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By João Martins

$9.00US $10.00CAN

FEBRUARY 2015

February 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 300, East Hartford, CT 06108, US Periodical Postage paid at East Hartford, CT, and additional offices.

Head Office:
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111 Founders Plaza, Suite 300
East Hartford, CT 06108, US
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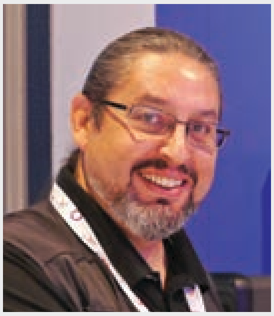

Crowdfunded Innovation

It appears that 2015 will be an exciting year for the whole audio industry with so many technological advances contributing to strong market growth globally. From mobile audio to wireless speakers and headphones, from immersive experiences to high-resolution audio, there's never been so much excitement for new technologies and integration.

Currently driving innovation is the fact that corporate investments in R&D have been reinforced by new technologies and standards supported by industry consortia. Also, there is increasing awareness toward disruptive innovations. The pace of innovation was accelerated by the ability of consumers to directly "sponsor" product innovations and new designs. Crowdfunding platforms are helping promote new concepts that companies would normally never consider promoting for themselves (for multiple reasons).

Kickstarter and Indiegogo alone, among hundreds of similar crowdfunding platforms, have helped secure nearly $500 million pledged to more than 4,000 successful technology projects. Of those 4,000 successful crowdfunded projects approximately 35% are audio related.

In the middle of computing contraptions that neither Apple nor Samsung would consider showing in public, smart motorcycle helmets, family robots and home security devices, we also find new fitness concepts and gadgets using biometric and environmental sensors, several 3-D printers and 3-D scanners and, of course, lots of wireless speakers and "next-generation" music players. Among the completely anecdotal projects, there are truly remarkable concepts such as the Olive ONE "All-in-One Home Music Player" with its gorgeous touchscreen interface and Geek Audio's high-resolution audio systems—including the Geek Wave, a portable high performance music player with a large, interchangeable battery, and 32 bit/384-kHz pulse code modulation (PCM) plus Direct-Stream Digital (DSD) support. Originally, Geek Audio had success with a simple digital-to-analog converter and headphone amplifier, crowd-designed and crowd-funded on Indiegogo. Now it seems ready to repeat the formula with its new Geek Source high-resolution music player with FLAC, DSD, DXD, and 384K support.

In contrast, some companies with reasonable "evolutionary" products have tried the crowdfunding formula with little success, mostly because they failed to excite. Apparently, the formula works when you promise something completely new, such as the "levitating Bluetooth speaker" (Mars by crazybaby), or announce "sound in space—similar to a hologram" as Canadian technology start-up Mass Fidelity did with its amazingly successful "The Core" Wireless Speaker system. Mass Fidelity's Indiegogo campaign for the "Core" Wireless Speaker System quickly became the top crowdfunded Canadian company, reaching $1,142,079 of its $48,000 goal (at the time of writing).

But, most surprising are products such as the Axent Wear Cat Ear Headphones, described as "the latest fusion of fashion and functionality with external cat ear speakers and LED lights," created by two UC Berkeley alumni, who have managed to raise $2,930,631 from their $250,000 original goal. Would Sony or any other manufacturer even consider such a concept?

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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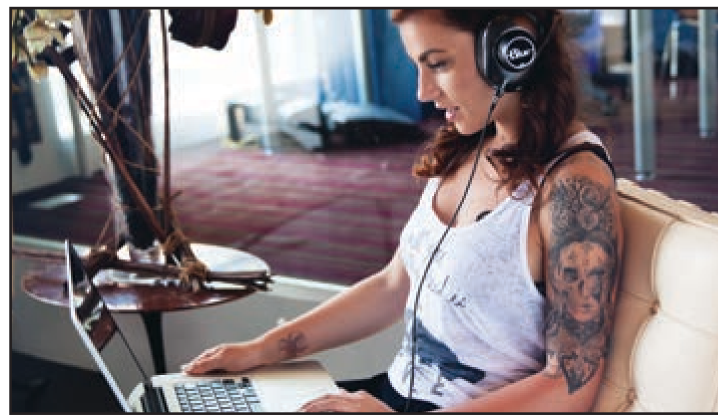

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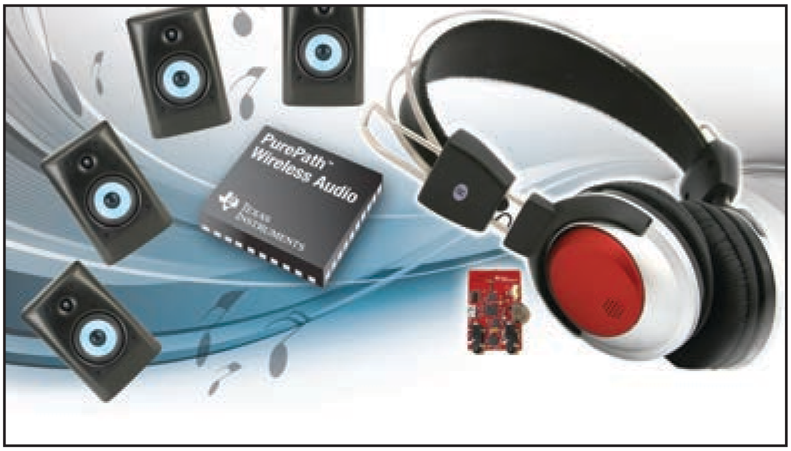

