people listen at greater distances so the levels at the listening position will be correspondingly lower.

Ultimax UM12–22 12″		Titanic MK 4 12″		Dayton DVC15 15″		Dayton DVC15 15″ (Two parallel)	
Acoustic Output (Decibels)	Power (Watts)	Acoustic Output (Decibels)	Power (Watts)	Acoustic Output (Decibels)	Power (Watts)	Acoustic Output (Decibels)	Power (Watts)
84 dB	1 W	86.6 dB	1 W	90 dB	1 W	93 dB	1 W
87 dB	2 W	89.6 dB	2 W	93 dB	2 W	96 dB	2 W
90 dB	4 W	92.6 dB	4 W	96 dB	4 W	99 dB	4 W
93 dB	8 W	95.6 dB	8 W	99 dB	8 W	102 dB	8 W
96 dB	16 W	98.6 dB	16 W	102 dB	16 W	105 dB	16 W
99 dB	32 W	101.6 dB	32 W	105 dB	32 W	108 dB	32 W
102 dB	64 W	104.6 dB	64 W	108 dB	64 W	111 dB	64 W
105 dB	128 W	107.6 dB	128 W	111 dB	128 W	114 dB	128 W
108 dB	256 W	110.6 dB	256 W	114 dB	256 W	117 dB	256 W
111 dB	512 W	113.6 dB	512 W	117 dB	512 W	120 dB	512 W
114 dB	1,024 W	116.6 dB	1,024 W	120 dB	1,024 W	123 dB	1,024 W

Table 1: Here we compare the power input vs. the acoustic output.

Determine Relative Power Requirements

A convenient way to determine the relative power requirements of different drivers is to create a chart of acoustic level vs. input power for the drivers I examined (see **Table 1**). For each driver, there are two columns. The first is the acoustic output and the second is the power input necessary to achieve that output. In addition, that same scenario is repeated for the two 15″ drivers that I used to show the advantage of multiple drivers in parallel. A good example is to look at how much power is required to reach 102 dB.

Looking at the Ultimax, 64 W is needed for a level of 102 dB. Because of the Titanic's higher sensitivity, only about 32 W is needed. For the 15″ Dayton, it can get by with 16 W. When paralleled, a measly 8 W is required. Considering the ear sensitivity at lower frequencies, over 110 dB is not an unreasonable expectation from a subwoofer. Re-examining the three drivers, to achieve 114 dB, you would need 1,024 W for the Ultimax, approximately 512 W for the Titanic, and 256 W for the 15″ Dayton. Now, the numbers really start to hit home. At some point, low sensitivity becomes the 800-lb gorilla in the corner. If you are going to use EQ to flatten the response below resonance, the situation gets even worse.

Building the Boxes

Information on assembling the 2-ft^3 boxes for the 12″ drivers is provided on the Parts Express website (www.parts-express.com). The assembly was very straightforward.

The 5-ft^3 boxes were a little more complicated because I decided to make them asymmetrical allowing me to put them back to back, if desired. In addition, I rear mounted the drivers for a cleaner look so I made the back panels removable. Instead of plate amplifiers, I decided to power them with an external amplifier. I removed the plate amps in the existing subwoofers and filled the holes with

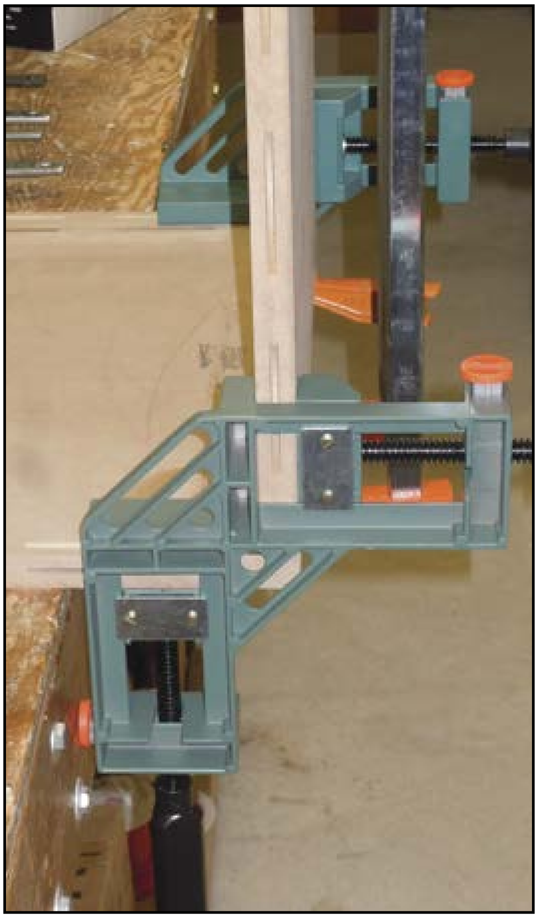

Photo 1: Corner clamps maintain true right angle joints.

Photo 2: The rear view of the input dish shows the connectors, the switch, and the wiring.

0.75″ MDF patches. Making two additional same-sized boxes to add to the existing boxes was easier than building two new 10-ft^3 boxes. It also made it possible for one person to move them with the drivers installed and provided flexibility in placement.

Another decision that added flexibility was to provide switching, enabling the use of the two 8-Ω voice coils of each 15″ driver either in a series or in a parallel configuration. That provided a choice of 4- or 16-Ω operation. If a subwoofer is independently used, the 4-Ω configuration is a better choice. If two are used in parallel, the 16-Ω setting is better. Paralleled, the impedance is then 8 Ω, but with the 3-dB gain in sensitivity, the resulting sound pressure level (SPL) is the same but with double the volume displacement capability.

I used 0.75″ MDF assembled with biscuits and Titebond II wood glue for the construction material. This method resulted in strong airtight joints, which I have discussed in previous articles and will not repeat here. One tool worth mentioning is a corner clamp that I purchased from Harbor Freight. It is a low-cost convenient way to hold adjoining pieces at right angles while the glue dries (see **Photo 1**).

Photo 3: The hanger bolt is installed in MDF near the driver opening.

To provide connectivity to the subwoofers, I chose Speakon connectors. They are the standard speaker connection for professional applications and have the advantage of being a locking connector with no exposed electrical connections. I decided to mount the subwoofers behind my large planar arrays with limited access so I could not afford to have loose connections.

To mount them, I used some metal input dishes specifically made with mounting holes for the connectors. They come in different configurations. I used the one that had openings for two connectors. I could then configure an "in and out" scheme connecting the two subwoofers on each side to one run of speaker cable. I drilled holes in the plates to mount the double pole, double throw switches needed for impedance conversion. **Photo 2** shows the rear of the input dish with the connectors and switch mounted and wired for later assembly into a finished cabinet.

Holes in the box pieces for the input dishes and drivers were cut before assembly. I used a sabre saw to cut the dishes and a Jasper Circle Jig Model 200 for the driver holes. I chose hanger bolts, which combine wood and machine screw threads on a single shaft, to mount the drivers. This enabled me to drive the hanger bolt into the MDF and use machine nuts and washers to secure the drivers. **Photo 3** shows one of the hanger bolts near a driver opening.

To assemble the panel, I applied glue into the biscuit slots on both sides of the pieces to be joined and inserted the biscuits, ensuring the mating surfaces (including the sides of the biscuits) were also covered with glue. Next, I tightly clamped the parts together and wiped up any excess glue that had been forced out of the joints. I always let the glue dry for 24 hours before moving on to the next pieces. The more clamps you have, the more pieces you can glue at the same time.

The main part of the box consisted of the four sides and a front plate. Since the back was removable, I cut the back piece about 0.125″ (1/8″) smaller in each dimension for an easy fit. To keep the back located and secure, I glued four strips of MDF into place at an appropriate depth inside the box's rear opening to act as a mating surface for the back plate. I drilled recessed clearance holes into the back plate and matching pilot holes into the support strips.

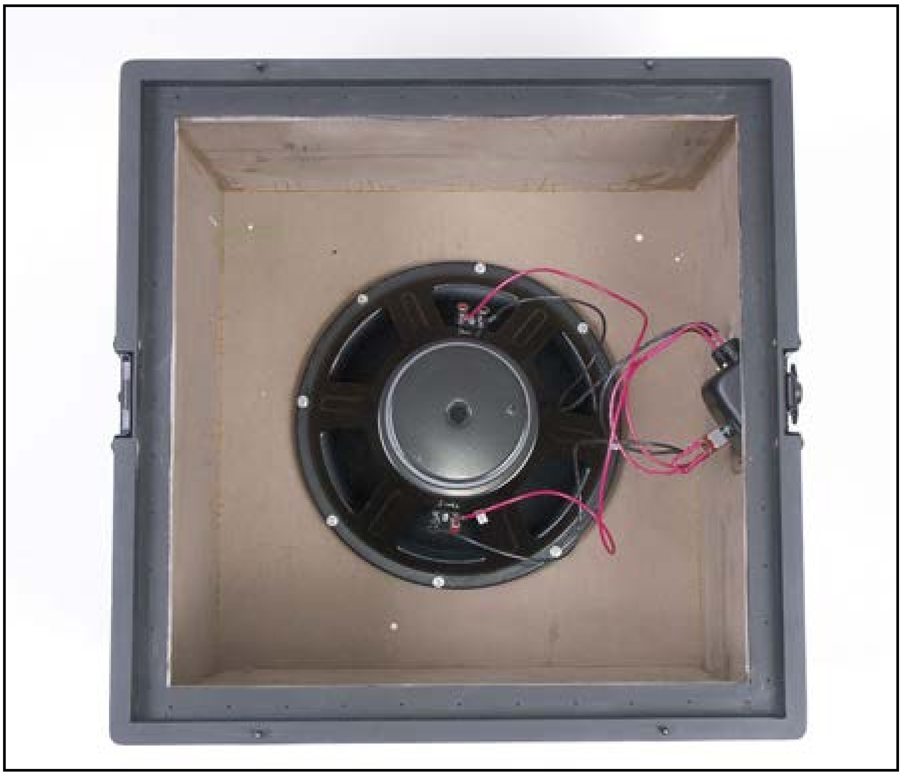

Photo 4: The DVC 15″ driver has been mounted and wired.

When the glue on all the panels was set, I routed the driver opening and all front and side corners with an edge bit. This step improves the looks and helps prevent damage to the sharp edges of the MDF panels, which can easily knick. Next, I sanded all the surfaces in preparation for paint. With MDF, it is important to use a solvent-based primer before painting with water-based paints. The primer not only provides a strong base for paint adhesion to MDF but also forms a barrier to the water in the paint. MDF is sensitive to water and can swell from the paint if not properly primed. I have found that Zinsser Bin works well for my projects.

I used flat black latex paint, which I allowed to thoroughly dry before installing the wired input dishes and the drivers.

The original subwoofers had latches placed on both sides to hold two of them together in a back-to-back configuration, if desired. I kept those latches even though I ultimately placed the subwoofers one on top of the other in a forward facing arrangement. **Photo 4** shows a mounted and wired driver.

Next, I fluffed up 5 lb of Acousta-stuff polyfill damping material and placed it into each box before mounting the back and screwing it shut. I attached rubber feet and put half-round foam weather stripping on the MDF back support strips to act as a seal. The subwoofers were then ready to go. **Photo 5** shows a side-by-side comparison of the cabinet sizes used for the 12″ and the 15″ drivers.

Listening Tests

I conducted all the listening tests with the subwoofers driven by a Crown Studio Reference amplifier. Crossover duties were provided by a Behringer DCX2496. The subwoofers were crossed over to the midbass arrays at 71 Hz with a 48 dB/octave Linkwitz-Riley slope. Room and speaker correction was from a Behringer DEQ2496. To compensate for the bass falloff of the subwoofers below 30 Hz, I applied a 1.5-dB boost at 25 Hz and a 2.5-dB boost at 20 Hz to all. I used source

Photo 5: The differing sizes of the finished 2-ft^3 and 5-ft^3 enclosures with the drivers mounted are shown for comparison.

material from .wav files stored on my laptop, which I played through an XMOS USB to SPDIF converter connected to the DEQ2496's AES/EBU input through a Canare Balun. The link between the DEQ2496 and the DCX2496 was through their AES/EBU connections to avoid multiple changes between analog and digital domains.

Group/ Performer	Album	Musical Piece	CD
Boston Audio Society	*Test CD 1*	Charles-Camille Saint-Saëns: "Organ Symphony"	CD-1
Boston Audio Society	*Test CD 1*	Giuseppe Verdi: "Requiem"	CD-1
Dallas Wind Symphony	*Fiesta*	Herbert Own Reed: "Prelude and Aztec Dance"	Reference Recordings RR-38CD
Jean Guillou	*Pictures at an Exhibition*	"Gnomus"	Dorian DOR-90117
Legacy Audio	*Music Sampler*	"Dynamic Drums"	Volume 1
Sergio Mendes	*Brasileiro*	"What is This"	Elektra 9 61315-2
Pink Floyd	*The Wall*	"Another Brick in the Wall Part 2"	Columbia C2K68519
Talking Heads	*Stop Making Sense*	"Slippery People"	Sire 9 25186-2
Clark Terry	*Live at the Village Gate*	"Hey Mr. Mumbles"	Chesky JD49
Turtle Creek Chorale	*Testament*	"We Fight Not for Glory"	Reference Recordings RR-49CD

Table 2: This is a list of the music I used in the tests.

Midbass duties were provided by my two dipole arrays consisting of six each 10" Peerless woofers driven by a QSC USA1310 amplifier. High frequencies came from my two Bohlender Graebener RD75 dipole mounted planar magnetic drivers, which were driven by a Crown Macro Reference amplifier. The mid to high crossover was provided by the DCX2496 at 303 Hz, with a 48 dB/octave Linkwitz-Riley slope. As you can imagine the mid- and high-frequency sections are capable of prodigious, very low distortion output, creating quite a challenge for the subwoofer section to match.

I chose 11 pieces of music that represented a mix of very low frequencies, dynamic impact, and complex bass lines to test the system. I ripped all the music from CDs in a .wav format and upsampled to 88 kHz/24 bit. **Table 2** lists the test tracks.

Ultimax Results

Charles-Camille Saint-Saëns' "Organ Symphony" piece has some strong notes at 18 Hz that are a major test for any subwoofer. The Ultimax did very well on most of the bass notes shaking the floor quite vigorously. On the lowest notes, the impact was not as strong or as clean as the reference system of the four DVCs. But, it was still quite amazing for two 12" drivers in such small boxes.

Giuseppe Verdi's "Requiem" is a killer piece not only because of the very strong bass drum but also the huge dynamic range. If you have to turn the volume down to prevent gross distortion on the bass, you will miss the subtle sounds of the solo vocalist at the end of the piece. The drum whacks were very forceful with no audible distortion.

Clifton William's *"Fiesta,"* performed by the Dallas Wind Symphony, has several selections with very strong bass drum notes mixed with complicated high frequencies. On the Prelude the drum was very powerful with a musical decay. This "Symphonic Dance No. 3" also had a good rendition of the bass drum.

Modest Mussorgsky's *"Gnomus,"* performed by Jean Guillou, is a piece with sustained pedal notes at various frequencies, levels, and durations. The lowest notes were quite strong but some of the growl was missing. The bass was a bit homogenized.

Legacy Audio's "Dynamic Drums" is one of the best pieces I have to test a system's dynamic range. There are some very low level cymbal hits and light skin taps followed by awesome kick bass whacks. This is a piece that will drive the Crown into clipping on this driver. There is no problem with the driver because if you reduce the level to just below clipping the sound is superbly clean. Volume displacement is not the issue at the higher

Driver	Area	X_{MAX}	Total Linear Excursion	Individual Linear Displacment	Number of Drivers	Total Displacement	Unit Cost	Cost per Liter	Total Cost
Dayton DVC385-88 DVC 15″	830 cm^2	15 mm	3 cm	2.5 ltr	4	10 ltr	$132.86	$53.36	$531.44
Dayton UM18-22 Ultimax 18″	1,213 cm^2	22 mm	4.4 cm	5.3 ltr	2	10.7 ltr	$266.65	$50.31	$533.30

Table 3: The Dayton DVC 15″ is compared to the Dayton Ultimax 18″.

frequencies that the kick bass produces, but rather available amplifier power given the low sensitivity. The clipping occurred at very high levels, probably higher than you would normally play but close to realistic. As with any dynamic music if you turn the volume down, you miss a lot of the low-level detail.

Sergio Mendes' "What is This?" has a very interesting bass drum that is strong and is difficult to cleanly reproduce. The Ultimax performed well with sound that was both strong and clean.

Pink Floyd's "Another Brick in the Wall" has probably had as much air time as any rock piece ever. The kick drum in this piece is interesting because it is not too deep but still round sounding. Any lack of volume displacement or amplifier power will actually make it sound sharper due to the introduction of high-frequency distortion products. With the Ultimax, it sounded great.

If you can sit still while Talking Heads' "Slippery People" is playing at high volumes, someone better call a mortician because you have probably died. This is my all-time favorite to get me totally hopping. When played on a clean system, the transients just go directly in with no stops along the way. The Ultimax was totally capable of doing justice to this piece. My notes said that "It really rocks!"

Terry Clark's "Hey Mr. Mumbles" has a combination of string bass and kick bass notes that require flat clean deep bass that is very transient to sound good. The string bass was natural, not bloated and the drums were quick and tuneful.

Turtle Creek Chorale's "We Fight Not for Glory" is another complicated dynamic piece that requires high performance from all parts of the spectrum. It starts relatively quietly but toward the end, there are some very strong bass drum whacks. If you have to turn the volume down because the drum is overloading the system, you will miss a lot of detail in the quieter passages. The Ultimax was both strong and clean giving a realistic presentation.

Titanic Results

The Saint-Saëns piece had slightly less very deep bass than the Ultimax, which is understandable given the frequency response curves of the two. However, it still shook the floor. There was no doubt there was a substantial amount of bass being produced.

The Verdi music still had a strong and tight bass drum even at the very high playback levels.

Herbert Owen Reed's "Prelude" was quite good and had a slightly better sound to the decay of the drum than the Ultimax.

"Symphonic Dance" did not have quite as much low bass as the Ultimax but was musical with lots of harmonic detail. "Gnomus" had good low pedal notes but with a bit more growl than floor shake.

"Dynamic Drums" had a great sound to the drum kit. The kick drum was very forceful. I made a note to myself describing it as a "sock in the gut." The amplifier was also a little more at ease with the transients. "What is This?" sounded clean and musical, although the lowest notes were not as strong as the Ultimax.

"Another Brick in the Wall" had a great rendition of the kick bass. It was quite tight without being sharp. While "Slippery People" was tight, a little of the rock factor was missing compared to the Ultimax. Just for grins, I bumped the EQ at 20 Hz to 4.5 dB and the rock factor increased. Because of the higher sensitivity, the amplifier did not complain.

"Hey Mr. Mumbles" also sounded good with much the same rendition as the Ultimax. "We Fight Not for Glory" also had a clean bass drum but not quite as deep as the Ultimax.