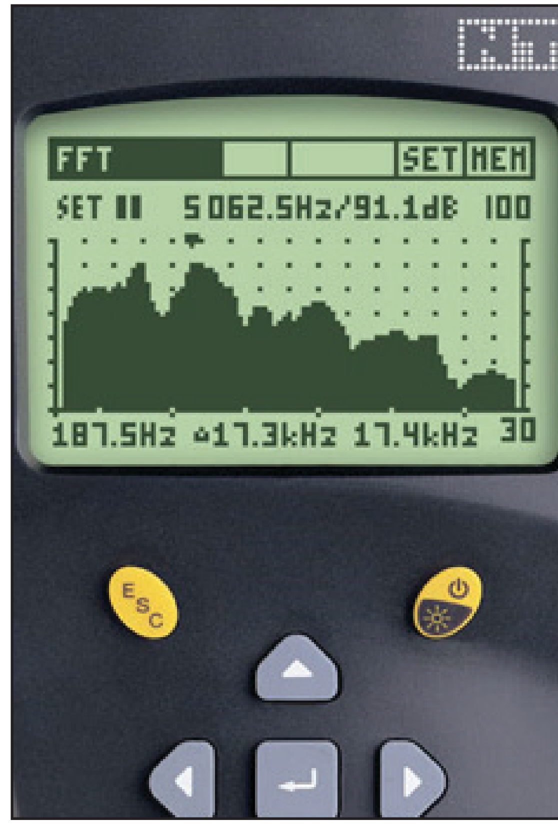

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Wireless Audio (Part 5)

Approaches to Wireless Audio Implementation

Previous articles in this series have discussed different technologies and provided examples of development platforms currently available. In this article, we are going to examine wireless audio from the manufacturers' implementation perspective, discussing some available solutions. We will also mention a few significant examples.

By
João Martins
(Editor-in-Chief)

Every speaker and audio component manufacturer needs to decide how to implement wireless audio into its systems, whether the project involves a simple portable Bluetooth speaker or a full series of networked whole-home audio products.

For every type of project, there's a choice of development platforms and basic modules from which to choose. However, the problem seems to be more how to select the best technology, depending on time to market, marketing promises, and market goals. In the last three years, many manufacturers have started projects from scratch, promised the world, and later discovered their designs had been superseded by some new, superior, and updated technology.

The Moos Audio Mini Aero speaker promised to achieve reliable wireless transmission of 100% lossless, high-resolution digital audio content at up to, and including, 24 bit/96 kHz.

So, it becomes increasingly important to understand how fast the companies supplying the development platforms are updating their offerings and how closely they will need to work with the implementers to ensure that their designs are up-to-date and upgradable. Even when the product design plan might include working with the best available suppliers, sometimes the challenges are simply too large for the project's scope.

When we started working on this wireless audio article series, we were impressed by the Mini Aero wireless loudspeaker from a company called Moos Audio, promising "the first-ever high-resolution, wireless loudspeakers with lossless, bit-accurate playback of high definition digital audio."

This Australian company started developing a completely new range of wireless speakers in 2011. The Mini Aero's mechanical and acoustic design was developed in Sydney. The company also partnered with the world's leading specialists in transducer design and manufacturing, analog and digital electronics design, signal processing, and amplification, including Scan-Speak A/S—which provided the drivers and assembled the speakers in Denmark—Danville Signal (US), Hypex (The Netherlands), and DSP Concepts (US).

Moos Audio's proprietary technology—Wireless Audio Transport Technology (WATT)—was described as "the first to achieve reliable transmission of 100% lossless, bit-accurate, high-resolution digital audio content at up to, and including, 24 bit/96 kHz." The Mini Aero was supposed to support playback of 24-bit/192-kHz content via ultra-high-performance asynchronous sample rate converters and achieve 24-bit/96-kHz wireless playback. Exactly what every demanding consumer dreams of.

Moos Audio introduced the Mini Aero at the 2013 International Consumer Electronics Show (CES) and even received an International CES Innovations 2013 Design and Engineering Award in the High-Performance Audio category.

Unfortunately, the scheduled delivery for the Mini Aero never happened, and at this point we don't know if it will ever make it to market. However, we know one important detail. Danville Signal, the company providing the wireless audio implementation for Moos Audio, became a member of the Wireless Speaker & Audio Association (WiSA) and started working on its own Super Harvard Architecture Single-Chip Computer (SHARC) DSP-based wireless audio implementations of the WiSA technology. We sincerely wish Moos Audio would deliver a WiSA-compatible solution, and we look forward to seeing the final product in stores.

A Simple, Homemade Approach

We think the Moos Audio example serves to demonstrate the importance of knowing the available choices in terms of technologies, marketing timing, and above all, choosing the right partnerships in this competitive industry.

Of course, some companies might try to make it on their own, especially if the purpose is to go beyond standard Bluetooth transmission and focus on simple high-quality audio transmission without depending on, or interfering with, Wi-Fi networks.

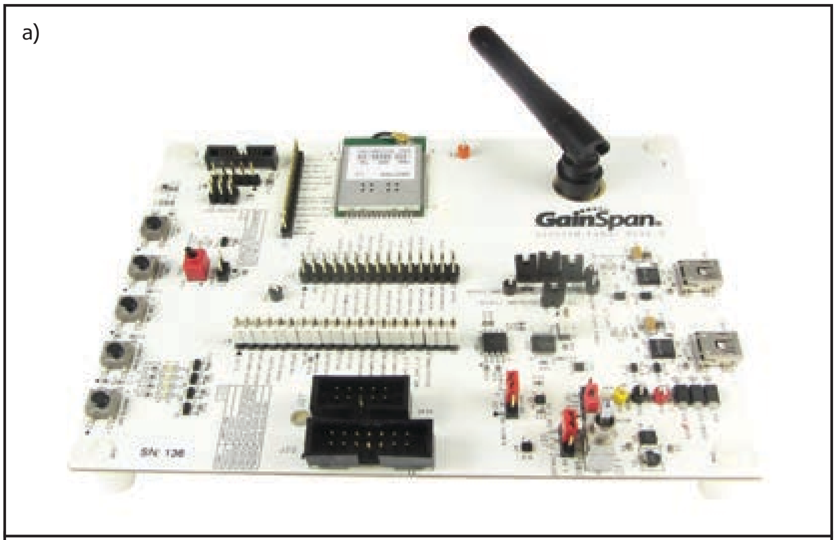
Audioengine is one company with available products that wirelessly stream 24-bit audio. The Audioengine D2 Premium 24-bit Wireless DAC solution was one of the first on the market to surpass the uncompressed 16-bit/44.1-kHz barrier in wireless audio, using dedicated hardware interfaces. Audioengine's D2 Premium wireless devices combine a DAC (Burr-Brown PCM1792), a USB to S/PDIF converter, a S/PDIF stereo link, and wireless transmission of 24-bit/96-kHz audio, independently of the source.

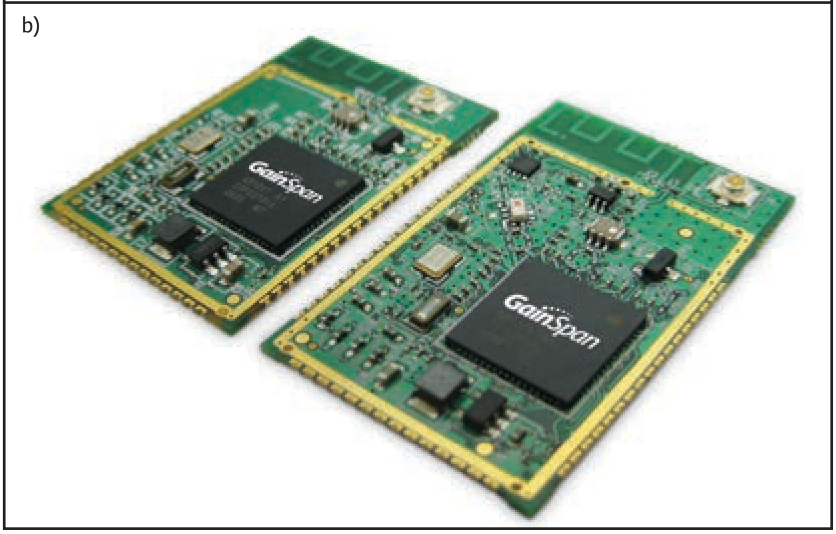
This means the D2 processes digital audio at any bit depth up to 24 bits and sample rates to 192 kHz, then sends music at 24 bit/96 kHz. It accepts USB or pulse code modulation (PCM) stereo optical and will automatically configure to match either of these inputs. The solution transmits all music file formats from any computer media player and can simultaneously transmit HD audio to up to three wireless receivers within 100′ (30 m) typical range, with low latency and resistance to interference and dropouts.

The system's proprietary radio frequency (RF) technology ensures data integrity with no impact on existing routers or networks and the volume control on the D2 sender is transmitted over an entirely separate channel so there is no impact on the digital audio stream. As Audioengine explains, the D2 system divides the band between 2,405 MHz and 2,477 MHz into 37 discrete, 2-MHz wide

Audioengine's D2 Premium 24-bit Wireless DAC solution was one of the first on the market to surpass the uncompressed 16-bit/44.1-kHz barrier in wireless audio, using dedicated hardware interfaces.

Audioengine's D2 Premium wireless devices process digital audio at any bit depth up to 24 bits and sample rates to 192 kHz, and send music over-the-air at 24 bits/96 kHz.





GainSpan's GS2000M Wi-Fi Modules and Evaluation Boards are Wi-Fi Alliance certified and provide a quick, easy, and cost-effective way for developers to add wireless audio to new product designs. GainSpan recently released the Music ADK with support for network streaming protocols.

channels. Channels numbered 2 through 38 inclusive are used for system operation. The system scans the spectrum and selects two channels that are 18 channels (or 36 MHz) apart and transmits 50% of the time on one channel (e.g., channel 2) and 50% on the other channel (e.g., channel 20). The system stays on these selected channels until the error detection rate reaches a predetermined level, indicating deteriorating RF conditions. The system will then select a cleaner channel for transmission and move there without any drop in audio. In this way, the D2 not only maintains its own audio integrity, but co-exists nicely with other local area network (LAN) devices.

Limitations of such a system, mean the Audioengine D2 does only point-to-point transmission since it uses a dedicated transmitter and a receiver. It's possible to connect multiple receivers for point to multipoint—up to three receivers—but they will all play the same signal.

However, Audioengine D2 devices work as advertised, bringing high-quality wireless audio to the market as a closed wireless link with almost universal compatibility with any source and direct computer USB interface for Windows, Mac OS X, and Linux that is completely independent of the playback application.

The Networked Devices Approach

Currently, many manufacturers want to implement simple wireless audio functionalities in their new product designs. These manufacturers seek the widest possible compatibility with existing networked and mobile devices—with direct compatibility to existing solutions widely available in the market. Those companies prefer to evolve along available solutions.

For such scenarios, there's a huge range of available platforms and reference designs from which to choose (e.g., those from GainSpan Corp., which is a spinoff of Intel Corp. and an innovator in semiconductor solutions for wireless connectivity).

Focusing mainly on ultra-low-power Wi-Fi for the Internet of Things (IOT) and industrial market applications, the company is also investing in the current wireless audio revolution and recently rolled out its new Music Application Development Kit (ADK) for streaming high-quality music over Wi-Fi.

As home connectivity increases, GainSpan understands that consumers will be seeking wireless solutions exceeding the basic music streaming capabilities offered via Bluetooth. Growing use of music streaming services such as Spotify are driving requirements for direct-Internet connectivity in wireless music products. Simultaneously, consumers are also demanding hi-fi music (24-bit audio and lossless codecs) and the ability to place wireless speakers anywhere.

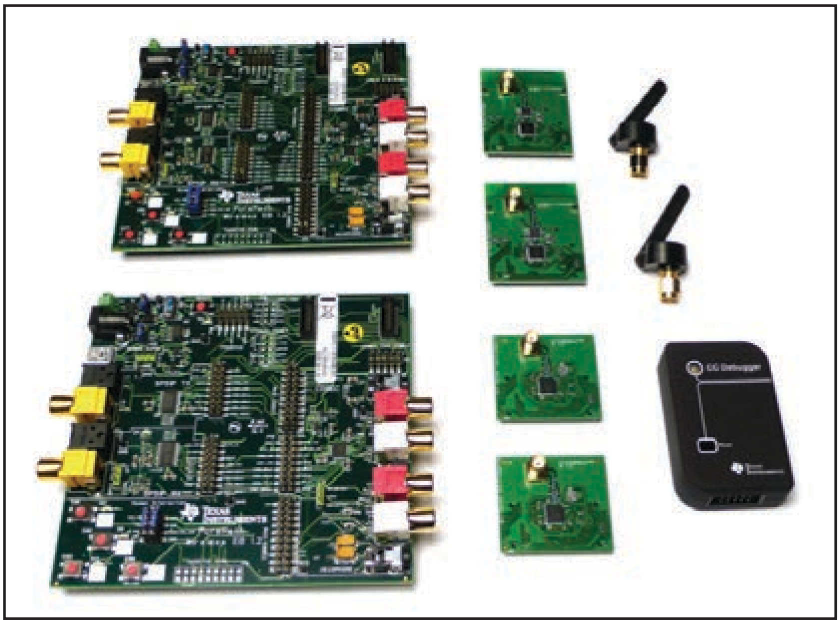
GainSpan believes that standard Wi-Fi, with its longer range and increasingly higher throughput will be the preferred route to satisfy these demands, and the company is developing solutions that target multi-room and multi-speaker capabilities.