DVC Results

To make a long story short, the four DVCs were superior to the two 12" in every test. Gee, what a surprise that more than 2.5 times the linear displacement makes a difference! If I were going to summarize, I would say the DVCs were more relaxed and realistic with every test piece. Here are just a few specifics.

The pedal notes in the Saint-Saëns were superbly clean and although the floor was shaking more than with the others, those notes were not as obtrusive and never interfered with the rest of the instruments.

The "Gnomus" had both the very deep bass and the growl. A friend that builds high-quality vacuum tube amps stopped by when I was playing this piece and he commented, "This is the way it is supposed to sound!" It really sounded like an organ. All the other pieces had the same quality—taking you closer to the performance.

The Bottom Line

There is no getting around the fact that like everything else in life, building subwoofers is a matter of making the correct choices. There are always compromises. And, there is no substitute for lots of linear volume displacement. Modern drivers have come a long way in reducing distortions while working in their linear range. However, once X_{MAX} is exceeded, all those improvements are for naught. Looking at the distortion graphs of all three drivers, it is apparent that distortion may vary slightly in the linear operating range between the drivers, but once X_{MAX} is exceeded distortion rises at a very high rate.

Probably the most amazing result was the high degree of performance that was available from both the 12" drivers in the small boxes. I would venture

Building Supplies

Part	Source	Part Number
Pittsburg corner clamp	Harbor Freight, www.harborfreight.com	38661
Speakon cable connector	Parts Express, www.parts-express.com	092-058
Speakon chassis connector	Parts Express, www.parts-express.com	092-059
Input metal dishes	Parts Express, www.parts-express.com	262-838
Jasper Jig	Parts Express, www.parts-express.com	365-250
Zinsser Bin	Rustoleum, www.rustoleum.com	shellac-base primer
Acousta-stuf	Parts Express, www.parts-express.com	260-330

Speaker System

Part	Source	Part Number
Behringer DCX2496	Parts Express, www.parts-express.com	248-669
Behringer DEQ2496	Parts Express, www.parts-express.com	248-661
Ultimax 18	Parts Express, www.parts-express.com	295-518

Sources

DCX2496 Loudspeaker management system and crossover and the DEQ2496 mastering processor
Behringer | www.behringer.com

RD75 Dipole mounted planar magnetic drivers
Bohlender Graebener | www.bg-speaker.de

to say that unless you are as crazy as I am, two of these drivers would probably suffice for the vast majority of any listening situations. Here is how I size up the drivers.

The Ultimax has more low bass in the small box than the Titanic. However, the sensitivity is the lowest of all the drivers. You need a lot of power if it is to achieve its full potential. If you are willing to live with that requirement, it is probably the choice for the lowest bass.

The Titanic requires less power but in that less than optimum box, the rolloff starts at a higher frequency. You can add boost below resonance, but you then need more power and negate some of the sensitivity advantage. If you don't need the very lowest bass or else can live with a bigger box, this may be the driver for you.

The DVC from a price, displacement, and sensitivity standpoint is better than the 12″ drivers, but you pay the price in box size. If you can live with the size, it is the winner as far as I am concerned.

Now that we have that out of the way I will throw a monkey wrench into the mix. There is a new driver that might be the answer to the over the top approach, if you are starting from scratch. There is now an 18″ version of the Ultimax with 22-mm X_{MAX} with a sensitivity rating of almost 86 dB/W. Two of these would have the same linear volume displacement as four of the DVCs but require only two 5-ft^3 boxes. It would be a good compromise between displacement, size, price, and sensitivity.

Table 3 shows a comparison of the DVC15 and the Ultimax 18. The DVC still has the sensitivity edge by 6 dB, but the smaller box size would be very tempting. It's great to have choices. If I get time next year, I may try that approach. In the meantime, this project met all my targets and sounds great. Happy building!

About the Author

Thomas Perazella is a retired IT director. He is a member of the Audio Engineering Society, the Boston Audio Society, and the DC Audio DIY group. He has authored several articles in professional audio journals.

Ribbon Planar Magnetic Designs

As we discussed last month, the vast majority of headphones use an optimized dynamic microspeaker consisting of a voice coil, a neodymium magnetic structure, and a polyester film diaphragm. As is the case with speakers and microphones, there are alternatives to these dynamic transducers. This month we will focus on orthodynamic/ribbon planar techniques.

Photo 1: An early Technics brand "leaf" tweeter got its name because of the conductor pattern on the substrate.

By
Mike Klasco and Steve Tatarunis
(United States)

During the past 40 years, several ribbon planar headphones have reached the market. In this article, we explore why the ribbon planar headphone continues to deserve its place in the sun and are some of the most desired audio products.

Pure Ribbons

The original implementation of the ribbon driver consisted of a thin metal foil ribbon suspended in a magnetic field. The upper and the lower points from which the metal foil is suspended also serve as the plus (+) and minus (–) terminals for the incoming signal. The metal foil design worked well for microphones and tweeters, but it was not quite right for the requirements of a headphone driver.

Orthodynamic

Orthodynamic, isodynamic, magnetostatic, or the more common term ribbon planars, use a derivative approach in which a flex circuit board consisting of a thin polymer film with a metal conductor pattern (this composite sandwich is called the membrane) functions as both the diaphragm and the (flat) voice coil. This membrane is suspended on a frame and is usually lightly tensioned. The magnetic system may be one sided and located behind the membrane or double sided with the membrane between oppositely aligned magnets.

Ribbon transducer manufacturing is a specialized craft, even when part of larger speaker manufacturing operations such as Foster (Fostex) and Matsushita. The "leaf tweeter" flex circuit diaphragm was introduced by Mashitusita's Panasonic Acoustic Research Lab in Osaka, Japan in the mid-1970s and was introduced as its Technics brand "leaf" tweeter (owing to the conductor pattern on the substrate). The current flowing through the coil interacts with the magnetic field of the magnets positioned on either side of the diaphragm (see **Photo 1**).

The driving force covers a large percentage of the membrane surface and reduces resonance problems inherent in coil-driven flat diaphragms. A Japanese competitor, Fostex (the premium group at Foster), soon followed with its own leaf tweeter models and called its products uniform phase ("RP") due to the even driving force of the distributed voice coil conductor pattern. Following these flagship products, Panasonic affiliate JVC and Foster supplied mid-priced ribbon leaf tweeters and midranges on an OEM/ODM basis. Offshore ODM vendors for these round ribbon tweeters included Sawafuji in Japan, Meiloon, and quite a few others.

These round ribbon planars found good success and earned high praise for their use in exceptional headphones from Yamaha, Fostex, and a dozen other brands that sourced similar ribbon planar components for their headphones. Fostex started making orthodynamic headphones in the mid 1970s with its classic T10, T20, T30, and T50 and continues to manufacture this class of headphones today.

Yamaha's Orthodynamic line of headphones appeared in the US market in the fall of 1976, consisting of the HP-1, the HP-2, and the HP-3 (see **Photo 2**). The Yamaha headphones were notable not only for their good sound but also for their comfort and elegant industrial design created by famed Italian designer Mario Bellini. Eventually, Yamaha faded from the headphone market during the headphone "Dark Ages" around 1989.

Photo 2: The Yamaha HP-1 was considered revolutionary for its day due to its Mario Bellini headset design. (Photo courtesy of Yamaha Corp.)

Tensioning Issues

While tightly tensioned ribbon tweeters might respond down to 7 kHz or even 3 kHz, to achieve low-end in a headphone driver, some excursion and a lower diaphragm fundamental resonance is needed. This is where the engineering gets complicated as even slightly widening the spacing so the diaphragm does not hit the magnets also reduces the magnetic field strength in this gap. Another issue is that loosely or non-tensioned diaphragms are likely to buzz. Getting these factors "just right" and applying the proper "secret sauce" made ribbon headphone design more of an art form than an engineering discipline.

In the last decade, material science has been kind to ribbon planars, both for the diaphragms and the magnetic systems. The polymer/conductor composite diaphragm uses the same construction as the flexible circuit boards commonly used in laptops, cell phones, and tablets. But to put it into perspective, the substrate film is a far thinner gauge. (A flex circuit board is likely to be 2 mil (2/1000"), while a ribbon headphone driver membrane might be a fraction of 1 mil.) The impedance of this conductor pattern/diaphragm can be readily designed by conductor length, width, and thickness to be in a range that the amplifier can drive—not just a reasonable impedance but also essentially a pure resistive load.

From our perspective, a diaphragm with aluminum conductors is preferred. Aluminum conductors are lighter than copper, but there are very few specialist vendors for this. Even harder to source are the very thin gauges and double-sided conductor (where the aluminum conductor pattern is on both sides of the film) that can provide enhanced performance and sensitivity.

For most designs, the film is tensioned, although non-tensioned "self-supporting" techniques have been successfully commercialized for some of the smaller headphone drivers. The film diaphragm materials have dramatically improved over the "coke bottle" polyethylene terephthalate (PET, also known as Mylar), although PET polyester is still common. For higher power applications outside of headphone drivers (e.g., autosound and pro-sound), there is polyimide, such as Kapton. Polyimides with a copper conductor have been used for headphone drivers, perhaps because of the ready availability of flexible circuit boards vendors. Teonex or polyethylene naphthalate (PEN)

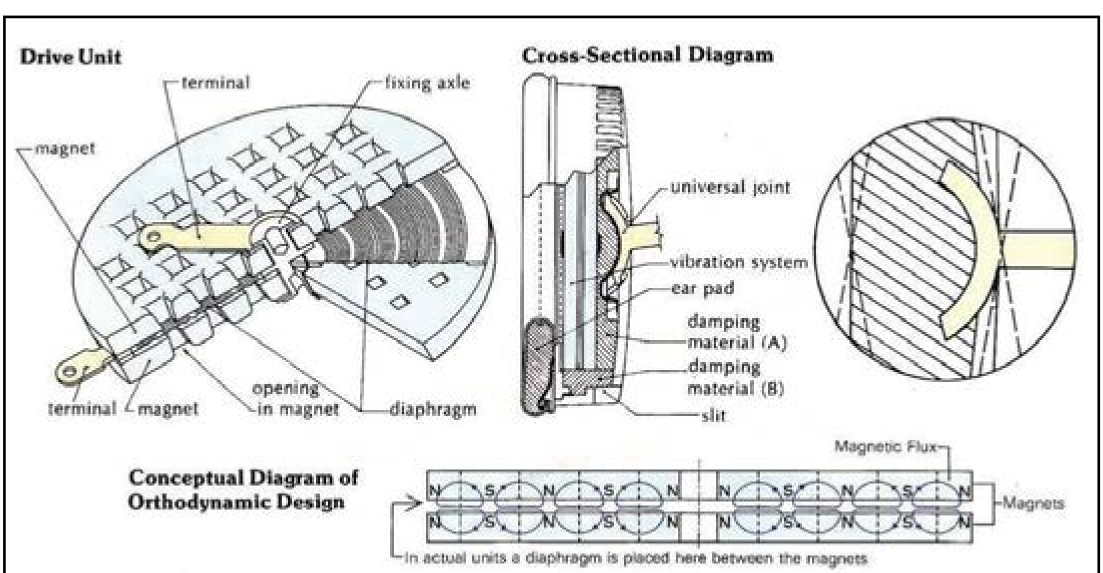

Figure 1: This diagram details the Yamaha HP-1 ribbon planar headphone driver. (Image courtesy of Yamaha Corp.)

Photo 3: The OPPO PM-1 uses a unique seven-layer diaphragm, double-sided spiraling coils, and an FEM-optimized magnet system. (Photo courtesy of OPPO Digital)

polyester is another option for headphone drivers and offers several performance benefits. However, the most exciting development is the laminated damped high-performance polyether ether ketone (PEEK) films such as those from AEE Tape. As you can imagine, every aspect of the composite diaphragm contributes to its overall sound quality, from the aluminum or copper conductor to the adhesive used to laminate to the film to the film substrate and its thickness.

The Magnets

The neodymium magnet has significantly improved planar magnetic performance over earlier ferrite and samarium cobalt designs, enabling modern ribbon planars to reach higher sensitivity. This stronger magnetic field also enables better electrodynamic control of the diaphragm. Older designs using low-energy density magnets were relatively thick. With their double-sided configurations, the sound radiating from the face of the membrane had to pass through the front magnets, which created a notch in the top-end response. As the neodymium magnets can be quite thin, this acoustic resonance now tends to be above the audio band enabling the exceptional unblemished response curves seen on today's ribbon planar drivers. The large double (push-pull) low-energy magnet configurations would often be considerably heavier than their dynamic cousins. But today, examples such as the OPPO PM series are actually more sensitive and lighter than their conventional microspeaker headphone driver counterparts.

The advantage of a ribbon planar driver is that the diaphragm has very little mass so it can quickly accelerate, yielding extended high-frequency response (see **Figure 1**). Whereas with a microspeaker headphone driver, the voice coil generates the sound energy, which then passes through the glue joint to drive the diaphragm. This configuration results in propagation delays, resonances, and other spurious acoustical by-products. In comparison, ribbon planar polymer-film drivers are driven in phase throughout the diaphragm surface as the voice coil conductor pattern is evenly distributed over the substrate.

Ribbon Renaissance

At the top-end of the market are headphones such as the Beyer T-1, the Sennheiser HD-800, and the AKG K712, which all have unique design aspects but still retain dynamic (voice coil with diaphragm) configurations. However, several new brands are now manufacturing headphones with new high-performance ribbon technology.

Audeze is a startup founded in California in 2008. The company can claim some of the credit for inspiring the renaissance of new generation of ribbon planar audiophile headphones. Audeze's LCD open-back series has been well received by the discerning international audiophile community and recently its new EL-8 closed back model has come to market.

Soon after Audeze introduced its LCD headphones, HifiMAN, a Chinese-American outfit launched its HM-801 in 2010 and in 2013, the HM-901. The HE-1000 is the current flagship of the product line and requires a serious headphone power amplifier to drive (like most of the Hifiman headphones). Hifiman has also recently introduced its least expensive ($399) ribbon planar with the HE-400 and this model boasts high sensitivity and can be driven by a smartphone.

OPPO Digital, known for its outstanding Blu-ray players, entered the headphone market in 2014 with the PM-1 open-back ribbons, bringing high comfort, high sensitivity, and impeccable ribbon sound quality. However, all those qualities also come at a high price.

The OPPO PM-1 double-sided conductor and other advanced technologies resulted in significantly higher sensitivity than most other ribbon planar headphones and thus, can easily be driven by smartphones. Rounding out the OPPO product line is the less expensive but still highly respectable ribbon planar models like the PM-2 and the just released PM-3 closed-back design (see **Photo 3**).

Fostex, one of the developers of the ribbon planar technology has launched its TH500RP open-back magnetic planar stereo headphone. The diaphragm is polyimide with a copper conductor pattern made of high-grade construction.

Of course, with many of these products receiving great reviews and commercial success, there are others jumping into the fray. Next month, we will cover some of the other remaining headphone driver types including electrostatic and piezoelectric technologies.


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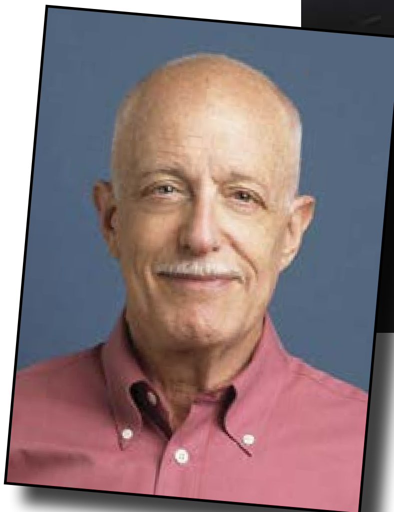


Audio industry veteran Richard Schram is Parasound's president and CEO.

An Interview with Richard Schram, Founder of Parasound

By
Shannon Becker
(United States)

SHANNON BECKER: Tell us a little about your background.

RICHARD SCHRAM: I grew up listening to family records and music on FM stations WFMT, WEFM, and WNIB in Chicago, IL. I took piano lessons very early and my parents later enrolled me in the Northwestern University Music Department's preparatory piano program when I was eight.

I came to the Bay Area in 1964 and graduated from the University of California at Berkeley in 1968. This was an extraordinary time and place with the convergence of music, FM multiplex broadcasting, advancements in audio technology, and the role of music in social change and vice versa (not necessarily in that order).

During school I had a part-time stock boy job at the original Pacific Electronics store near campus where I installed phono cartridges and assembled the Quadraflex house brand speakers. I moved on to sales, store manager, and a management position in the main office. The company changed its name to Pacific Stereo and had grown into a regional retail chain when CBS purchased it in 1973.

I was tasked with developing and sourcing the company's house brand products. I traveled to Japan dozens of times in the 1970s and with letters of credit and the prestige of the CBS-Sony joint venture, I was able to cultivate relationships at levels in Japanese companies that would have been unattainable for someone in his 20s. I also worked with engineers at the CBS Technology Center in Stamford, CT.

Only a handful of audio manufacturers developed products while working for a retailer and none of those was anywhere near as large as Pacific Stereo. I had the unique benefit of our size, CBS's backing, and direct, unfiltered feedback from a very demanding sales force. I learned to create products that were salable, reliable, and most of all, relevant.

One of those house brands, Concept, has a loyal following on Audiokarma and when I look at the robust designs of my products from the mid–late 1970s, it's clear where Parasound's roots are. Take for example, the 1978 Concept 16.5 Receiver (www.chrisinmotion.com/MyFavoriteReceivers.htm#16.5). I conceptualized and developed the details of the 1978 Concept 16.5 Receiver's product appearance, its performance objectives/specs, and its feature set. The circuits were designed by a Japanese team led by engineer Ken Hashida.

SHANNON: How did you become interested in audio electronics?

RICHARD: Like many audio manufacturers, I started my journey as a music lover/performer and audio enthusiast. One hot and humid summer day, I wandered into an Allied Radio store three blocks away from the Northwestern music building. The store was air conditioned and in the back room they were playing Stan Kenton's "Intermission Riff" (1957) in stereo on a reel-reel tape deck. I was enthralled.

In 1963, I attended one of the first live vs. recorded demonstrations with Chicago's Fine Arts Quartet and, if I recall correctly, an Ampex studio recorder, Dyna MKII amps, and AR3 speakers. At that time, it was magical and I was hooked on audio electronics to bring lifelike music into the home.

SHANNON: Can you tell us about your work? How and why did you start your company, Parasound (www.parasound.com)?

RICHARD: I left Pacific Stereo in December 1980. I had some wonderful consulting gigs with JBL, Yamaha, the original Jensen, and I helped my late friend Chuck Kittelson launch Proton in the US. But I wanted to make my own products and Parasound's first model came to market in September 1981.

SHANNON: Why did you decide to have Parasound products built in Taiwan?