GainSpan's Music ADK is a complete hardware and software reference design that enables OEMs and ODMs to design for the new requirements in wireless music streaming. Based on its GS2011MIZ Wi-Fi module, GainSpan recently released the Music ADK with support for network streaming protocols enabling media management, discovery of content, and control, based on Universal Plug and Play

Texas Instruments' PurePath Wireless technology enables development of audio applications with multichannel and multipoint audio streaming capabilities for consumer applications with USB support.

(UPnP). Using standard apps such as Bubble UPnP or similar DLNA apps from Play Store or AppStore, the Music ADK enables streaming from a networked device to a wireless speaker.

GainSpan's reference design also leverages the GS2000's high data rates to deliver high-definition audio (24-bit audio, up to 192-kHz sampling rate) and lossless codecs such as free lossless audio codec (FLAC) and multichannel codecs such as Dolby 5.1.



The CC85XXDK development kits enable evaluation of Texas Instruments' PurePath Wireless technology for wireless audio applications and work with the PurePath Wireless Configurator PC tool, which helps designers configure and test PurePath Wireless audio devices.

Using a proven audio DSP from a leading vendor with support for major codecs (MP3, AAC-LC, ALAC, FLAC, Dolby, and DTS) the solution also offers advanced post-processing/preprocessing functions such as surround sound, enhanced bass, and compressed music restoration.

The hardware platform includes a variety of interfaces such as SPI, I^{2}S, and flexible connectivity to a 24-bit DAC and an amplifier, enabling connectivity to various audio DSPs. It also offers a robust roadmap for integration of new features such as multi-zone speakers and cloud connectivity.

From ICs to SoCs

Complementary or alternative solutions for wireless digital audio streaming are available from major semiconductor companies, such as Texas Instruments (TI).

TI's family of PurePath Wireless audio products are well known 2.4-GHz audio integrated circuits (ICs) that transmit uncompressed CD-quality wireless audio (16-bit 44.1/48 kHz) over a solid RF link for consumer, portable, and high-end audio applications (e.g., wireless headphones, headsets, and speakers).

TI's CC8520 (two-channel) and CC8530 (four-channel) system-on-chips (SoC) are complete integrated solutions with RF protocol, a microcontroller, an audio codec setup support, application designs, and a free PurePath Configurator PC software tool. Additionally, the CC8521 and the CC8531 Simplelink Wireless MCUs contain USB audio support for all major operating systems.

Resources

Audioengine, http://audioengineusa.com.

Avnera, www.avnera.com.

Caskeid, www.caskeid.com.

GainSpan Corp, www.gainspan.com.

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

Summit Wireless, http://summitwireless.com.

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

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

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

Avnera system-on-chip (SoC)-based wireless audio solutions are powering some of the biggest brands' wireless HD (24-bit/96-kHz) multichannel solutions, such as the Harman Kardon Omni 20.

Caskeid is a family of technologies developed by Imagination Technologies. Caskeid is available for licensing as part of its portfolio of products for custom system-on-chip (SoC) design to deliver efficient and accurate wireless multi-room audio. Pure is one of the companies already shipping Caskeid-enabled speakers.

These solutions also have multichannel and multipoint capabilities, allowing streaming of up to four simultaneous channels to up to four receivers in coexistence with Bluetooth, WLAN, and other 2.4-GHz devices. It also supports any audio device that is configured through I^{2}C and/or up to 4 GIO pins.

At the time of writing for this article, TI still had not released any information about its new generation platforms with updated specifications for the most recent Wi-Fi standards, including the required dual and multi-band support.

In past articles, we have mentioned other solutions such as the SoC products offered by Avnera (Beaverton, OR) that combine sophisticated analog circuits for the consumer audio industry. The analog circuits are also used by major manufacturers such as Harman International, Logitech, Onkyo, Panasonic, Samsung, Skull Candy, and Sony, among others.

Avnera is a pioneer of high-quality analog SoC (ASoC) technology. The company is also successful with its SoC-based wireless audio solutions, which are anchored by a custom digital radio and communications system that delivers uncompressed, low-latency digital audio.

In previous articles, we have also mentioned Imagination Technologies' Caskeid licensable wireless audio Internet protocol (IP) solution, which is a collection of hardware and software technologies that enables synchronized wireless multi-room and multichannel audio solutions over standard domestic Wi-Fi networks.

Several manufacturers have adopted Caskeid's solutions mainly because of its low-latency synchronization—less than 25-µs delay—and audiophile-quality stereo playback or fully wireless surround sound systems. Caskeid's solutions also support direct cloud-based music and radio services. A Caskeid App Developer Kit (ADK) enables agnostic app support over Wi-Fi for any Caskeid speaker. Caskeid is also supported by an app framework for Android/iOS that provides full control and configuration for all Caskeid products. OEMs/ODMs can leverage the off-the-shelf app framework or can work with Imagination Technologies to customize an app for their brands. ax

Surround Sound—The Next Generation

Dolby Laboratories

By
Ron Tipton
(United States)

In our quest for realistic recorded sound, we explore the legacy of quadraphonic sound and the early success experienced by Dolby Laboratories.

Although Quadraphonic was not a commercial success, it paved the way for change. In the early 1980s, Dolby Laboratories introduced a four-channel analog system, Dolby SR—which is not to be confused with Dolby SR (Spectral Recording) noise reduction format.

This Dolby SR was primarily developed for the motion picture industry to add that important center channel quadraphonic was never able to successfully achieve. Most motion-picture dialog occurs front and center so a center channel is needed. The four analog channels—left, center, right, and surround—are optically encoded onto the film just as mono and later stereo had been.

Note that the "surround" channel is mono so it's the same audio regardless of the number of side or rear loudspeakers. Four microphones were used for recording: left, center, right, and surround (which was probably located at rear center). This is still a four-channel analog system (such as Quad) but the rear surround information has been diluted to provide a real center channel.

Dolby Surround

In 1982, Dolby Laboratories introduced Dolby Surround, a matrixed, two-channel version of Dolby SR, intended for home use. Dolby Surround used the following encoding matrix:

Left (total) = Left − j0.7Surround
Right (total) = Right + j0.7Surround

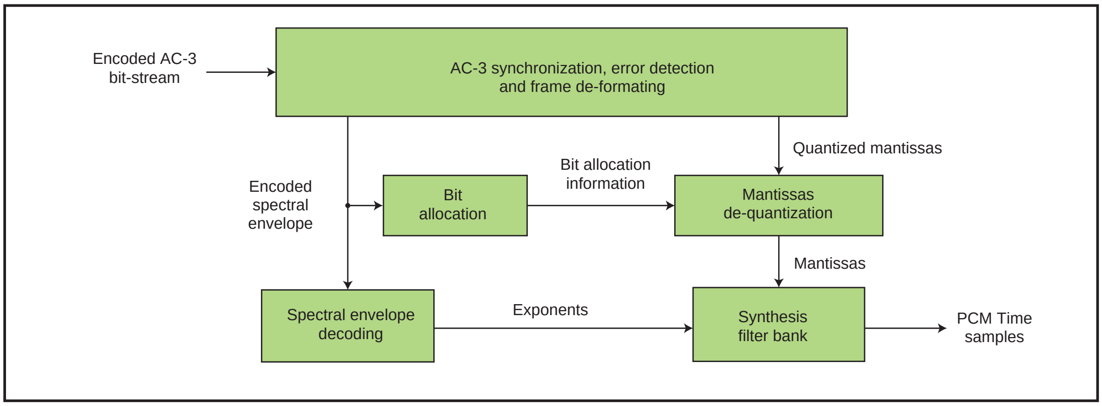
In this matrix, j is a 90° phase shift.

Surround is stereo compatible, which was important when motion pictures were released on VHS or Betamax cassettes for home use. I examined several VHS tapes in my collection and most have the Dolby Surround logo on the back of the slip case. Those folks fortunate enough to have a surround-sound speaker system got to enjoy some side or rear audio but still no real center dialog.

Dolby Pro Logic

In 1987, Dolby Labs updated Surround to Dolby Pro Logic, which finally added a center channel for

Figure 1: This is the AC-3 simplified decoding block diagram from the Advanced Television Systems Committee (ATSC) Digital Audio Compression Standard.



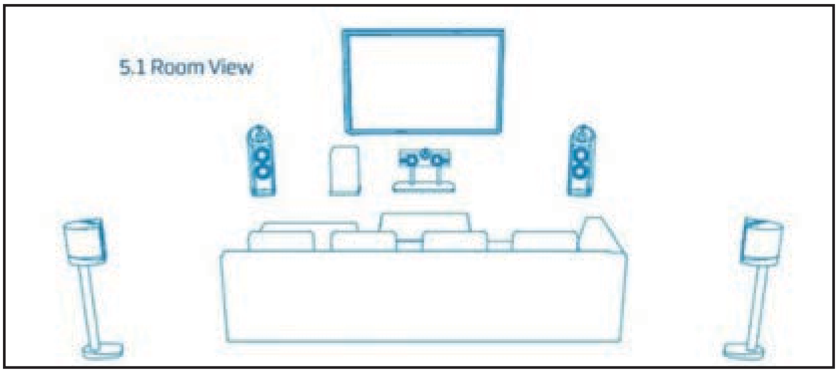


Figure 2: The 5.1 room view shows the suggested speaker placement. The rectangle under the left edge of the TV screen is the subwoofer, but its placement is not critical because the low frequencies are not directional.

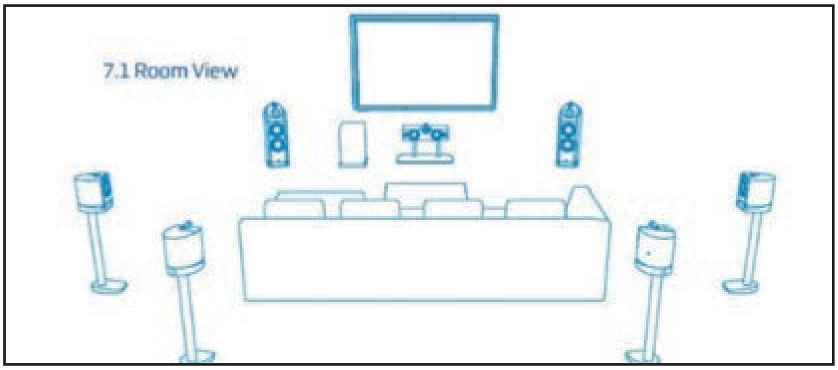


Figure 3: The 7.1 room view adds two more speakers to fill in the rear center. They can be placed closer together than shown if the room shape permits.

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.

the home market but still at the expense of a diluted Surround. The encoding matrix changed to:

Left (total) = Left + 0.7Center – j0.7Surround

Right (total) = Right + 0.7Center + j0.7Surround

The company also added the words "Pro Logic" underneath "Surround" to its logo. I examined several of my later VHS cassettes and none are Pro Logic, even the 50th anniversary release of *Gone With the Wind* from 1998.