RICHARD: Most of my products in the 1970s were sourced in Japan and the earliest Parasound models were also built in Japan. But small audio manufacturers couldn't compete as the yen soared and nearly all of them went out of business by the early 1980s.

I chose Taiwan because there were some quality factories owned by audio enthusiasts who welcomed Parasound as a small customer. I've continuously worked with one contract manufacturer (CM) for 33 years, another for 26 years, and the third for seven years (I've known the general manager since she worked for the first CM 25 years ago). I've challenged them with increasingly ambitious projects over the years and these Taiwanese will persevere until they overcome the most difficult obstacles. In contrast, most Chinese CMs will walk away from a small audio customer's project when difficulties arise and become inconvenient for them.

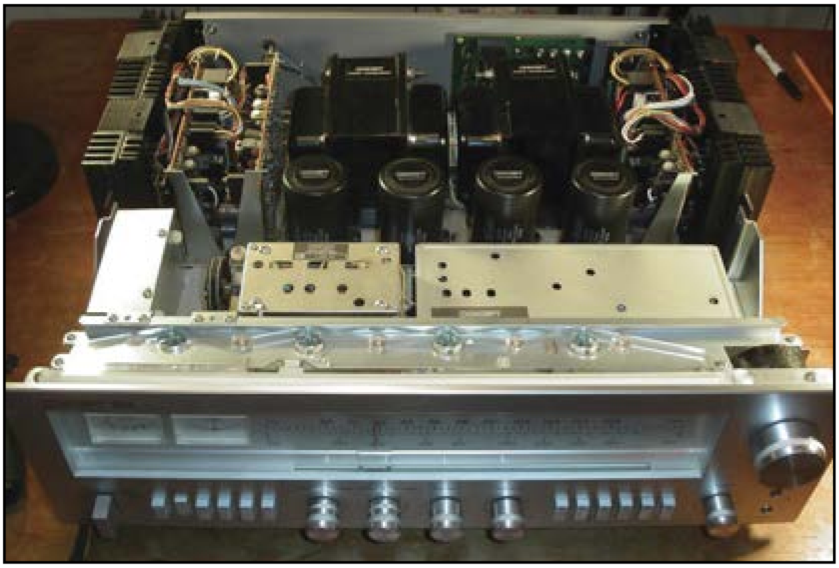
SHANNON: How has your strategy changed since you opened in 1981? What has stayed the same?

RICHARD: My original plan was to produce and sell audio products around a core of the largest US audio chains: Pacific Stereo, my ex-employer, Tech HiFi, and CMC. In this plan, the retail chains purchased with letters of credit, I'd work three days a week and make a million dollars a year. What a plan! As these companies failed in the 1980s, I stopped targeting large customers and moved Parasound upscale to appeal to the finer independent audio retailers. I engaged John Curl in 1988 to assist us in creating flagship products.

Our products' reliability caught the attention of audio companies who became our OEM (original equipment manufacturer) customers. With my OEM background developing OEM business was second nature.

Our OEM business included: customized CD changers, which AEI Music (later DMX) installed in the listening kiosks at every Borders Book Store; Sonance's power amplifiers for the first five years; sub amplifiers and a line of outdoor speaker models for Atlas/Soundolier and Memorex; 500,000 CD laser lens cleaners to Memorex; and CD loaders, servo boards, and FL displays to UltraAnalog, which built complete players for Krell, Classe, and later for Parasound. We produced a fetal heart rate monitor for a local medical instrumentation company; built Magnavox speakers for Philips; and FM-AM tuners for Aragon, Magnum Dynalab, and Classe. We also built a subwoofer-satellite system for Mitsubishi North America that contained the first commercial subwoofer driver to use Ferrofluidics 027 thanks to Mike Klasco.

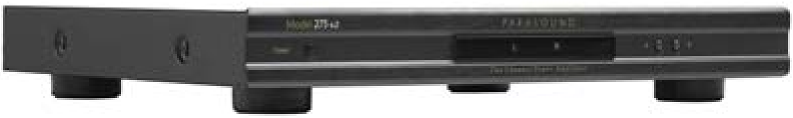
SHANNON: How has John Curl contributed to the success of Parasound and how did you two meet?



Richard Schram conceptualized and developed the details of the 1978 Concept 16.5 Receiver's product appearance, its performance objectives/specs, and its feature set. The circuits were designed by a Japanese team led by engineer Ken Hashida. The Concept 16.5 has the distinction of being the most powerful receiver ever fielded by a so-called "house brand," in this case, Pacific Stereo, a subsidiary of CBS.

Parasound CEO/President Richard Schram is pictured with John Curl, a legendary amplifier designer. The men have worked together for several years, perfecting amplifier design.

Questions & Answers



Parasound's Model 275 stereo home theater power amplifier is the most compact member of its NewClassic line of high-end home-theater products.

RICHARD: I first met John on an airport bus in Tokyo. He was visiting the manufacturer of a Harman Citation amplifier he had worked on with Matti Otala. I was re-introduced to John by one of my employees who was friendly with him. John offered to help us because he felt that some of the products we were already building were worth his attention.

SHANNON: Tell us a little about the company's fabled Halo JC 1 mono power amplifier.

RICHARD: The JC 1 project was therapy for me. In 2000-2001, I was roasting in DSP hell with the Finnish design house working on our C 1 and C 2 7.1 channel surround processors. I needed something to lift my spirits and the idea of making a really great stereo amplifier—or better, an awesome mono amp—would pull me out of my funk. John Curl assembled his team—Carl Thompson, his long-time partner who designed circuit boards for all of his products (including the Vendetta phono preamplifier), and the late Bob Crump, who specified many of the key parts. Bob was a legend because he really could hear the differences in sound between materials and the brands of audio parts. The JC 1 came out in March 2003 and its popularity hasn't diminished.

Parasound's Halo JC 2 is a remote-controlled, solid-state preamplifier with switchable absolute polarity, a rear-panel IR repeater input and loop out jacks, four 12-V trigger output jacks, and a two-way RS-232 serial port.

Parasound's Zdac v.2 is designed to sound natural and less "digital" than more expensive DACs thanks to its asynchronous sample rate converter. A precise re-clocking system eliminates bitstream-contaminating jitter that's picked up along the digital signal chain.

SHANNON: Do you have a favorite product?

RICHARD: I imagine it's easy for most audio manufacturers to identify their favorite products. This is all but impossible for me because of how intensely I'm involved in the conception and development of every product.

SHANNON: What is it about your amplifier designs that give them such longevity in the audio market?

RICHARD: We take the old carpenter's adage "Measure twice, cut once" to an extreme. In my opinion, the audio manufacturers who come out with "new and improved" models every year didn't know or care enough to get them right in the first place or are looking for additional revenues with questionable value propositions. Rather than using valuable resources to churn models and annoy our customers, we aim for lasting value and respecting the investments our customers make in the brand.

SHANNON: What do your attribute your company's continuing success?

RICHARD: Since I've been doing this for so long I expect I've had the opportunity to make plenty of mistakes. As I matured I realized I could learn something—and I have. We're very good at what we do these days. History is littered with audio companies with big ambitions that failed because their owners put making money ahead of product value, product quality, and their business practices. I've learned that if I value our customers and do my job well the money will follow. We respect our customers because most of them don't have trust funds and have to work for their money. So we give them their money's worth and then some. Plus, we provide technical support that's second to none. Do you know any audio company that answers every phone call and every email request for pre-sales or after-sales support?

SHANNON: Can you detail anything you're using for our *audioXpress* readers?

RICHARD: I would like *audioXpress* readers to know that Parasound will continue to use traditional audio technologies as long as they're superior for the reproduction of music. Behind the scenes, we are researching ways that new audio technologies can meet and surpass our standards. I'm confident that future products will keep Parasound ranked as the audio industry leader in delivering outstanding performance, reliability, value and customer support.

Updating the Bogen MO100A Amplifier

The Bogen MO100A was originally designed for constant-voltage audio distribution, rated at 100-W. Although it is configurable to various installations, its output tubes are difficult to find. Learn how to modify this amplifier, using more common tubes.

By
Bruce Brown
(United States)

The Bogen MO100A was a sound reinforcement amplifier originally designed for theaters and stadiums (see **Photo 1**). It used four 8417 output tubes driven by a 7247 dual triode at 100 W RMS. (The 7247 has one high mu and one medium mu triode).

Bogen made a later version of the amplifier that used two 6550s for the rated 100 W. The output transformer had taps for 16 Ω, 25 V (6.25 Ω), 25 V CT (for line transformer networks), 70-V line, and 115-V lines. The rated frequency response is 8 Hz to 50 kHz ±2 dB. Bogen also made a version of this amplifier that had a 200-W output as well as a 50-W unit. Both the power and output transformers are quite substantial, with the amplifier weighing 33 lb.

Because the MO100A uses 8417 output tubes, which are very hard to find, the amplifier can usually be bought at a bargain price. I purchased several of these, without the output tubes, and decided to see if they could be rebuilt to use a more common output tube (see **Photo 2**).

The power supply in the MO used a voltage doubler and sent more than 650 V to the plates of the 8417, which eliminates the use of many of the more common output tubes. This is about 100 V more than my manuals say the 8417 was rated. However, Bogen got away with this by running the screens at a conservative 340 V.

New Bias Supply

I decided to rebuild one of the amplifiers to use either EL34s or 6550s. The original bias circuit provided −18.5 V and needed a range of around −25 V for the EL34s to −55 V for 6550s. The bias voltage is derived from the AC high-voltage tap of the power transformer through a 0.5-µf 630-V mylar cap (see **Figure 1**). I redesigned the bias system with an adjustable pot. I used some of the original resistors from the Bogen bias system and added a 50-kΩ pot and a couple of 4.7-kΩ, 2-W resistors in a voltage divider. This setup yielded −15 to −55 V, so I had plenty of range of bias voltage (see **Figure 2**).

New Power Supply

The original power supply in these amplifiers used two separate can 100-µf 400-V capacitors in the doubler and a third can with a 100-µf 400-V section for the screen grid power supply and a 10 µf 400-V section for the phase inverter/preamp section. I used some new JJ Electronic can capacitors to replace the 100-µf sections and a new metal can (CET makes several new metal can capacitors from original Mallory tooling). These are available from tube supply companies. I get mine from Antique Electronic Supply (www.tubesandmore.com) in Tempe, AZ.

I chose to use a double 100-µf can and a double 50-µf (in parallel sections used to replace the capacitor that is connected between the high-voltage leads of the transform—not grounded) to replace the three 100-µf sections. A 20/20 µf can be used to replace the 10-µf section.

You may have to do a little metal working and drilling to get the new caps mounted, depending on the type you use. I also mounted a fuse clip under the chassis and wired the center tap (CT) of the output transformer through the fuse. Use a 500-mA fuse to protect your output transformer in case of a catastrophic tube or capacitor failure.

Chassis Update

These amplifiers had multiple inputs, a switchable high and low Z (with a plug in matching transformer) and several other functions not needed for a hi-fi amplifier, so I eliminated them from the circuit. This frees up some space and eliminates potential noise from two slide switches and two sockets. There were 0.001-µf ceramic capacitors across the music/speech switch to limit bass response in the speech position, which I also eliminated.

If you need DC blocking from your preamplifier, you can directly hook the new input terminal to the leg of the volume control or install a 0.1-µf capacitor between the input jack and the volume control. If you wish to completely eliminate the volume control connect a 470-kΩ resistor between pin 7 of the 7247 and ground and the capacitor between pin 7 and the input connection.

I also found the multiple output taps and bridging connections were not needed. I removed the terminal strip from the rear panel and mounted an under-chassis strip to cleanly terminate the extra wires (see **Photo 3**). I only used the 6.5-Ω and the 16-Ω taps on the output transformers, so I mounted the three output jacks on the new panel. This does limit your speaker selection a bit, but you can use the 6.5-Ω tap with 4- or 8-Ω speakers. I mounted a terminal strip under the chassis and terminated the rest of the output taps to contain the wires.

I also removed the sockets and switches from the top section of the chassis. I moved the power switch and pilot light from the front panel to the rear. This design change means all the controls and the connections are now on the rear panel of the amplifier.

Then, I built a wood surround for three sides of the amplifier to give a little less utilitarian look (and to cover the excess holes). I mounted a couple aluminum

Photo 1: This is the stock Bogen MO100A amplifier as found.

plates to the top and rear of the chassis to cover additional holes and added new jacks and binding posts to facilitate connections. I also added a pot and terminal for setting the bias voltage from the top of the amplifier. If you aren't going to try different types of output tubes or just want a cleaner top, you can mount the pot underneath the chassis. I made the jeweling on the aluminum plates with a steel circle brush and a drill press.

Before you start anything up, you should check all the resistor values throughout the amplifier. I have often found that the plate resistors on the 7247 and the 8417s are off value. Replacing them is a good

Photo 2: The upgraded Bogen is shown with all its modifications.

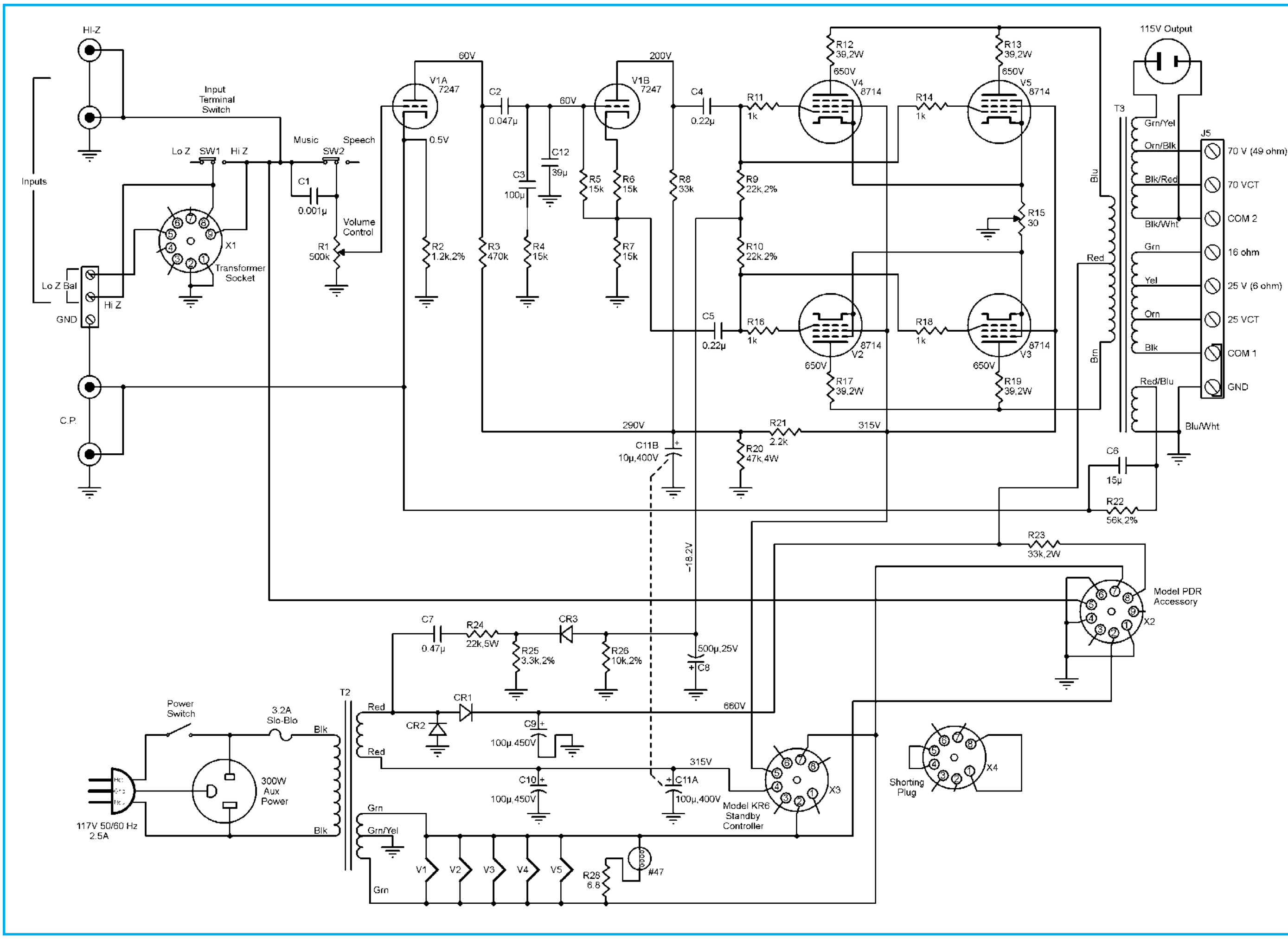


Figure 1:This schematic details the power supply of a stock Bogen MO100A amplifier.

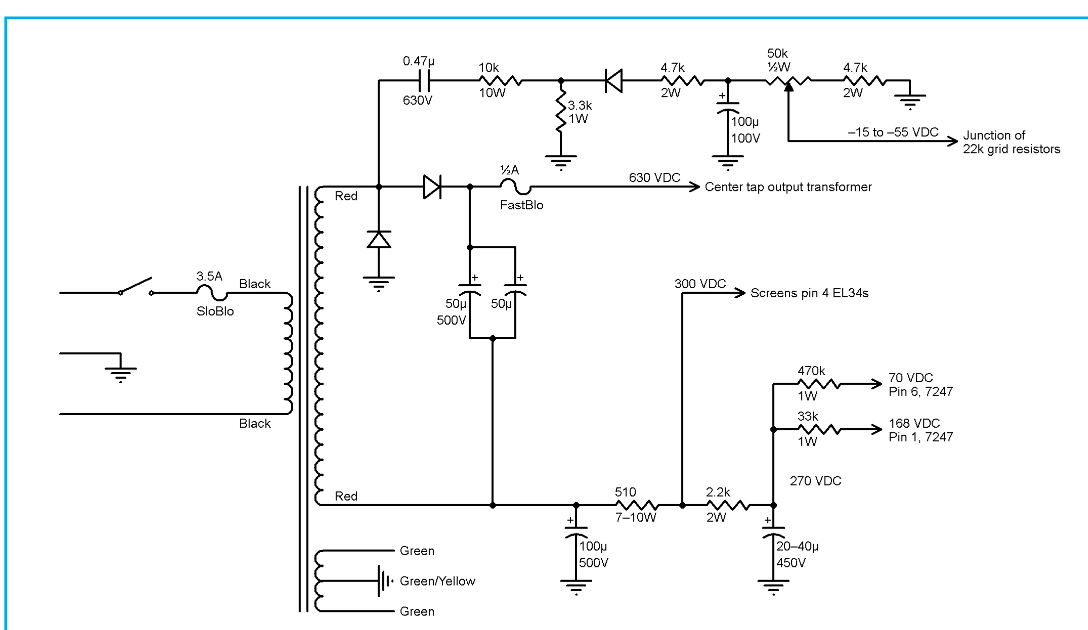


Figure 2: This new power supply schematic will help you rebuild this amplifier.

idea. The 470 kΩ on the plate of the first section was 190 kΩ when I checked it. This really messes up the gain for the first stage, which is where you need all the gain you can get. Also, carefully check R 9 and R10, the 22-kΩ bias feed balance resistors. They need to be 1 to 2% within each other. I used a matched set of 5% metal films, as the carbon comp ones in the original circuit were way off value. An additional bit of small rewiring is necessary.