Pro Logic was updated in 2000 to Pro Logic II with the addition of rear left and right channels replacing the mono Surround. This became a five-channel system with another microphone needed for recording, more about that later. The encoding matrix grew to:

Left (total) = Left + 0.7Center – j0.87Left Rear – j0.49Right Rear

Right (total) = Right + 0.7Center + j0.49Left Rear + j0.87Right Rear

In 2000, this was still an analog system even though the DVD had been introduced in 1995, making digital recording of motion pictures practical for the home market. There is some lingering confusion surrounding this release. Dolby Digital was introduced in 1995, so I am not sure why an inferior analog system was pursued and introduced five years later. In my research, I did not discover the answer, but I suspect marketing considerations played a large part.

Dolby Digital

Dolby Digital, also known as AC-3, is the current "work-horse" because of its many consumer products. It exists in several variations including 7.2, 9.2, and 11.2. However, the 5.1 and the 7.1 channels are the most popular configurations for home use because many AV receivers, DVD players, and computer sound cards have analog 5.1 outputs. Some also support 7.1. The bit stream encoding and decoding is complex. It is described in great detail in the 270-page Advanced Television Systems Committee (ATSC) Standard A/52-2012 (see Resources). It is a "lossy" system compared to the digital recording of stereo .wav files because a degree of compression is used.

Figure 1 shows a simplified block diagram of the AC-3 decoder. The diagram shows what occurs in your DVD player and on the multichannel sound card. The detailed decoding flow-chart documented in ATSC Standard A/52-2012, along with the text, would make it possible to write decoding software.

The 5.1 variation is an up-to-six-channel system. **Figure 2** shows one suggested speaker placement option. The "5" means there are five full-range speakers and the "1" represents the subwoofer. The 7.1 variation adds two more full-range channels. **Figure 3** shows a 7.1 speaker placement. As of this writing, it appears only Blu-ray movies have 7.1 sound tracks.

There is a freeware codec program for Windows that will encode and decode 5.1 audio. Using an input of six mono pulse code modulation (PCM) .wav files it will produce an output bit stream. Or with an input bit stream, you can recover the six .wav files. The file name is aud-x_1.24_install.zip. Written by Aud-X, this is version 1.24 (see Resources). Aud-X requires a six-channel input file but there is another 5.1 encoder that will also take a six mono file input: wavtoac3encoder. In some ways I like this one better. It seems more user friendly.

Let me digress here for a moment to show you how to work a bit of magic. In the "Quadraphonic" article (*audioXpress*, January 2015), I mentioned a free program, SQD12sfx.exe, to convert SQ to four mono files.

What if we expanded the four files into six by some judicious mixing in a digital audio workstation (DAW) or audio editor and then use the Aud-X encoder (or wavtoac3encoder) to make a 5.1 bit stream that you could listen to from your DVD player? An SQ LP changed to a Dolby Digital CD or DVD would be pretty neat! There is a step-by-step tutorial on this in the Supplementary Material section, available online at www.audioxpress.com.

Dolby Digital is a bit particular about the volume levels of the input mono files so I have included an article from the Doom9's Forum on "How to Properly Encode Dolby Digital." Another good resource is the article "Audio-Only-AC3-Using-DVDLabPro2.pdf," also included in the Supplementary Material.

Another free Windows program, AC3Filter, also decodes Dolby Digital. In addition, there are some

Photo 1: This is a Core Sound TetraMic, which is a type of quad microphone.

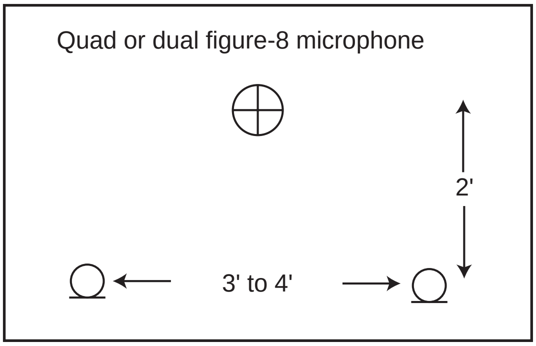


Figure 4: This diagram shows a basic microphone placement for recording 5.1 surround sound, using a quad or dual figure-eight microphone. The two rear-facing cardioid microphones are for the rear center speakers.

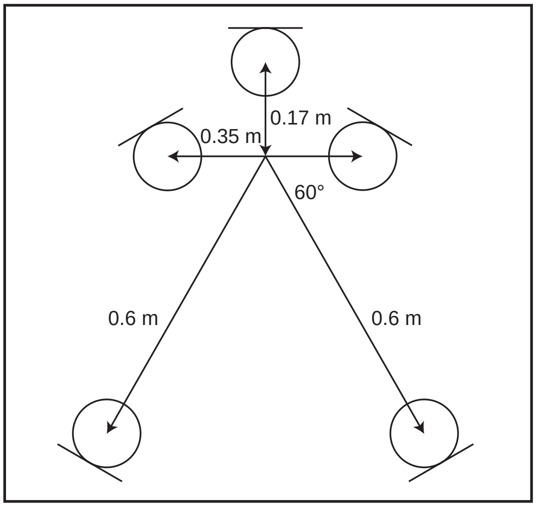


Figure 5: The Fukada Tree (diagrammed here) uses five cardioid microphones to capture 5.1 surround sound. The dimensions are in meters so it's a fairly small array.

useful plug-ins. AC3Filter Tools is a set of command-line programs for audio file manipulation, including an AC-3 encoder and decoder.

Microphone Placement

The article briefly describes these surround sound bit streams but where does the input come from? We need six mono files for 5.1 and eight for 7.1. The answer, of course, is from the microphones. However, there are different opinions as to how to arrange the microphones.

For example, to record a symphony orchestra in concert some favor multiple microphones to capture individual instrument groups but this leads to the overlapping pickups, which must then be sorted out in post processing. This may or may not result in a pleasing mix, and besides, it's rather complex in practice. Others prefer a minimum number of microphones and this probably works best overall.

A single Quad microphone (see **Photo 1**) or a pair of figure-eight microphones placed 90° to each other can work well for 5.1:

Left front = left front mic

Right front = right front mic

Center = a left and right mix often 0.5L + 0.5R or 0.7L + 0.7R to keep the RMS level equal to the left and right fronts.

Left rear = left rear mic

Right rear = right rear mic

Subwoofer = a four-channel mix low-pass filtered at about 100 Hz and amplitude adjusted for balanced sound.