The output tube sockets need to have pins 1 and 8 connected together. Three of the sockets have nothing connected to pin 1, but one uses pin 1 as a tie point for the 1-kΩ grid resistor. It is easy to just use a new 1-kΩ resistor to replace the one that is tied to pin 1 and connect the other end to the 22-kΩ resistor junction.

I could walk you through testing the coupling capacitor, but on an amplifier this old, I automatically replace all of them with the same value and voltage rating. I like the Russian paper in oil (PIO) capacitors coming out of the Eastern bloc right now. Most of these are 1950 to 1960 Russian military vintage stuff. They are well made, very smooth, and pretty inexpensive. But, you can use whatever capacitors you want, just make sure they have at least the same voltage rating as the originals, as this amplifier runs a lot of plate voltage.

Starting the Amplifier

When firing up the amplifier for the first time, first set the factory balance control to the centered position. Your new bias system should be verified before you install the output tubes. (It is okay to have the 7247 installed.) With no output tubes in the amplifier, use a variac transformer to bring up the line voltage to about 60 or 70 V. Continue to bring up the line voltage until you reach 100 AC. Now check the B+ voltages to make sure the rest of the power supply is fully functioning. Connect your voltmeter plus the lead to the bias jack (or the center wiper of the bias pot). Adjust the pot to give you approximately the highest negative reading.

Selection of output tubes is not critical, but it is important that they be matched. Since you only have on bias control for all four tubes, they need to be matched. You can do this on a mutual conductance tube tester or you can buy a quad of factory-matched tubes. (You could also add a pot for each output tube and be able to bias unmatched tubes.)

If everything is good, then power the amplifier down, let it set for about 15 min., and then plug in the new output tubes. Bring the amplifier back up on the variac transformer and carefully monitor the voltages. Pay special attention to the bias voltage. If any of the plates start to glow red, you need to immediately shut down the amplifier.

Photo 3: Much of the wiring had to be revamped.

I initially tried some 6550s, but the bases and the glass on the tubes touched. I was concerned the heat might be a problem, so I put in a quad of EL34s. With the bias set at about −25 V, the amplifier functioned and I got a signal. You might be able to use some 6550s, but I just went to the EL34s since they have smaller envelopes. The factory balance control can be adjusted with an oscilloscope for the most symmetrical sine wave and lowest distortion, as outlined in the factory manual. If you don't have an oscilloscope, you can center the wiper on the pot and it will function just fine. All the voltages were within 10% of the factory specifications (except bias voltage, which is set for EL34s)

I hooked up the amplifier to my oscilloscope, HP AC voltmeter, and Sencore audio analyzer and it produced 87 W RMS with a completely symmetrical sine wave and no clipping. This is about where I expected it to be considering the 7247's drive capability. You could adjust the 7247's plate and grid resistors to get a little more gain, but I was happy with the results. The amplifier refused to clip with 2 V of input drive.

The Result

The finished amplifier sounds very good. It has tons of power and seems to be very reliable, although I only have about 30 hours on it at this point. If you have any questions or comments, you can contact me at tuninfork@aol.com.

Resource

Bogen Model MO100A Operating Manual, www.bogen.com/support/discontinued/pdfs/MO100Am.pdf.

About the Author

Bruce Brown has been a hobbyist in electronics for more than 45 years. He currently restores vintage tube equipment and builds custom hi-fi and guitar amplifiers. He has two websites: www.vintagetubeaudio.com and Junkwerks.us. He can be reached at Tuninfork@aol.com.

GLASS AUDIO

Use an Antique Radio to Listen to Your Vintage Playlist

Playing old music through antique radio equipment is an experience. Booming vocalists, harmonious chorus, and best of all no higher frequency "pops" and "hisses" that would be reproduced by hi-fi equipment thanks to the limitations of an old radio's audio signal chain. In this article, I will describe how to choose an antique radio, outline a step-by-step restoration process, and show how to make simple modifications so it can be plugged into any modern audio source.

Photo 1: The Silvertone Model 4686 is a fine example of the iconic American console radio.

By
Gregory L. Charvat
(United States)

Photo 2: The Silvertone 4686 contains dial lights and a magic eye.

To tune in the past and play vintage music through an antique radio, first you need to find the right radio. Antique radios can be found at garage sales, estate sales, hamfests, antique radio swap meets, and online. Most antique radios are relatively inexpensive and likely do not work when you buy them. Sometimes, you may get lucky and find an external phonograph audio input where the input is not RIAA compensated (early phono inputs were not RIAA compensated), in which case you would simply plug your audio source into this. Be sure to find a radio that is clean and uses a power supply transformer or runs on batteries. Do not use hot chassis radios to avoid risk of feeding line voltage back down the shield of your audio input.

Restoring Antique Radios

Much has been written on how to restore antique radio equipment, the reader is encouraged to research further (see Resources). Generally speaking, follow these basic guidelines to restore your radio:

- Do not power on your radio. You could damage it if you power it on before replacing the capacitors.
- Find a service manual, most are available online.
- Replace all electrolytic and paper capacitors with new ones of similar value and same or better voltage rating.
- Replace anything that looks damaged.
- Try the radio. Most antique radios will work once the capacitors have been replaced.
- If it does not work then signal trace through circuit, replacing resistors where needed. Old resistors have a tendency to increase in value with age. Often a bad stage in a tube radio is due to a resistor that has increased in value to the point of biasing the tube into cutoff. Rarely do the tubes need to be replaced. Sometimes the winding on an intermediate frequency (IF) or audio transformer is open. In this case, replace with an identical part from another radio or re-wind the transformer.
- The radio should work.

Confusing and difficult problems do arise during restoration. Fortunately there is an online community called Antique Radios that is devoted to the restoration of antique radios (see Resources).

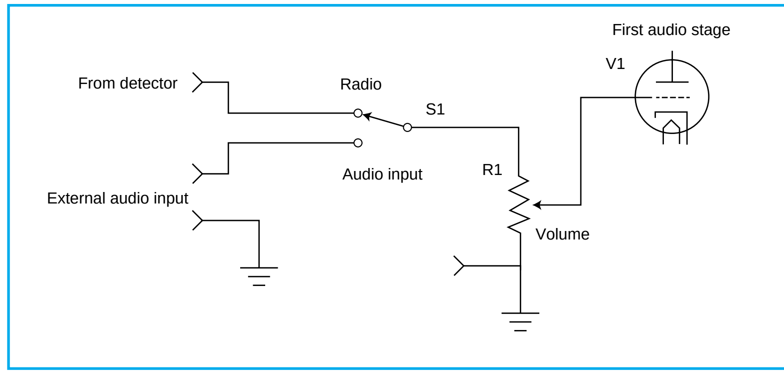


Figure 1: This is a schematic of the Silvertone 4686's audio input circuit.

Resources

Antique Radios: The Collectors Resource, www.antiqueradios.com.

G. Charvat, "Welcome to the Old School: Restoring Antique Radios," Hackaday, http://hackaday.com/2014/09/09/welcome-to-the-old-school-restoring-antique-radios.

——, "Restoration of a Silvertone Model 4686 Console Radio, ca 1937," YouTube, http://tinyurl.com/oxthx7p.

——, "Sentinel Radio Repair and Modification," YouTube, http://tinyurl.com/ky42oro.

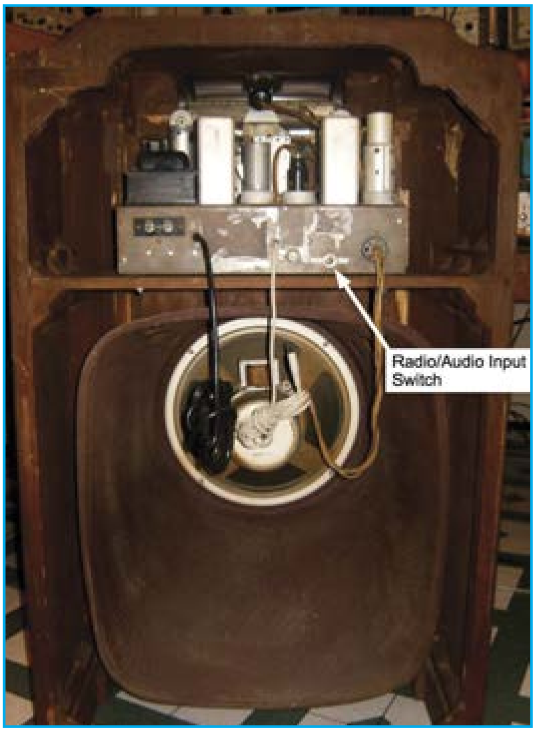


Photo 3: I drilled a radio/ phono toggle switch into the Silvertone 4686's rear panel.

The Iconic American Console Radio

If your radio does not provide an external audio input, then you can make one by locating where the detector ties into the first audio stage. I have provided an example of this modification for a Silvertone Model 4686, which is a fine example of the iconic American console radio (see **Photo 1** and **Photo 2**).

Splice in a toggle switch (S1) between the radio's detector output and the volume control (R1) before the first audio input stage (V1) to select between radio and external input (see **Figure 1**). Use your favorite audio input connector for the input connection. For this project, I used a 3.5-mm stereo plug. Place the radio/audio input switch in a convenient location. For large console radios like this there is plenty of room on the rear panel to drill a mounting hole for S1 (see **Photo 3**).

Portable Tube Radio "Boom Box"

Battery-powered tube radios are good candidates for this modification. In this example, I modified a pre-war Sentinel battery radio for an external audio input (see **Photo 4**). The same circuit was leveraged, except this time, I placed S1 outside the radio enclosed in heat shrink tubing.

This radio requires two type-B 45-V batteries in series to supply 90 V for the plates and one type-A 1.5-V filament battery. Unfortunately, these batteries are no longer available. For this reason, you must make your own or re-stuff the old batteries (see **Photo 5**). I chose to re-stuff the old batteries. I removed the old battery material, soldered together five 9-V batteries in series for each type B and soldered in one size C battery as

Photo 4: The Sentinel pre-war battery radio has an external audio input that can be connected to a laptop.

About the Author

Gregory L. Charvat only plays vintage music through old radio equipment, is the author of *Small and Short-Range Radar Systems*, co-founder of Hyperfine Research, Inc., Butterfly Network, Inc. (both of which are 4catalyzer companies), visiting research scientist at Camera Culture Group Massachusetts Institute of Technology Media Lab, editor of the Gregory L. Charvat Series on Practical Approaches to Electrical Engineering, and guest commentator on CNN, CBS, Sky News, and others. He was a technical staff member at MIT Lincoln Laboratory where his work on through-wall radar won best paper at the 2010 MSS Tri-Services Radar Symposium and is an MIT Office of the Provost 2011 research highlight. He has taught short radar courses at MIT where his "Build a Small Radar" course was the top-ranked MIT professional education course in 2011 and has become widely adopted by other universities, laboratories, and private organizations. Starting at an early age, Greg developed numerous radar systems, rail SAR imaging sensors, phased-array radar systems; holds several patents; and has developed many other sensors and radio and audio equipment. Greg earned a PhD in electrical engineering in 2007, an MSEE in 2003, and a BSEE in 2002 from Michigan State University. He is a senior member of the Institute of Electrical and Electronics Engineers (IEEE) where he serves on the steering committee for the 2010, 2013, and 2016 IEEE International Symposium on Phased Array Systems and Technology, and chaired the IEEE AP-S Boston Chapter from 2010–2011.

Photo 5: Vintage battery packs supply 90 V across two type-B batteries in series and 1.5 V from one type-A battery. These battery packs can be re-stuffed with modern conventional batteries. In this case, use five 9-V batteries in series to replace one type-B battery and only one size C cell to replace the A battery.

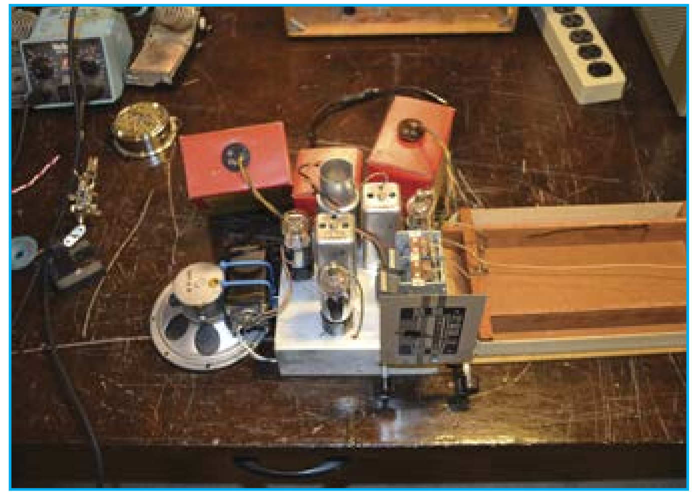


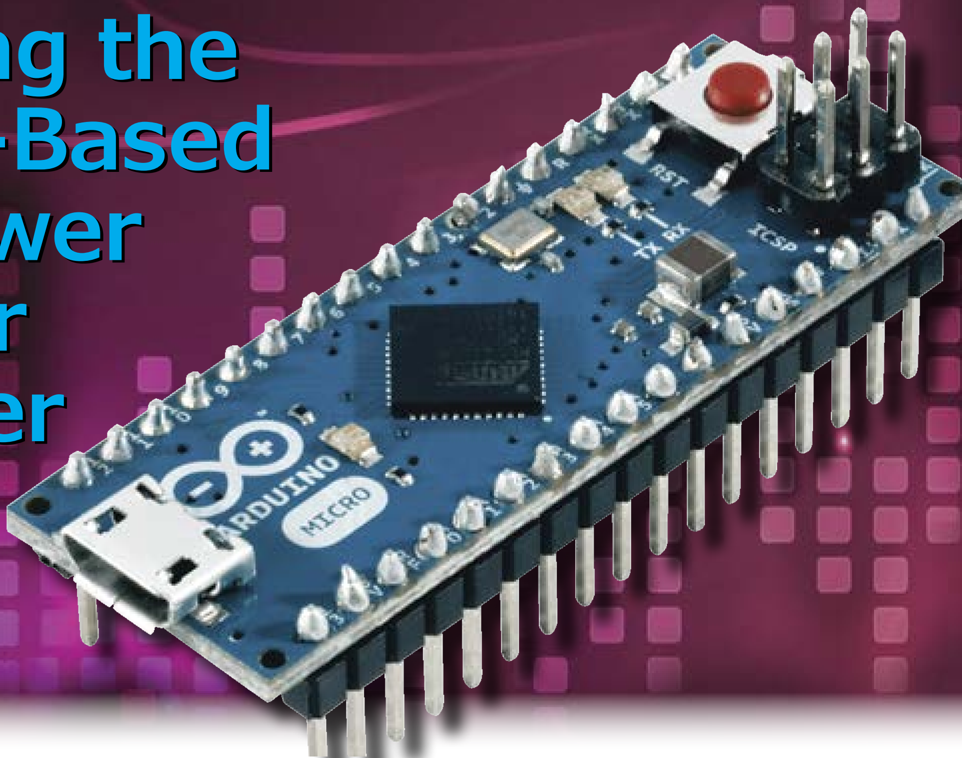
Photo 6: The restored and modified Sentinel battery radio is shown running atop my work bench.

a replacement for the type A. I placed these new batteries into the old battery boxes and stuffed in packing material so that the new batteries would not rattle around inside the boxes. When completed, the original look is maintained but it has new batteries (see **Photo 6**).

Play Vintage Music through the Equipment

It is not always true that lower distortion, high fidelity, and greater dynamic range are better. In my subjective opinion, old recordings sound best when played through the vintage radios from which the music originated.

Extending the Arduino-Based Tube Power Amplifier Controller



In the February issue of *audioXpress*, I discussed a tube power amplifier controller based on the Arduino open-source microcontroller. In an earlier issue, I showed how to build an accurate bias meter, making it simpler to set bias in push-pull tube output stages. Here, I will explain how to use some of the Arduino's untapped capabilities to incorporate the bias meter into the amplifier controller with virtually no additional hardware. A temperature sensor and over-temperature shut-down capability are also included.

By
Mark Driedger
(United States)

The Arduino is a popular open-source microcontroller. For less than $25, you can have a compact and sophisticated controller, plus a huge network of open-source software and support.

In the power amplifier controller project, the Arduino detects the presence of an audio signal and automatically powers on the amplifier, stepping through a timed sequencing of filament and B+ power supplies. It turns off the amplifier several minutes after the audio signal is lost, helping to increase the life of the tubes while preventing unnecessary power cycling. The controller also has an override switch to manually turn the amplifier off or on, independent of the audio signal.

Figure 1 shows how the Arduino audio controller connects to the tube amplifier. The audio detector consists of a hardware precision rectifier combined with software processing. It detects the audio input signal's presence. The state machine is the software that controls the logic and the timing of the two control signals, P0 and P1, based on the audio detector and the mode switch. The P0 signal switches AC to the amplifier's power transformer using an optoisolated, zero switching relay. The P1 signal drives an optoisolated MOSFET switch and controls the B+ power.

My article, "Arduino-Based Tube Power Amplifier Controller" (*audioXpress*, February 2015), has additional details on how the software works and the design of the MOSFET switches based on the IXYS FDA217 isolated MOSFET driver (see Resources).

Bias Meter Recap

The bias meter works by inserting 10-Ω sense resistors, RS, in the cathode legs of the push-pull output tubes. The voltage across the sense resistors is measured and used to control LEDs, which indicate if the bias is properly set. There are two common variations of tube bias controls. I will refer to **Figure 2** as the bias/balance arrangement and **Figure 3** as the independent bias arrangement. The bias meter can work in either configuration. In the first, the average and difference voltages drive the calibration LEDs. In the second, each LED is driven based on an individual tube's sense voltage.

My original design, which I detailed in a 2012 *audioXpress* article uses a hardware window comparator (LTC1042) to control three LEDs (red, green, and yellow) per bias adjustment. In operation, the bias controls are adjusted until the corresponding green LED lights. The desired tube bias, Io, is set with a single trim pot and needs only be adjusted once when the meter is built.

Combined Design

In the bias meter's new implementation, I simply connected the bias sense resistor voltages to four of the unused analog to digital converter (ADC) inputs on the Arduino and drove the indicator LEDs using a software version of the window comparator function. **Figure 4** shows the combined design.

An extra Arduino input pin (D5) is used to select the bias mode. If this input is grounded, then each window comparator directly uses the sense voltage (independent bias mode). If the input remains unconnected, then the average and the difference are taken before the window comparator (bias/balance mode).

To save on output pins, the LEDs are driven in a multiplexed configuration. They are arranged in a matrix (four LEDs × three colors), and the software quickly cycles through them, lighting one LED in the matrix at a time by controlling the associated row (LED) and column (color). This happens faster than the eye can detect, so the LEDs appear to be continuously on.

Rather than use three physically separate LEDs for each bias point as in the earlier design, I selected an Red/Green/Blue (RGB) LED. These have three LED devices in one package, in this case with a common anode. That way you only need a single mounting hole adjacent to each bias control to mount the LED. Thanks to a bit of extra software, the bias error is more accurately indicated with a combination of solid and flashing LED colors (see **Figure 5**).

To turn on calibration mode, a double pole single throw (DPST) switch is used to take the Manual ON and Manual OFF inputs both low. The software interprets this as calibration mode. If the calibration mode is selected while the amplifier is off, then it is turned on and goes through the normal power cycling. The desired bias point is set in the software and easily modified by the builder, as is the nominal 2-mA sensitivity. If you adjust the 2-mA threshold, then the 5- and 10-mA thresholds will proportionally scale. With 10-Ω sense resistors, the full-scale current that the meter can measure is 250 mA per tube.

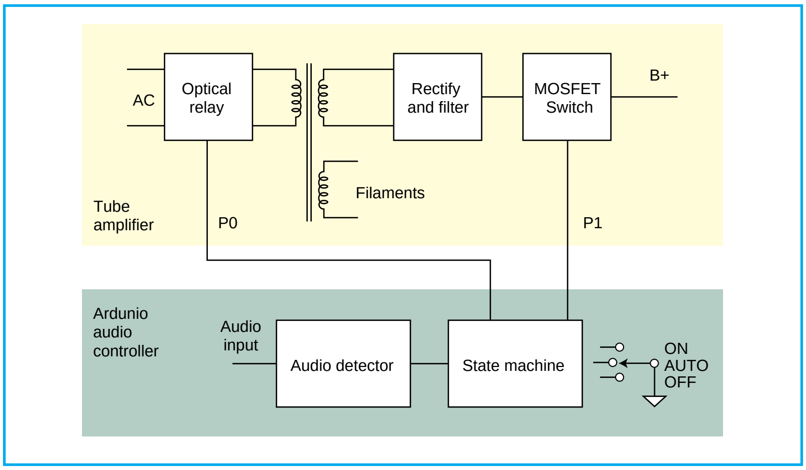


Figure 1: The block diagram details how the Arduino audio controller connects to the tube amplifier.

To use the bias meter, remove the audio input to the amplifier and turn on the calibration switch. Adjust each bias control until the corresponding LED is solid green. It is good practice to let the amplifier stabilize for 30 min or so prior to adjusting bias.

I added an LM34 temperature sensor and software functionality to shut down the amplifier in an over-temperature condition. The LM34 provides an accurate output of 10 mV/F in an inexpensive TO-92 package. If the temperature exceeds 120°F then the amplifier shuts down and the bias LEDs are alternately flashed

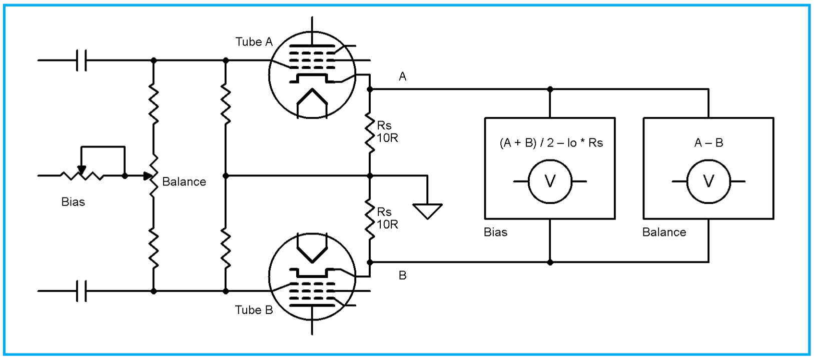


Figure 2: This is one of two common variations of tube bias controls—the bias/balance configuration.

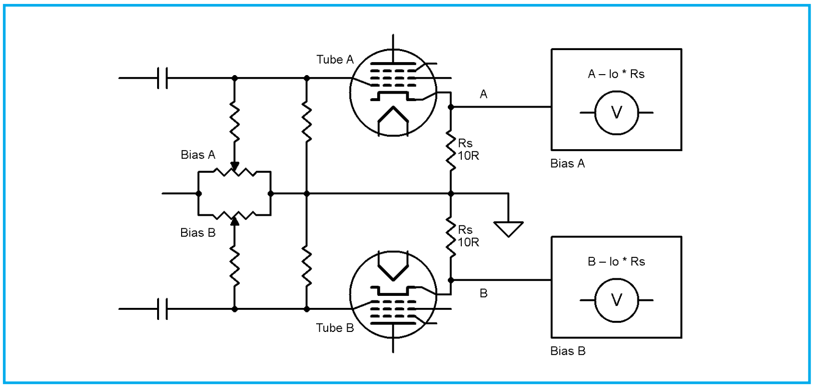


Figure 3: This is the other common variation of tube bias controls—the independent bias configuration.

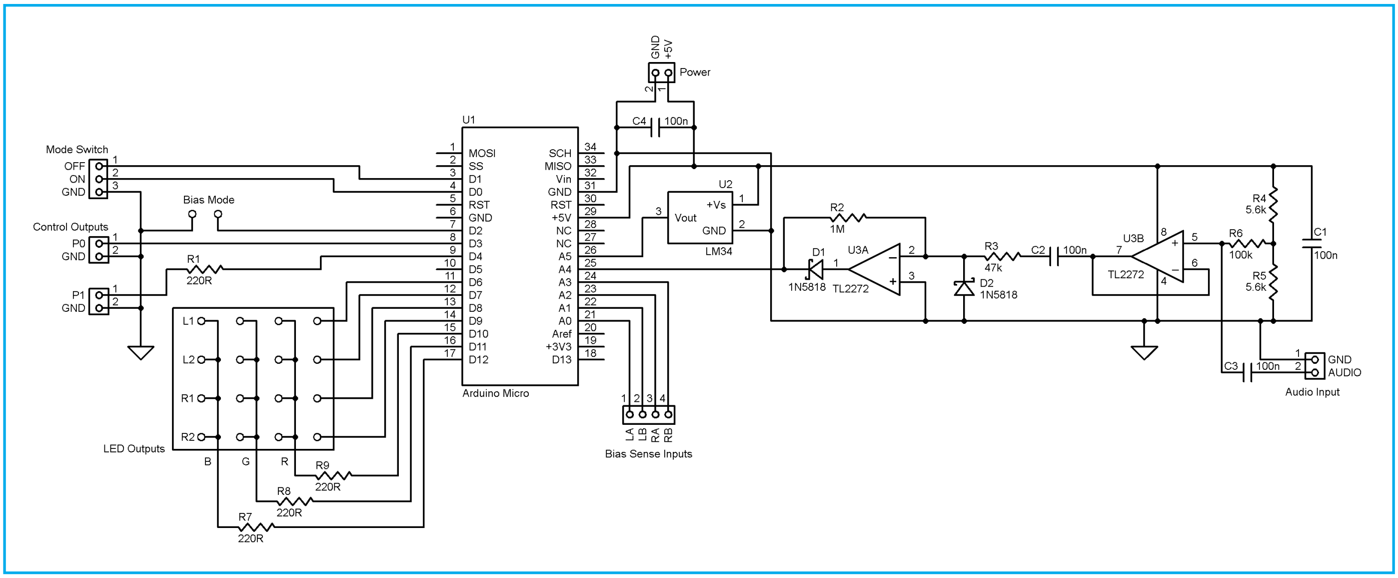


Figure 4: The schematic details my new implementation of the bias meter.

red at 2 Hz. When the temperature drops below 120°F the amplifier will restart in whatever mode it was in previously (manual on, automatic, or calibration). You can adjust the threshold in software.

I made a couple of simplifications to the controller portion compared to the earlier design. Only one analog input is used—since we just need to sense the presence of audio, either left or right channel will suffice. Also, I removed the ability to independently control the output and driver B+ voltages (which eliminates the P2 control from the original design).

Building It

The PCB design, shown in **Figure 6a** (component side) and **Figure 6b** (solder side), is 3.75″ × 2.25″, single sided and uses through-hole components to make construction easier. A soft copy of the layout for PCB Artist software is available in the Supplementary Materials section of *audioXpress*'s website. If you are using the bias-only configuration, then short the "BIAS MODE" pins together with a jumper. For bias/balance mode, leave the "BIAS MODE" pins unconnected. If you are not using the LM34 temperature sensor, then

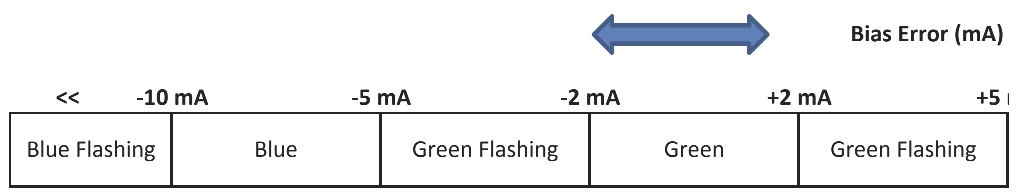


Figure 5: The solid and flashing LED colors indicate various bias errors.

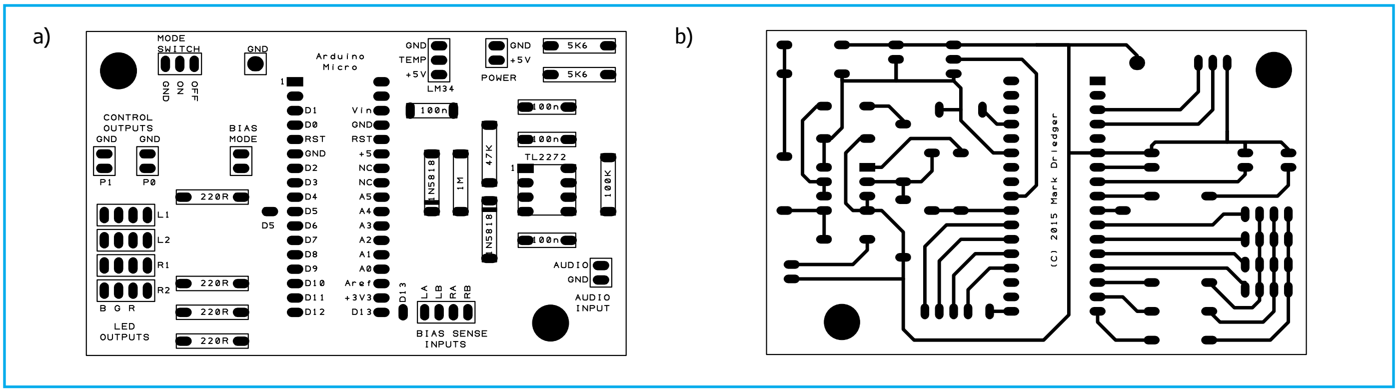


Figure 6: The PCB design is shown on the component side (a) and the solder side (b). Note, the layouts shown here are not to scale. The properly scaled versions can be found in the Supplementary Materials section of the *audioXpress* website.

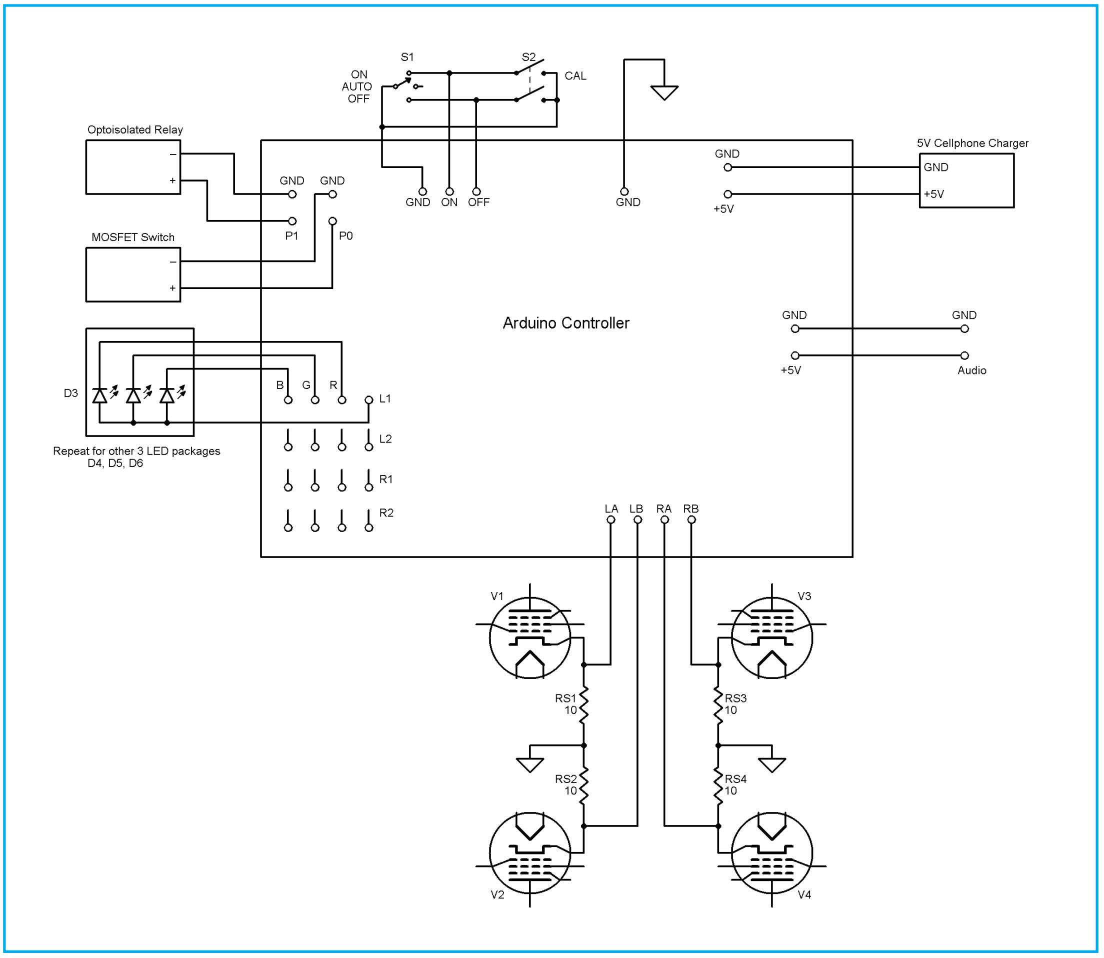


Figure 7: The controller is connected to the amplifier and continuously powered by a small cellphone charger.

short the TEMP and GND pins where the LM34 would normally be installed. For this design, I have assumed the LEDs would be mounted adjacent to each bias pot and wired to the PCB. If instead you want to directly install the tricolor LEDs on the PCB, you will need to insulate the common anode (long lead) and bend the leads to swap the red and common anode leads. You may also need to swap the blue and the green leads as there are two pinout variations on the market.

Figure 7 shows how the controller is wired to the amplifier. It is continuously powered by a small 5-VDC cellphone charger. Make sure that there is a ground connection between the controller and amplifier. More details on the MOSFET switch design can be found in my February article.

Always observe proper safety precautions when working on the AC voltages and tube amplifier high-voltage supplies. Unplug and safely discharge all the capacitors before working on the amplifier. If you are not comfortable doing something, then don't attempt it.

Installing Software and Configuring

My February article also contains full instructions on installing and running the Arduino software. Here is a summary:

- Download and install the Arduino Integrated Development Environment (IDE) software from www.arduino.cc.

Project Files

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

Resources

M. Driedger, "Arduino-Based Tube Power Amplifier Controller," *audioXpress*, February 2015.

——, "An Accurate Bias Meter for Push-Pull Tube Output Stages", audioXpress, July 2012.

Sources

70S2-04-B Opto-isolated relay

Magnecraft (now Schneider Electric) | www.schneider-electric.com

TLC2272CP Op-amp

Texas Instruments, Inc. | www.ti.com

- Retrieve the software "sketch" from the *audioXpress* supplementary materials by downloading and unzipping the file "AC_V2_00.zip". This will create a directory called "AC_V2_00" with two files. Move this directory to the default Arduino sketch folder (look under File, Preferences in the IDE application if you are not sure where this is).
- Open the software sketch from the IDE application (File, Open, "AC_V2_00").
- Change any of the parameters you want to modify. They are defined at the top of the file.
  - o Filament warm up delay (ON_DELAY, default 10 s)
  - o Amplifier shut down delay (OFF_DELAY, default 45 s)
  - o Target amplifier bias current (TARGET_BIAS, default 62.5 mA)
  - o Window comparator sensitivity (THRESHOLD_BIAS, default 2.0 mA)
  - o Over-temperature shutdown threshold (THRESHOLD_TEMP, default 120.0 F)
  - o Output tube cathode resistor (RS, default 10 Ω)
- Connect the Arduino module to your computer using the USB cable and upload the software (File, Upload). You may need to select the Arduino type (micro) and COM port (varies with your computer) from the Tools menu in the IDE.
- Disconnect the USB cable and install the Arduino module in the controller circuit. If you need to reprogram it, remove the Arduino module from the power amplifier for safety.

Parts List

Capacitors

C1–C4	100 n

Components

U1	Arduino micro module
U2	LM34 temperature sensor
U3	TL2272 or similar single-supply dual op-amp

Diodes

D1, D2	1N5818
D3–D6	RGB LED, common anode

Fuses

S1	SPDT, center off toggle switch
S2	DPDT toggle switch

Miscellaneous

Cellphone charger	5 V
Optoisolated relay	
MOSFET Switch	

Resistors

R1	220 Ω, 0.25 W
R2	47 kΩ, 0.25 W
R3	47 kΩ, 0.25 W
R4, R5	5.6 kΩ, 0.25 W
R6	100 kΩ, 0.25 W
R7–R9	220 Ω, 0.25 W
RS1–RS4	10 Ω, 0.5 W

Final Thoughts

The Arduino's software flexibility makes it easy to implement a wide range of functions with a simple hardware design. And, the Arduino can do much more. The open-source Maker Movement is overflowing with interesting possibilities. You can find online projects for preamplifier switching and control, MP3 players, DACs (e.g., http://hifiduino.blogspot.com), and many types of test equipment. With a bit of time, I am sure you can add your own innovative applications to the mix.

About the Author

Mark Driedger was born in Canada in 1963. He has an MSc in electrical engineering from the University of Waterloo and he has been experimenting with tube audio since 1980. Mark has worked in the telecom industry for 25 years in various technical, business, and executive roles. He is currently Vice President for a division of Tektronix and lives in Dallas, TX. His other passions include his wife and two children, woodworking, and the guitar.

Build a Push-Pull Triode Power Amplifier

Photo 1: The JJ Electronic ECC 99 vacuum tube has been available for many years.