If a center microphone is used, the left and right front mix isn't needed. Two more rear-facing cardioid microphones can be added for 7.1 arrangements (see **Figure 4**). These two output signals could be added to the subwoofer mix or used to just drive the rear left and right speakers as shown in **Figure 3**.

There are other microphone arrangements that have been used such as the Decca Tree (so named because it was originally used by Decca Records); the Fukada Tree (see **Figure 5**), which is actually a compact-sized Decca Tree; and the Hamasaki arrangement.

Dolby Surround Tools

As a final thought, I will mention Dolby Surround Tools, plug-ins for the Avid Pro Tools DAW. There are encoding and decoding plug-ins to make (or play) a stereo mix from four analog input files for Dolby Surround or Pro Logic. You can download the 94-page user guide from www.avid.com/US/products/Dolby-Surround-Tools.

You could also make this mix in your favorite DAW using the SQD12sfx.exe program mentioned in my "Quadraphonic" article to complete the necessary 90° phase shift and the encoding matrix. Or, you could skip the analog output and use one of the free AC-3 encoders for a digital output.

Project Files

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

Resources

AC3Filter, www.ac3filter.net.

Advanced Television Systems Committee (ATSC) Standard, A/52-2012: Digital Audio Compression (AC-3, E-AC-3), www.atsc.org/cms/standards/A52-2012(12-17).pdf.

Aud-x codec, www.softpedia.com/get/Multimedia/Audio/Audio-CD-Rippers-Encoders/aud-x-Surround-Codec.shtml.

R. Tipton, "Quadraphonic," *audioXpress*, January 2015.

wavtoac3encoder.exe, code.google.com/p/wavtoac3encoder.

SB ACOUSTICS

"building your sound"

NEW RELEASES

5" SATORI MW13P-8 preliminary

Nominal Impedance	: 8 Ω
DC-resistance, R_e	: 6.0 Ω
Resonance frequency, f_s	: 40.5 Hz
Total Q-factor, Q_{ts}	: 0.29
Equivalent volume, V_{as}	: 15.4 ltr
Piston area, S_d	: 70 cm²
Sensitivity (2.83 V/1 m)	: 87 dB
V.C. diameter	: 30.5 mm
Linear Xmax	: ± 5 mm
Rated power handling	: 40 W

- Cast aluminum chassis for optimum strength
- Proprietary cone material with EGYPTIAN PAPYRUS™
- Soft low damping rubber surround for optimum transient response
- Advanced BIMAX spider for improved linearity
- Powerful optimized low distortion neodymium motor system
- Extended copper sleeve on pole piece for reduced distortion
- Non-conducting fibre glass voice coil former for low damping
- CCAW voice coil for reduced moving mass
- Long life silver lead wires
- Vented pole piece for reduced compression

4" SB12PAC25-4

Nominal Impedance	: 4 Ω
DC-resistance, R_e	: 3.1 Ω
Resonance frequency, f_s	: 52.5 Hz
Total Q-factor, Q_{ts}	: 0.31
Equivalent volume, V_{as}	: 5.3 ltr
Piston area, S_d	: 50 cm²
Sensitivity (2.83 V/1 m)	: 87 dB
V.C. diameter	: 25.4 mm
Linear Xmax	: ± 5 mm
Rated power handling	: 30 W

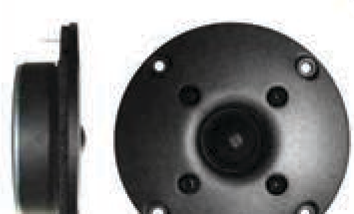
- Vented reinforced plastic chassis
- Anodised aluminium cone
- Low damping rubber surround
- Large vented motor system
- Non-conducting fibreglass coil former

2½" SB65WBAC25-4

Nominal Impedance	: 4 Ω
DC-resistance, R_e	: 3.6 Ω
Resonance frequency, f_s	: 115 Hz
Total Q-factor, Q_{ts}	: 0.68
Equivalent volume, V_{as}	: 0.43 ltr
Piston area, S_d	: 20 cm²
Sensitivity (2.83 V/1 m)	: 83.5 dB
V.C. diameter	: 25.4 mm
Linear Xmax	: ± 2.7 mm
Rated power handling	: 20 W

- Dented aluminum cone for extended usable frequency range
- Copper cap for increased high frequency output, reduced phase shift at higher frequencies and improved power handling capability.
- Low damping surround and non-conductive voice coil former to ensure dynamics and an open / transparent sound character with excellent detailing / resolution.
- Linear neodymium motor system for reduced distortion.

¾" SB19ST-C000-4



Nominal Impedance	: 4 Ω
DC-resistance, R_e	: 3.4 Ω
Resonance frequency, f_s	: 980 Hz
Piston area, S_d	: 3.8 cm²
Sensitivity (2.83 V/1 m)	: 88.5 dB
V.C. diameter	: 19.1 mm
Rated power handling	: 30 W

- Optimized damping in pole cavity to eliminate internal reflection
- Fine weave soft fabric dome for smooth frequency response
- Saturation controlled motor system for reduced distortion
- CCAW voice coil for low moving mass
- Long life silver lead wires

Marrying design talent of Danesian Audio, Denmark and Manufacturing expertise of Sinar Baja Electric, Indonesia

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- www.sbacoustics.co.id
- www.sinarbajaelectric.com
- OEM contact : mkt@sinarbajaelectric.com

www.facebook.com/sbacoustics

Fresh from the Bench

Native Instruments Molekular

A Strange and Powerful Tool

In the crowded world of plug-ins, every once in awhile something stands out due to its uniqueness and originality. Not that it should be valued more just for that and those that do not show originality should be valued less, but the software world should sometimes offer more. And, that's exactly what Native Instruments' Molekular, the brainchild of Zynaptiq CEO Denis H. Goekdag, does.

By
Fernando Rodrigues
(Portugal)

Molekular is difficult to categorize. It certainly is aimed at audio processing, but its uniqueness and complexity makes it much more than that. It's a powerful tool for sound design and sound creation, and its modularity offers an immense world of possibilities.

Besides being modular, it is also very performance oriented, with lots of controls for real-time and dynamic modulation. That part adds much to its complexity, to the point that sometimes it's hard to know exactly what is happening and why.