This Design Uses ECC 99 and ECC 832 Tubes

Tube technology beginners often shy away from higher operating voltages. However, many of them also want to try the somewhat unusual tubes first. Tube enthusiasts want a simple and convenient power amplifier with low performance that is supposed to offer a good sound when connected to high-efficiency loudspeakers. Sometimes, they want a small power amplifier (e.g., to playback music from an MP3 player or to listen to the radio, requiring only low amplifier ratings for background music). Last but not least, many guitarists need a small tube power amp for practicing. This design would work for any of these scenarios.

By
Gerhard Haas
(Germany)

Original article published in German in Elektor's *Roehren Sonderheft 7, June 2011*

For this design, I wanted to build an amplifier with a few watts of amplifier power that would provide a good sound with little effort. Traditional tube receivers with single-ended type-A power amplifiers mostly have an output of approximately 2 to 5 W at a total harmonic distortion (THD) of 10%! The JJ Electronic ECC 99 vacuum tube has been available for several years (see **Photo 1**). With a relatively low anode voltage and a sufficiently high anode current, when utilizing the admissible anode power loss, you can achieve an output power of 3.5 W at 250 V, and a THD of only 2%!

The JJ Electronic ECC 832, being a preamplifier and a phase splitter tube, features two different systems in one bulb, which is apparent in its type designation. The "8" stands for a nine-pin socket, the "3" stands for a system of the ECC 83 and "2" stands for a system of the ECC 82. This practical tube is available in the US under the designation 12 DW 7 or 7247. In this case, with minimum effort, you have a high-gain triode as well as another high-current triode system in one bulb. Half, the ECC 83, is used for amplification, and the other half, the ECC 82, is used for phase splitting.

A small power amplifier like we are presenting here only requires two tubes. One tube provides idle amplification and the other half allows the application of low-impedance circuit technology, which is less prone to interference and less problematic for tube technology beginners. In this case, we deliberately chose the triode push-pull AB operation.

An output of 3 W can also be achieved, using a single-ended type-A power amplifier. In this case, the single-ended transformer should be good (i.e., feature a frequency response of 20 Hz to more than 20 kHz under load). So, the effort concerning the core and the winding rapidly increases, which in turn costs a lot of money. Virtually all the transformers in the old radios are too small and show a significant drop below 100 Hz. However, it doesn't look much better with

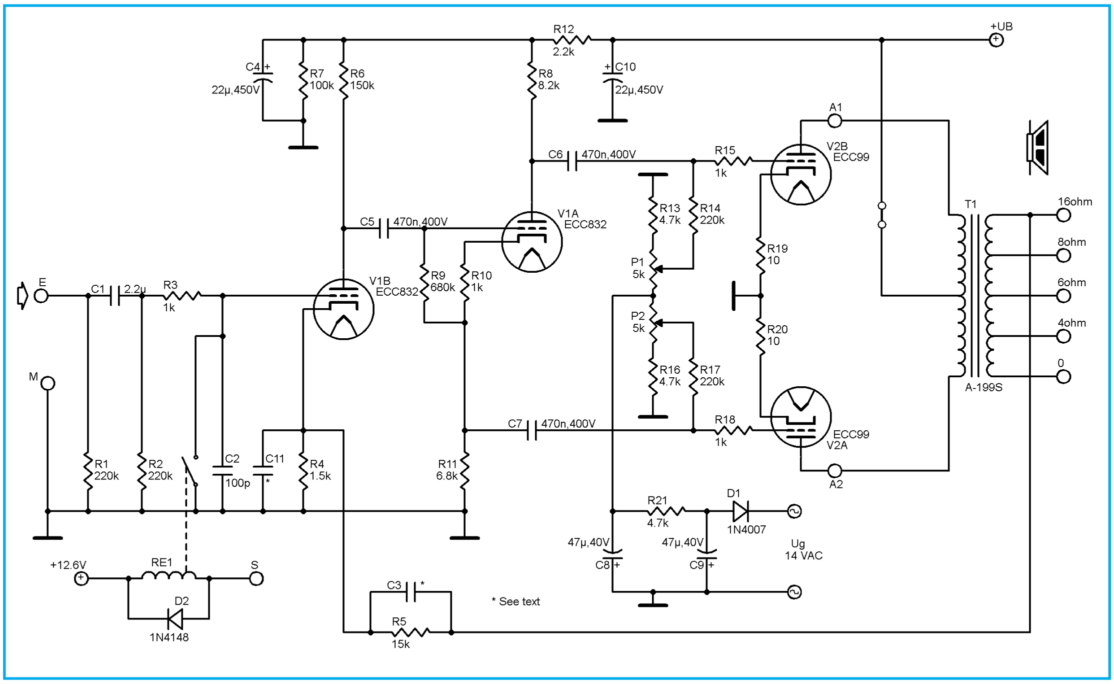


Figure 1: The push-pull triode power amplifier's design shows a typical circuit.

regard to the upper frequency limit. The often much too small transformer core saturates at full output level and low frequency, which causes bad distortions. The problem always is the direct current (DC) and today's frequency range demand of approximately 20 Hz to more than 20 kHz. This is also why a small but hi-fi-ready single-ended type-A transformer quickly becomes quite expensive. However, you can avoid the problem with a push-pull amplifier. Due to the omission of a DC preload, the transformer core can have substantially smaller dimensions while still reaching the lower cut-off frequency of 20 Hz. For those who want to use this small power amplifier as a practice amplifier, it should be capable of dealing with 80 Hz as a lower cut-off frequency for guitars, 40 Hz for four-string bass guitars, and 30 Hz for the five-string versions. The low-frequency requirements of a practicing amplifier don't differ too much from those of hi-fi equipment, so the transformer and the power amplifier can be designed for hi-fi quality.

The Circuit

Figure 1 shows the circuit. It doesn't show any unusual features. In front of the grid of ECC 83, the usual resistor capacitor (RC) element is installed. Furthermore, a Reed relay is provided with which the input can be muted. ECC 83 assumes the main amplification, ECC 82 provides the phase splitter. The idle currents of both systems of the ECC 99 are set by means of potentiometers. The idle current is set so the power amplifier works in Class-A operation and the admissible anode power loss doesn't exceed 5 W. The individual idle current setting gives full control over any tube system. The voltage drop is measured at the cathode resistors R19 and R20 at the measuring points against ground, which is a measure of the idle current.

In case of a voltage drop of 200 mV, the idle current is 20 mA, which results in 5 W at an operating

Photo 2: The tube sockets can be mounted so they stand out from the chassis.

About the Author

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

current of 250 V. The coupling capacitors are of relatively large dimensions. This is natural in low-impedance circuit technology so as not to have a drop in low frequencies already in the amplification path. The capacitor C3 is used only if there is a high-frequency oscillating tendency. In principle, the amplifier is very stable. However, if there is a high-frequency oscillating tendency, C3 needs to be fitted. In this case, reference values are 10 to 47 pF. Shielded cables are used for the negative feedback line so that the amplifier does not catch any disturbances here.

If you take a closer look at **Figure 1**, you will notice that the cathode and anode resistance of the phase splitter show different values. This is not a typographical error, but true intention. Push-pull power amplifiers suppress the even distortion contents, preferably the second harmonic.

If you are hoping for a beautiful triode sound with this amplifier design, you will be disappointed. The pleasant distortion content second harmonic (triode sound) is very low. However, the annoying third harmonic (pentode sound) is very high. This problem cannot be solved by taking measures in terms of negative feedback. Therefore, an artificial unbalance is implemented, so the second harmonic is dominant and the higher distortion contents show dropping levels. Then again, you have the triode sound. Some guitar amplifier manufacturers follow the same

Parts List

(All resistors are metal film, 0.6 W, 1 % tolerance or metal oxide (MO), 2 W, 2 % tolerance.)

Capacitors
C1 . . . . . .2.2 µ/50 V, nonpolar
C2 . . . . . 100 pF, ceramic
C3 . . . . . 10–47 pF, only when ringing (see text)
C4 . . . . . 22 µ/450 V
C5,6,7 . . 0.47 µ/400 V MKP 10
C8,9 . . . . 47 µ/40 V
C10 . . . . 22 µ/450 V
C11 . . . . 1 nF or 1.5 nF, MKT (see text)

Diodes
D1 . . . . . 1 N 4007
D2 . . . . . 1 N 4148

Resistors
R1,2 . . . . 220 kΩ
R3 . . . . . 1 kΩ
R4 . . . . . 1.5 kΩ
R5 . . . . . 15 kΩ
R6 . . . . . 150 kΩ, MO
R7 . . . . . 100 kΩ, MO
R8 . . . . . 8.2 kΩ, MO
R9 . . . . . 680 kΩ
R10 . . . . 1 kΩ
R11 . . . . 6.8 kΩ, MO
R12 . . . . 2.2 kΩ, MO
R13 . . . . 4.7 kΩ
R14 . . . . 220 kΩ
R15 . . . . 1 kΩ
R16 . . . . 4.7 kΩ
R17 . . . . 220 kΩ
R18 . . . . 1 kΩ
R19,20 . . 10 Ω
R21 . . . . 4.7 kΩ

Miscellaneous
P1,2 . . . . 5 kΩ trimmpot cermet
Rel1 . . . . SIL-relais 12 V SPST
2 . . . . . . Noval sockets, ceramic, pc mounting
1 . . . . . . PC board, epoxy resin with glassfiber, 70- µm copper laminated (69 mm x 131 mm)

Transformer
T1 . . . . . Output transformer A-199 S

Vacuum Tubes
V1A/V1B . ECC 832
V2B . . . . ECC 99

Technical Data

Output power	3.5 W with THD at 2%
Input voltage for maximum output level	1 V
Frequency response at 1 dB with maximum output level	13 Hz (–0.5 dB), 38 kHz (–1 dB), or 87 kHz (–3 dB)
External voltage with input closed with 1 kΩ	–95 dBV (17 µV)
Harmonic distortion at 1 dB below maximum output level	THD at 0.92%
	second harmonic at 0.77%
	third harmonic at 0.17%
	fourth harmonic at 0.06%
	fifth harmonic at 0.04%
Operating voltage	250 V, maximum 260 V/approximately 50 mA
Filament voltage	12.6 V/0.55 A

The kit, PC board and special parts are available at www.experience-electronics.de.

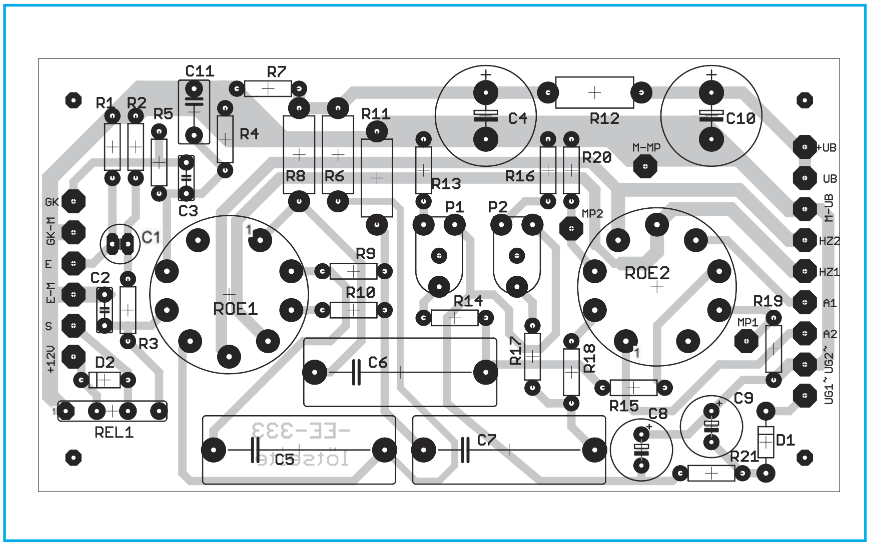


Figure 2: The PCB details the push-pull triod power amplifier's layout.

procedure. Whether guitar amplifier sound or hi-fi, a second-harmonic-focused amplifier sounds better.

The Power Supply

When it comes to the preamplifier tube ECC 83, the capacitor C11 is provided, bridging the cathode resistor R4. Depending on the circuit design and the resulting switching capacitances, the frequency response can show a minor drop at 20 kHz. The aforementioned capacitor can compensate for this. Use approximately 1 to 1.5 nF as reference values.

Especially for smaller power amplifiers, you should pay particular attention to the power supply. It was normal for early steam radios to hum. Today, more attention is paid to sound quality, and you want as little background noise as possible, so don't try to save money on the power supply.

The high voltage can be treated with a C-L-C filter, or you can alternatively use a stabilized power supply. We also recommended using a regulated DC heating. Provided that the system is neatly built, it will have the lowest possible interference, which is beneficial when using high-efficiency loudspeakers. This also applies to guitar amplifiers, because in this case, mostly high-efficiency loudspeakers are used. The provided technical data shows the requirements in terms of currents and voltages for one channel. The filaments of the tubes are connected in series for a voltage of 12.6 V, which reduces the power loss at the heating voltage control.

The tube sockets are equipped from the solder side. In this way, the board can be mounted so the tubes stand out from the chassis (see **Photo 2**). It is a nice design and as a side-effect, good heat dissipation is guaranteed.

Although screen grid tapping is not necessary with triodes, they have been provided at the output transformer. In this way, it can also be used for small pentode power amplifiers. The secondary side provides outputs for 4, 6, 8, and 16 Ω. This enables you to use various loudspeakers and headphones. At full output level, about 7.5 V are available at 16 Ω, 3.75 V at 4 Ω. This is enough for operating virtually all dynamic headphones. For headphone operation, it is recommended terminating the 16 Ω output with a resistor featuring 47 Ω and a 2-W rating so that there is always a base load and the voltage cannot ramp up in case of modulations in the transformer. **Figure 2** shows the board's design.

Why Tubes Sing and Crackle

Some audio questions are repeatedly asked, and although partially answered in some publications, reappear almost regularly. This article hopes to answer some of those questions, including why tubes sing and crackle, audio transformers hum, and many filaments glow brightly when switching on.

By
Gerhard Haas
(Germany)

Original article published in German in Elektor's *Roehren Sonderheft 4*, September 2008

After completing a DIY tube amplifier and correctly setting the power amplifier's idle currents, there are tests to perform using the sine generator, the oscilloscope, and the load resistance. The tests help verify whether the amplifier correctly works. Normally, the first test begins at 1 kHz.

During the testing, you might hear a singing noise coming from the tubes and the output transformer as the signal level increases. This is hardly noticeable with small power amps. However, when it comes to the performance class of 30 W and higher, the power amplifier can indeed start "singing." Those unfamiliar with this effect often search for errors that don't even exist. Or, they might call the supplier and ask if microphonic substandard tubes and not quite flawless transformers were shipped. However, this is not usually the cause. What happens inside the amplifier and why this singing takes place requires further investigation.

Singing Amplifiers

To determine the cause, it is best to refer to physics for general validity for things in our universe. The grids of amplifier tubes are made of very thin wire, which is about 15-to-200-µm thick. An average human hair has a diameter of about 40 µm. Also, the distance between the cathode and the control grid should be as small as possible to get the optimum control effect. Looking at the most frequently used power tubes, this distance is in the region of about 150 to 250 µm. If a typical power tube is fully driven, the grid voltage may vary between –5 and –50 V. If there is a distance of 200 µm between the cathode and the control grid, the field strength may show values between 250 and 2,500 V/cm.

A pentode's screen grid normally shows a potential that corresponds to that of the anode or nearby values. Although the distances to the next electrodes are much bigger than that between the control grid and the cathode, the voltage differences are substantially higher. On the one hand, the screen grid lies between the suppressor grid, which has cathode potential or is connected to ground and shows no voltage changes when driving the tube. On the other hand, it stands opposite to the control grid which, depending on the level, can achieve more or less negative values in contrast to the screen grid. Here, the screen experiences a much more negative potential than in contrast to the suppressor grid. Assuming that the amplifier is operated in ultra-linear mode, which is common with hi-fi amplifiers, the screen grid voltage oscillates according to the modulation level.

So, if the control grid shows the lowest negative voltage and the highest current, the screen grid potential and the anode potential will also be as low as possible. Therefore, the voltage difference between the screen and the control grid will be lowest. The potential difference between the two grids is highest, if the current is lowest. From this, it follows that forces are acting on the grids, depending on the level.

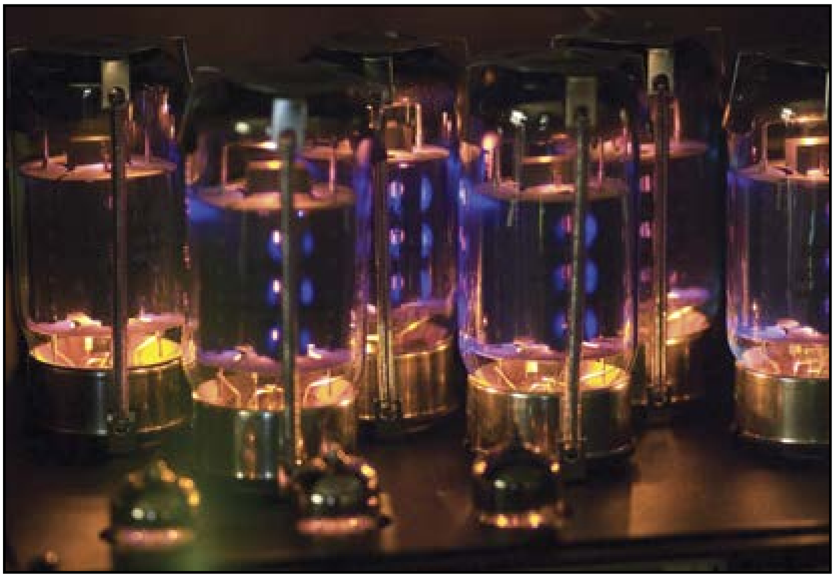
Since the grid wires are thin, they will inevitably start moving due to the influence of the varying field strengths. This motion is transmitted via the fixing rods of the grids onto the mica plates that fix the electrodes to the other tube components and finally to the glass bulb, which results in an audible "singing" of the tube. The load resistance connected to a tube power amplifier used to test the amplifier can also "sing," depending on the construction of the resistances and how they are mounted. In this case, the noise is often drowned out by the transformer and the particularly loud tubes. However, this effect has often been observed during amplifier tests.

Cause of the Singing

Inside the transformer, it is the electromagnetic fields that cause the "singing." When power is applied to the coil, it creates a magnetic field, which is concentrated by the iron core. This causes the core and the coil to contract and expand once power decays. If an audio transformer is non-impregnated, the material can easily work mechanically. Non-impregnated, dip-coated, or badly impregnated mains transformers eventually start buzzing because the windings, the insulation, and the iron core are not tight enough.

The other extreme is a vacuum impregnation under the best conditions. This ensures that the core and the winding are as tight as possible. However, even in this case, a residual singing is not completely avoidable. Powerful transistor power stages, as well as dimmers are good examples. In this case, semiconductors are used as products of solid-state physics. Also here, powerful power amplifiers "sing" and dimmers hum with increasing loads. It's always the electric fields that make the material work.

The looser the component is, and the more it is stressed, the louder the "singing." The frequency and the level setting also play a role in the level.



If your tube power amplifier starts "singing," don't immediately start searching for errors that don't exist.

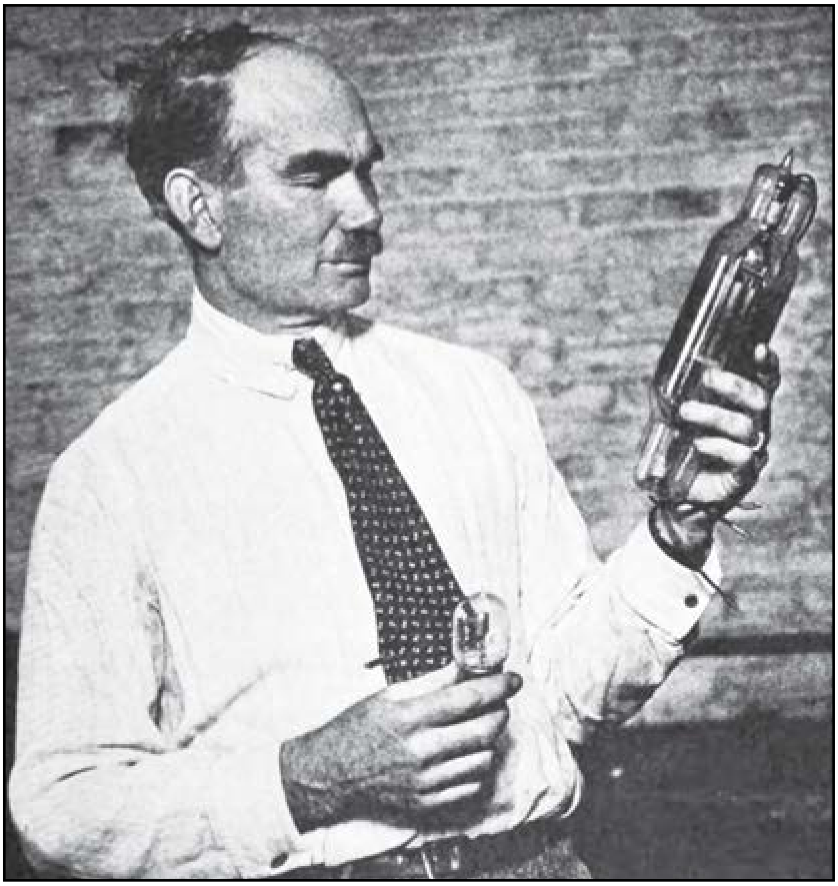
High-voltage electricity lines are certainly not filigree. However, if you stand under them, you clearly hear the basic frequency of the mains grid, which is 60 Hz. You can also hear a high-frequency buzz deriving from the various loads resulting from harmonics and various frequencies that stray around in the mains grid or are otherwise picked up. The high-voltage electricity lines periodically attract and repel each other, which cause the noises.

So, if an amplifier "sings" at a high level setting, this has nothing to do with material defects or with low-quality components, except for the non-impregnated or only dip-coated transformers that tend to make noise anyway. Here, physics alone is responsible. When listening to quiet music, the above-mentioned forces only act to a small extent and remain unnoticed.

When listening to loud music, the loudspeakers drown these background noises by a multiple so that they carry no weight. If only a load resistance is connected, these noises are not drowned out. Now,

About the Author

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



Lee de Forest (August 26, 1873–June 30, 1961) was an American inventor with more than 180 patents to his credit. In 1906, de Forest invented the Audion, the first triode vacuum tube and the first electrical device that could amplify a weak electrical signal and make it stronger. (Image courtesy of http://en.wikipedia.org/wiki/Lee_de_Forest)

you could start philosophizing about the impact of these effects on the sound. While doing so, go back to the discussion about the "sound" of cables and solder types without really considering the physical principles. One has to accept that everything in this world has its own limits and that any feasible technology is subject to the unshakable laws of physics.

The whole phenomenon can be phrased in a different way. If electricity flows through two parallel conductors, they attract or repel each other, depending on the current direction in the conductors. With low currents, you hear and see nothing. If the currents are very high and if the current's frequency is in the audible range, this effect can be perceived. If you put a little stone onto a railroad track, the latter will deform under it due to the force of gravity that the mass of the little stone applies to the track. This may not be visible to the naked eye, but it is definitely measurable! At Carl Zeiss (Oberkochen, Germany), large expensive measuring devices are built that enable you to measure down to an atomic level.

Crackling Tubes

Power tubes crackle, too. This is less perceived when the power supply is switched on than when it is switched off. When switching on an amplifier, you can listen to music after just a few minutes. The crackling deriving from the tube's increasing heat and the associated expansion of the tube components is not usually perceived anymore.

After switching the power supply off, the crackling can be heard for a longer time because the material contracts again. If no music is played, nothing can drown out the crackling. Therefore, it is clearly noticeable. Even that is not a quality defect. Depending on the temperature, the material expands or contracts. The glass bulb acts as a resonance body that gains these mechanical noises. Although the material expansions are only in the micrometer range, they are enough to create the noises.

You also have to consider that temperature changes with power tubes are significant. When switching on, a tube can heat up from 70°F room temperature, to more than 400°F on the outside of the bulb. It is even hotter inside at an assumed temperature change of about 480° (e.g., copper expands by about 4%, iron and nickel by approximately 3%).

Copper rods are used as supports for the grid spirals. The standard anode consists of iron, and the cathode of nickel. With tubes such as EL 34, 6L6, and similar sizes, this means a 0.25-mm length expansion with a temperature change from room temperature to full-operating temperature. Since the temperature-driven movements of materials have been known since the early days of tube technology, appropriate constructive measures have been taken.

The best way to demonstrate that is using the 300 B, where the filament is suspended on springs. If the filament elongates with increasing temperature, the springs keep it under tension. In this way, the distance to the control grid is constant, regardless of the temperature. Similar precautions are taken in every light bulb to let the glow wire move according to the temperature. Otherwise, it would break.

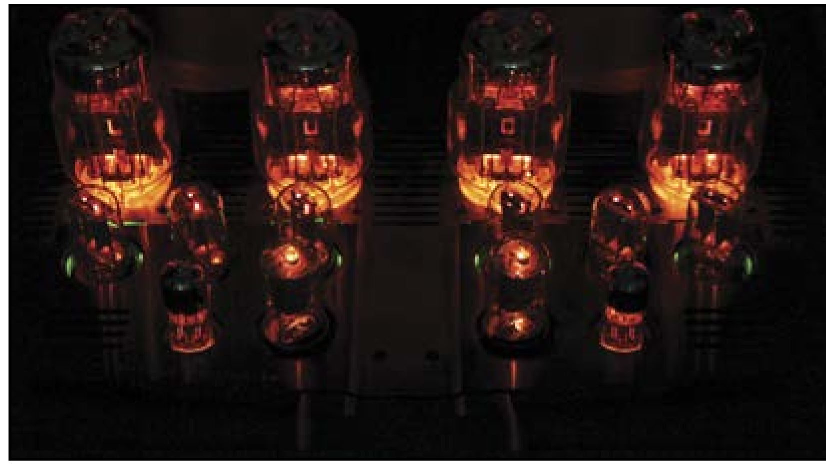
Preamplifiers Decrease "Singing"

If a preamplifier is put into operation, one normally won't hear any "singing." The explanation is simple. Preamp tubes only use a fraction of the energy processed inside power tubes. Due to the low energetic and thermal load, you would virtually hear no "singing" or crackling, even though it actually takes place.

There is one other effect that occurs for which the amplifier designers and tube suppliers are often blamed. With many tubes from different manufacturers, the filaments at one point glow brightly for a short time when you switch on the amplifier. Then, they become darker and glow normally. This effect scares a lot of people.

In earlier days, the radios, the jukeboxes, the TV sets, and so forth used to be completely closed so nobody noticed it, nobody cared, and the devices kept working for years without any problems. Since the renaissance of tube technology in the audio frequency range, the tubes have become visible because the beautiful glow and the look contribute to the sound experience. You can also see what happens when the amplifier switches on. But there is no reason to panic. Depending on the filament design, the tubes can start to glow slowly and continuously or brightly light at the moment with the lowest impedance, especially during the start-up, before returning to a normal glow.

For many decades, both filament variants have provided reliable service in hundreds of millions of tubes and devices. The brightly lighting filaments can be reduced when using a regulated DC-heating with soft start.



When switching on the amplifier, the filaments at one point glow brightly for a short time before they become darker and glow normally. This effect scares a lot of people. However, there is no reason to panic.

Facts of Physics

To sum up, neither the "singing" of tubes and transformers nor the crackling and bright glow of the filaments during power up are quality defects. They are caused by technical and physical effects that do not have any significant impact on your listening pleasure. Be it a DIY amplifier or an expensive ready-made device, you will find the same effects everywhere and without exception.

Pro Tube Power Amplifier Circuits

In this article, we explore a few of the more well-known vintage pro tube power amplifier circuits. We also learn why they remain popular today.


Photo 1: The pair of 300A power triodes and the 274A full-wave rectifier can be seen in this photo of a Western Electric WE 92A power amplifier.

By
Richard Honeycutt
(United States)

The first significant sales of hollow-state devices were for military applications. After World War I, huge numbers were sold for home radio receivers. After the introduction of the electric phonograph in the late 1920s, vacuum-tube phonograph amplifiers became a factor in the market, though they were never as large a factor as radio receivers. In fact, to offer home music listeners an alternative to purchasing separate electronics for playing phonograph records, "phono oscillators" were introduced. These were small AM radio transmitters that would enable the user to transmit the signal from a phonograph player to a radio receiver.

The development of "talkies" (motion pictures with synchronized sound) in the late 1920s paralleled the growth in popularity of electric phonographs. While the number of vacuum tubes employed in cinema amplifiers was even smaller than the number sold for phonographs, the technology of cinema sound marked the real beginning of the pro audio industry.

Probably the most momentous event in early pro audio was Rudy Bozak's system used at the 1939 New York World's Fair in Flushing Meadows. This system employed eight 27" speakers, each feeding a horn with a 14′ mouth, and driven with a 500-W power amplifier built around a radio transmitting tube. A separate group of horns covered high frequencies. The system provided sound for audiences of more than 100,000. (Eat your heart out, Grateful Dead "Wall of Sound.")

Western Electric's Early Days

Despite Bozak's sensational accomplishment, the bread and butter of pro sound remained in the cinema market. Western Electric and RCA owned the cinema amplifier market, which pretty much defined commercial sound in its early days.

Western Electric was founded in 1872, acquired by the Bell Telephone company in 1881, and incorporated as a subsidiary of AT&T in 1915. The company's Vitaphone system for presenting motion pictures with sound was introduced in the mid-1920s.

Photo 1 shows a Western Electric Model 92A amplifier. **Figure 1** shows the schematic of the WE 92A amplifier, produced in the mid-1930s. As with many professional amplifiers of the time, its input was transformer-coupled, using a transformer with

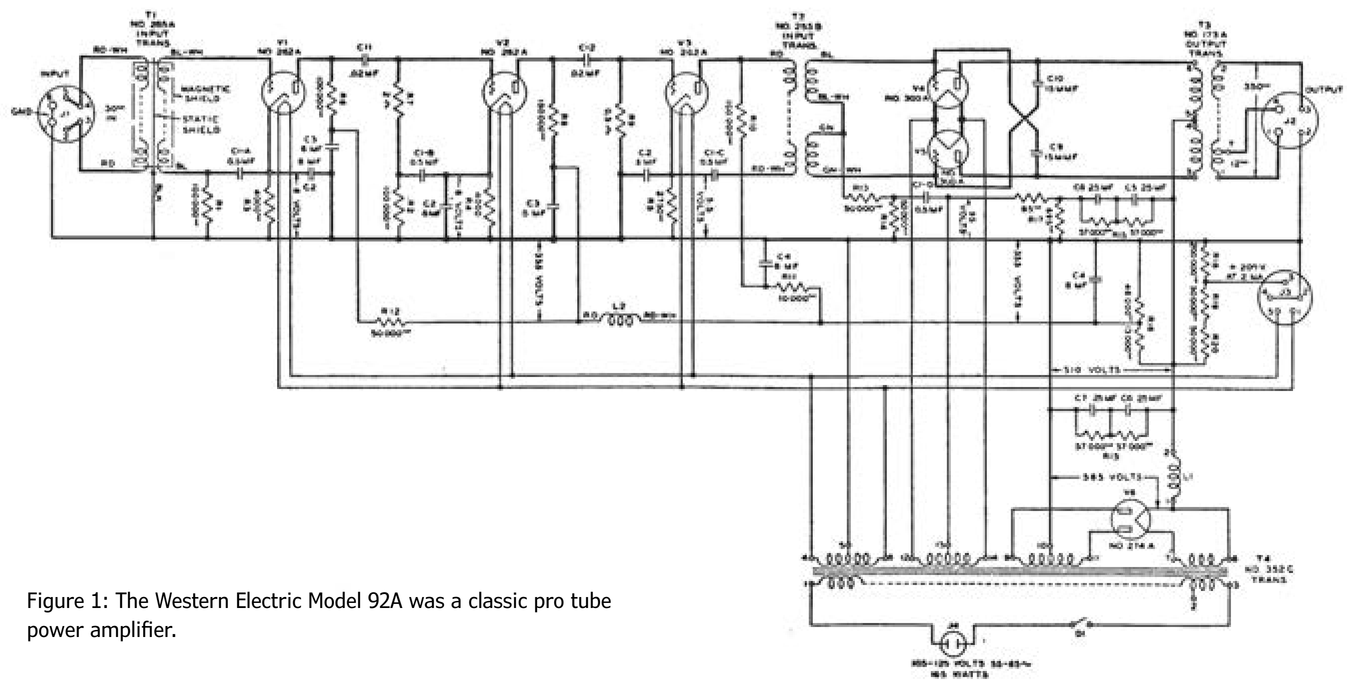


Figure 1: The Western Electric Model 92A was a classic pro tube power amplifier.

magnetic shielding to block hum and noise from external magnetic fields, and internal electrostatic shielding between the primary and secondary windings. Tubes V1, V2, and V3 (all triodes) served as voltage amplifiers. The push-pull pair of model 300A triodes was fed by a phase-splitting interstage transformer. B+ power was supplied via a 274A full-wave rectifier feeding an inductive-input inductor capacitor (LC) filter.

The cathode was biased at +95 V through a voltage-divider from the B+ supply and the control grids were fed by the driver transformer secondary,

Figure 2: The RCA mi12246 was another cinema amplifier of the 1930s.

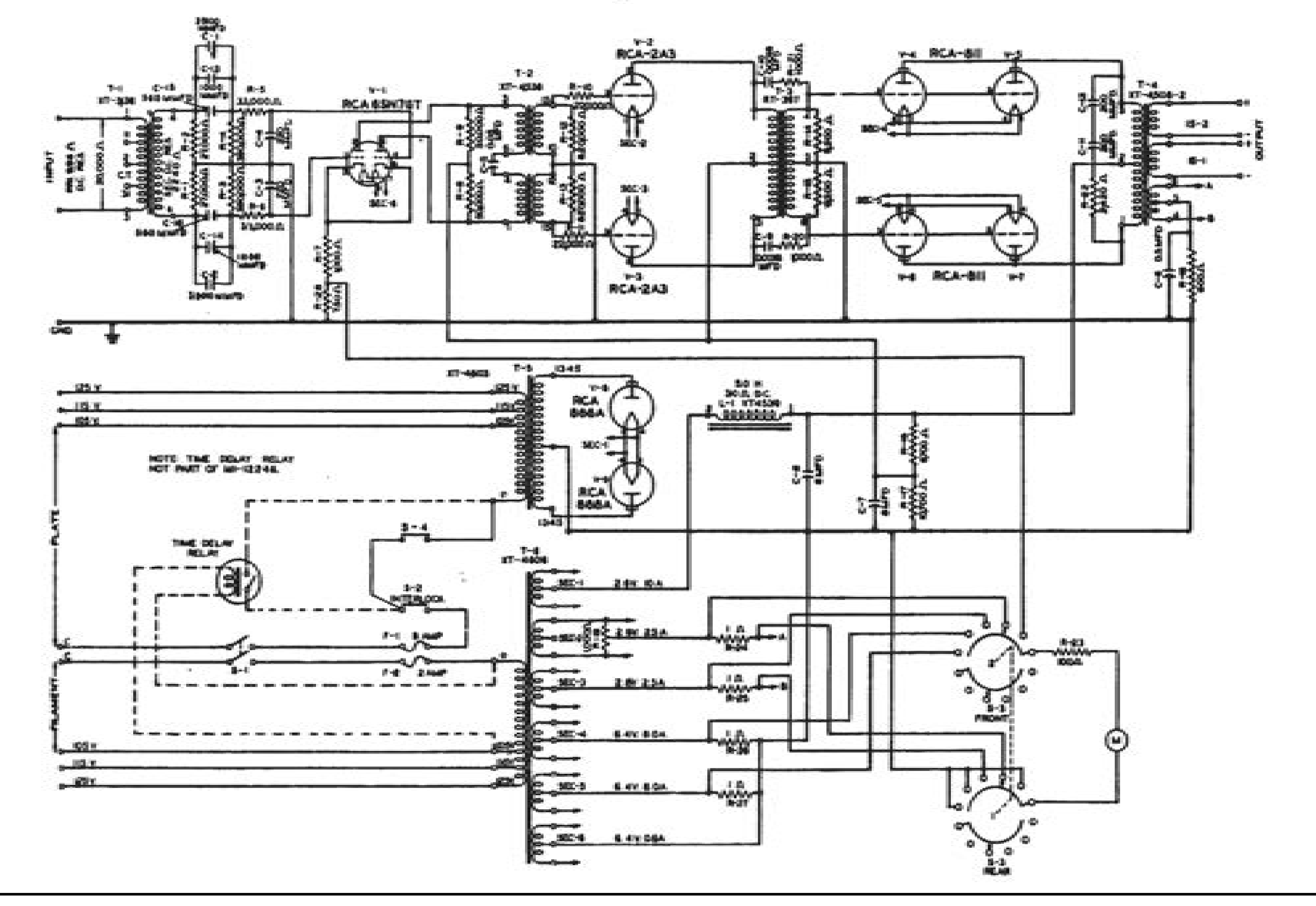


Photo 2: This 100-W DuKane amplifier dates from the early 1960s.

whose center tap was grounded through two 50-kΩ resistors in series, placing the control grids effectively at DC ground. The datasheet rates the 300A at a maximum power output of 20 W, using a fixed control-grid bias of −67.5V (350-V B+), in a Class A1 push-pull configuration. The more negative grid-bias voltage used in the 92A would seem to indicate Class AB_2 operation, likely with an output power closer to 40 or 50 W.

RCA's Early Days

RCA was founded in 1919, and major stockholders included General Electric and Westinghouse. In 1926 and 1927, RCA demonstrated its Photophone variable-area sound-on-film system, which directly competed with Vitaphone. Early talkies often included a credit line identifying the technology used for sound: either Western Electric's or RCA's.

As with the WE 92A, the RCA mi12246 amplifier was designed for cinema use, and was an all-triode design. Its input was transformer-coupled to a voltage amplifier consisting of a 6SN7GT octal dual triode connected as a differential triode pair driving a pre-driver transformer (see **Figure 2**).

This transformer fed a pair of 2A3 power triodes, also driven differentially, providing additional voltage amplification. The use of differential drive canceled out even-order harmonic distortion in much the same way that a push-pull output circuit does. The 2A3s were used to drive the phase-

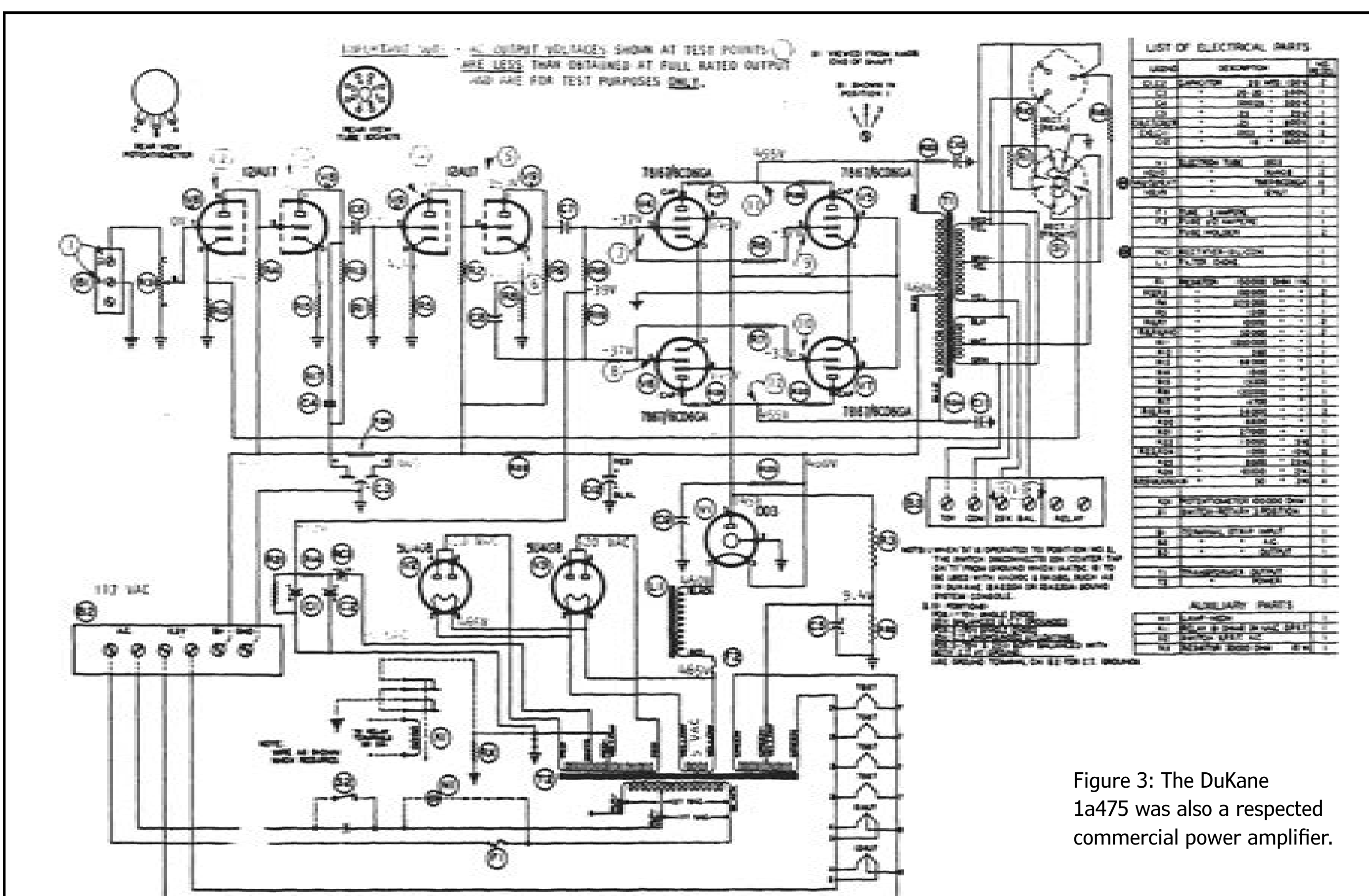


Figure 3: The DuKane 1a475 was also a respected commercial power amplifier.

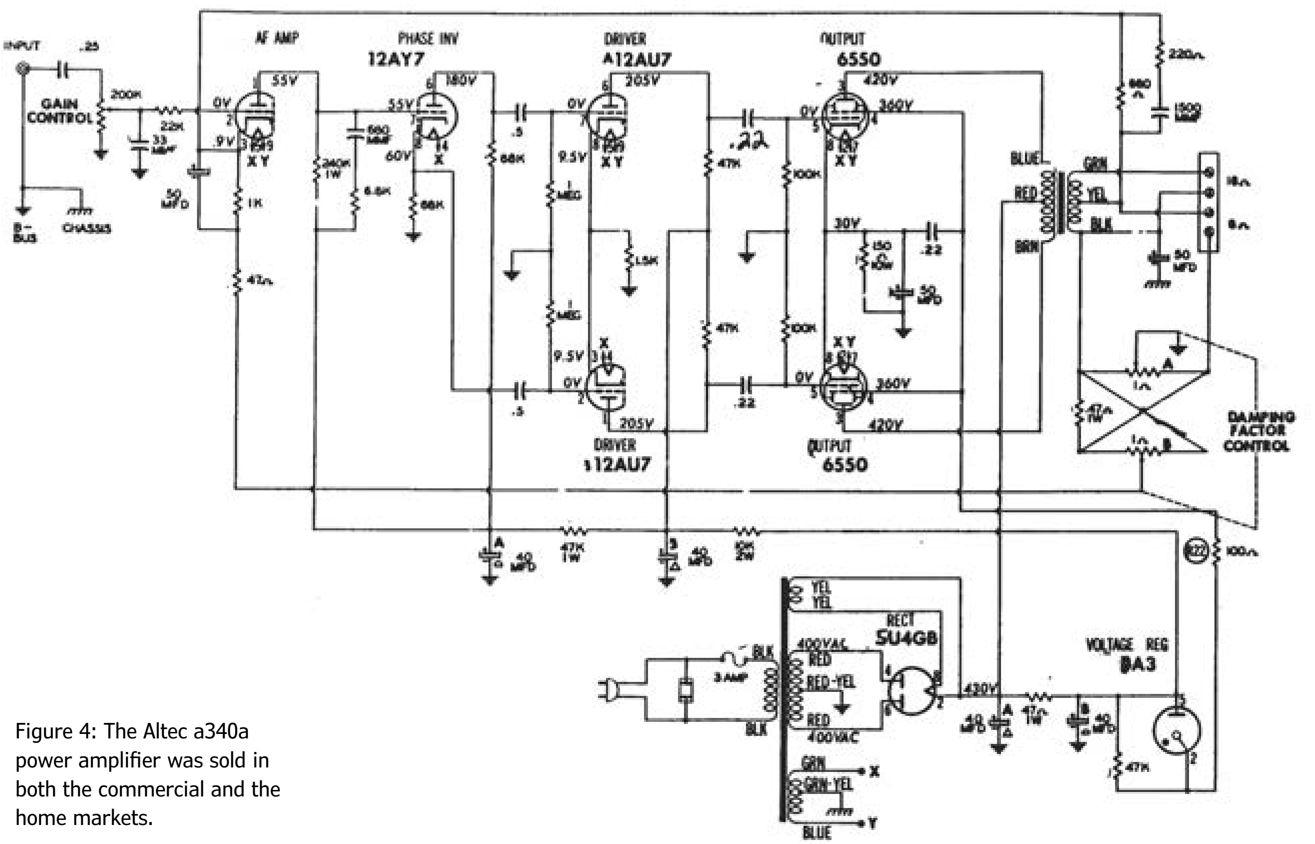


Figure 4: The Altec a340a power amplifier was sold in both the commercial and the home markets.

splitter transformer, which fed a signal to the two paralleled push-pull pairs of 811 power triodes. The 811 triode was mainly used for radio frequency (RF) power amplification. It had a maximum plate voltage rating of 1,250 V and a plate dissipation of 65 W. A push-pull pair could produce up to 340 W, implying that the mi12246 could put out 680 W, if the output transformer could handle it. However, the manufacturer's power rating was a more modest 250 W. B+ was supplied via two 866A vacuum diodes feeding an inductive-input LC filter. A resistor-capacitor (RC) network was connected from primary to secondary of the final driver transformer to prevent RF oscillation.

DuKane

Another prestigious manufacturer of professional audio equipment is DuKane. Founded as Operadio by J. McWilliams Stone in 1922, the company initially manufactured self-contained radio sets. (Most early radios consisted of one cabinet containing the electronics and a separate speaker cabinet.)

In 1925, the line was expanded to include commercial sound systems, control panels, and intercoms, all branded with the DuKane name. In 1927, Operadio decided it could no longer compete with radio manufacturing giants such as Atwater Kent, and discontinued radio production. The company has gone through some changes since then, including splitting its Commercial Systems Division (CSD) from the AV Division in 1954, and selling the CSD in 2002. DuKane still has an AV division and a separate division producing plastic welding systems.

In 1950, a DuKane sound system was installed at Comiskey Park in Chicago, IL, and through the 1960s, DuKane was an important commercial sound equipment manufacturer. The DuKane model 1a475 100-W power amplifier (see **Photo 2**) uses the two triodes of a 12AU7 as a 2-stage voltage amplifier. A second 12AU7 serves as a split-load phase splitter, driving two paralleled push-pull pairs of 7867 beam power tubes. **Figure 3** shows the 1a475's schematic.

An unusual feature of this amplifier is the use of an OD3 voltage-regulator tube to stabilize the output stage's screen-grid supply. Two 5U4GB full-wave vacuum rectifiers supply the B+. In the OD3,

Photo 3: The Altec a340a is still extremely popular and any specimen in good condition sells for thousands of dollars.

there is an internal jumper connected between pins 3 and 7. This jumper is used to break the circuit for the B+ supply in the event that the OD3 is not connected in its socket, preventing risk of circuit damage and possible fire.

The 7867 datasheet indicates that under these operating conditions, a push-pull pair should produce a maximum output of 65 W. Thus, two push-pull pairs paralleled should be capable of 130 W. Although the 1a475 was advertised as a 100-W amplifier, that specification was followed by "(75 RMS)." One would assume that (1) DuKane chose to rate the product very conservatively, (2) they rated the maximum power at a lower distortion level than Tung Sol specified when they produced its data sheet, or (3) the output transformer limited the output power to a lower value.

Altec Lansing

Altec Lansing is a more recent entry into the ranks of commercial power amplifiers. In 1936, Western Electric divested its audio equipment manufacturing and sales division in response to an antitrust suit. Its "All Technical Services" division was acquired and renamed "Altec Service Company," the "Altec" being a shortened version of "all technical." Altec's company executives promised they would never make or sell audio equipment.

The Altec Service Company purchased the Lansing Manufacturing Company and formed the Altec Lansing Corp. on May 1, 1941. The company has had several owners and product lines since then, but in the 1950s, it produced both commercial and high-end home music speakers and electronics.

The a340a power amplifier dates from that time (see **Figure 4** for its schematic). This amplifier had an advertised output power of 35 W at 0.5% distortion, and a frequency response of 5 to 100,000 Hz within 0.5 dB.

The a340a employed one triode of a 12AY7 as a voltage amplifier, and the other triode as a split-load phase splitter (see **Photo 3**). A 12AU7 serves as a differentially configured driver stage feeding a push-pull pair of 6550 beam power tubes. A 5U4GB vacuum rectifier and a capacitor filter supply the B+. An OA3 voltage-regulator tube is used in an unusual way to stabilize the screen-grid voltage.

In the DuKane circuit shown in **Figure 3**, the OD3 was simply connected as a shunt regulator, but in the a340a, the OA3 is connected so as to provide a stable 70-V drop below the B+ supply. So, rather than directly controlling the screen-grid voltage, the regulator tube controls the difference between the plate and screen voltages. Considering the loose voltage regulation to be expected from a capacitively filtered B+ supply, this approach probably provides superior performance across a wide range of output powers. (Interestingly, the OA3 datasheet specifies a 75-V drop across the tube during operation, but the a340a schematic only indicates a 70-V drop between the 430-V B+ at the filter capacitor's positive plate and the screen grids.)

A push-pull pair of 6550s operated at a 600-V plate voltage can provide a 100-W output if fixed bias is used. Instead, Altec chose to use a 420-V B+ and cathode bias. Under these conditions, the 6550s can output 41 W, so Altec's claim of 35 W seems conservative. Indeed, the 6550s in the a340a have been described as "loafing along."

You can easily see that the multi-thousand-dollar prices that pre-World War II commercial power amplifiers now command, especially in Asian countries, are not simply driven by nostalgia. The performance of some of these units merits a premium price!

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

May 14–17, 2015 **High End Munich 2015**

Munich Order Center (MOC), Lilienthalallee 40, Munich, Germany

www.highendsociety.de

The largest and most important show for high-quality consumer electronic will take place at the MOC in Munich, Germany, with over 400 exhibitors presenting more than 900 brands for a complete overview of the audio market. The world's largest high-end exhibition showcases network-capable audio systems, modern audio streaming solutions, digital wireless systems, multimedia stations, and legendary hi-fi units from all corners of the globe. There will also be a unique range of traditional and modern equipment on display. Promoted by the High End Society, this show is the place to meet for all trade companies. It is also a public fair and boasts a long tradition of more than 30 years.

Booth spaces at the High End 2015 are fully booked with hundreds of renowned domestic and international exhibitors. The 2015 High End Show will be significantly larger, which is one reason why the organizers decided to open another hall, now occupying 27,610 m^2. In addition to Halls 2, 3. and 4, Hall 1 will now be used as a presentation and an exhibition area. So, there will now be four exhibition halls, and two bright atria with open market places, seats, catering services and relaxation zones. The first exhibition day (which is a public holiday) is exclusively reserved for visitors from all over the world who have been accredited as traders. The High End Show is also an end-user exhibition and enjoys huge public enthusiasm from year to year. From Friday to Sunday, anyone interested in music and technology can experience the entire range of the audio industry at the MOC.

May 18–22, 2015 **Acoustical Society of America**

Wyndham Grand Pittsburgh Downtown Hotel, Pittsburg, PA, US

www.acousticalsociety.org

The 169th Meeting of the Acoustical Society of America will be held Monday through Friday May 18–22 2015 at the Wyndham Grand Pittsburgh Downtown Hotel, where a block of rooms has been reserved. With a long traditional of high-level presentations and courses, the 2015 agenda is already online and includes a wealth of topics for those interested in the acoustic science, including special sessions for students. Subjects include "Time Series Analysis and Adaptive Signal Processing in Acoustics;" "Auditory Perception of Moving Sounds by Moving Listeners;" "Psychological Acoustics;" "Acoustics Measurements;" "Psychoacoustics;" "Structural Acoustics and Vibration;" "Telecom and Audio Signal Processing;" "Noise, Soundscape, and Health;" and "Acoustic Metamaterials."

May 29–31, 2015 **The Home Entertainment (T.H.E.) Show Newport**

The Hotel Irvine, Irvine, CA, US

http://theshownewport.com

The Home Entertainment (T.H.E.) Show has announced sweeping changes for T.H.E. Show Newport Beach. T.H.E. Show Newport Beach will now be held at The Hotel Irvine in Irvine, CA. T.H.E. Show Las Vegas, usually scheduled for January will be put on hiatus after 16 years. The promoters explain "all of T.H.E. Show's resources and time will be spent making 2015 a game-changer and spectacular event."

Visitors to T.H.E. Show Newport Beach can visit the Headphonium exhibitors, the Marketplace, Vinyl, CD, and Accessories vendors all in one location—a giant 15,000-ft^2 Grand Ballroom in the center of the Hotel Irvine. Returning to the event in 2015 will be the gourmet food trucks, a giant high-end auto show, live entertainment day and night, wine tastings, and many popular educational seminars orchestrated by the Los Angeles and Orange County Audio Society (LAOCAS).

May 31–June 5, 2015 **Euronoise 2015**

MECC Congress Center, Maastricht, The Netherlands

www.euronoise2015.eu | Euroacoustics.org

Euronoise 2015, the 10th European Congress and Exposition on Noise Control Engineering, will be held in Maastricht, in the heart of Europe where the first treaties leading to the creation of the European Union were signed. Acousticians and noise experts from all over the Europe will gather for the event on noise control, and soundscape in Europe, organized by the European Acoustics Association (EAA).

Promoted this year by the Belgian and Dutch acoustical societies, ABAV and NAG, the Euronoise 2015 conference comprises keynote lectures, tutorials, parallel technical sessions, an exhibition, and short courses. The program focuses on environmental noise control, acoustics, and improvement of people's quality of living. The organization confirms it has received more than 600 abstracts.

Euronoise 2015 will host an extended European Acoustics Exhibition, addressing all European providers of acoustic products, services, and information. In this exhibition, companies, manufacturers, engineering service providers, research and administrative units, technical and scientific organizations, and publishers will discuss the newest results in research and development, about equipment and instrumentation, methods and software solutions and materials, standards, guidelines, and publications.

June 13–19, 2015 **InfoComm 2015**

Orange County Convention Center, Orlando, FL, US

www.infocommshow.org

InfoComm International is the trade association representing the professional audiovisual (AV) and information communications industries worldwide. InfoComm 2015, the largest audiovisual show in North America, will be held in Orlando, FL, with the convention taking place from June 13–10, and the exhibits open from June 17–19. InfoComm 2015 has more than 500,000 net ft^2 of floor exhibits and special events space. More than 37,000 professionals are expected to attend the show, with more than a third of attendees coming from technology managers, specifiers, and end-user communities.

More than 980 companies are confirmed exhibitors. In addition to the show floor and hundreds of InfoComm University sessions, Manufacturers' Training, and specialized conferences, InfoComm has numerous networking events. The 2015 opening session at the Chapin Theater, on June 16, will feature a panel of industry thought leaders on The Internet of Everything, moderated by Nick Bilton, Lead Technology Writer/Reporter for *The New York Times*. The traditional Awards Dinner is also confirmed for the same day, recognizing honorees of Adele De Berri Pioneers of AV Award, Distinguished Achievement Award, Educator of the Year Award, Sustainable Technology Award, Women in AV Award and Young AV Professionals Award.