Molekular is described as "a semi-modular, variable routing multi-effects processor that has four simultaneous processor "slots," comprehensive modulation, macro-control options, parameter morphing, and more. The Molekular software is based on Reaktor, a modular platform that was created years ago by Native Instruments. Today, it is one of the most powerful tools for creating innovating synthesizers and other audio machines, as well as signal processing processors such as Molekular.

Reaktor was the platform for the conceptual base of several Native Instruments products (Massive is one well-known example) and it has a gigantic user library full of great examples of what can be done with it. Native Instruments has also been launching several products that reaffirm it (e.g., Razor, and more recently Monark, which is considered by many the most faithful recreation of the Minimoog currently available).

Main Concept

Molekular is a Reaktor ensemble, which means it doesn't run on its own. Those who have Reaktor are used to this and will open it within the main application, which can run as a standalone or as a plug-in in any of the usual formats via their respective hosts. Those who do not have Reaktor will need a runtime version of it, called Reaktor Player, which will be installed with Molekular. It's the same with the much-heralded Minimoog replica Monark, or Native Instruments' previous synth, Razor.

Having been created within Reaktor, it's no surprise that the entire Molekular concept is modular. In fact, many little modules interconnect beneath the surface to create all those intricate things that happen to the audio content and become audible as soon as we make it pass through the software. The DSP modules have a considerable degree of influence in those intricacies, but they are not solely responsible. The processing architecture has an immense arsenal of modulators, real-time controllers, and a powerful main screen, called the Morph screen. Many of the DSPs themselves have built-in modulators, mainly pattern sequencers.

It appears as if Molekular was designed as a kind of hybrid between a musical instrument (a complex modular synthesizer such as those we sometimes find in the Reaktor library) and an audio processor. This is even more obvious in some of the DSP modules, which can be programmed to "play" sequences or patterns. It can even be controlled by MIDI to create musical cells or small melodies.

With the myriad of processors and modulators, the audio sometimes is completely transformed and becomes a mutant alien that screams through our monitors. Other times, it's like there is someone playing something over it, and what comes out is completely different from what went in.

User Interface and Architecture

Being a Reaktor Ensemble, Molekular follows the paradigms of its host. Since we already have Reaktor installed, we used it to work with Molekular. When we open Molekular, we see Panel A, which is divided into two parts—the upper part is occupied by the real-time controls with the modulators on the left and the routing panel on the right. In the center, prominently placed, is the powerful Morphing Window. This is one of the "magical" parts of Molekular, where the modulators and the processors may be programmed to change over time, and therefore, change the way they affect the sound. Using the Morph controls, Molekular may go from subtle to crazy in a snap, or slowly and gradually change from one setting to another.

In Panel A's lower part, we have the four slots for the processors. Although we have many processors available, we can only have four active processors per patch. Some are exclusive to a single slot and others are slot independent, and can be used more than once. The slot-independent ones are the Dual Delay, the Equalizer, the Filter, the Level, and the Metaverb.

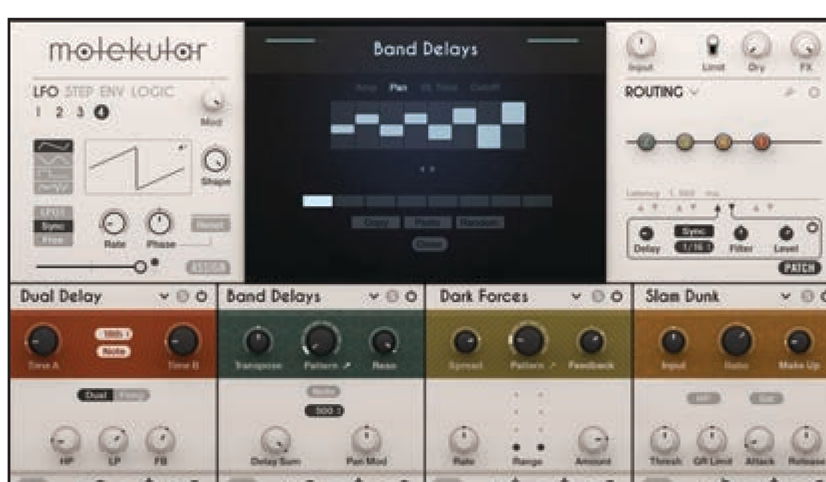
The specific processors vary in number and type. Slot 1 seems dedicated to spectral shaping and filtering and has seven processors. Slot 2 is the delay slot and it has even more processors. Many are pretty intuitive, but several of them are not exactly

Molekular is a Reaktor ensemble, and we tested it with the stand-alone version of Reaktor.

easy to access. Their names reflect that "personality" and they include: Iteratron, Reverseold, Ryuichi, and Trails. Slot 3 specializes in frequency-type processors, and provides time-modulation effects and some filters. Slot 4 is where the distortion-type processors and the "like nothing else" types can be found. It is perhaps the most radical (but pretty much any slot can be very radical in its processing).

Each processor has its own panel with parameter controls, and all of them have the usual mix, wet, and out controls in the lower part of the panel. Besides their specific controls, parameters can be modulated by MIDI and/or by Molekular internal modulators. Molekular has a generous amount of these: four low-frequency oscillators (LFOs), four step sequencers, four envelopes, and four logic processors (I will discuss more about these later).

There is also an alternative to Panel A—a Panel B—which is typical in Reaktor Ensembles. This second panel only has the Morph Window, and our guess is



The control panel's main graphic window is not just used to display and control the Morph and the Motion. It doubles as a graphic control panel for several modules and modulator components. Here, we see the control panel for the Band Delays' DSP module.

that it is performance oriented. This Morph Window is a distinctive mark of Molekular, as we will see.

To understand Molekular's architecture, we will need to have the full version of Reaktor, which enables us to open the structures, observe how things are patched, and determine what happens to the audio (kind of, since this ensemble is highly complex in many levels of the structure's encapsulation).

It is possible to change a DSP position in the Routing control panel.

Dark Forces is one of the most interesting DSPs. The control panel is shown in the Morph window.

Since we have so many DSPs to choose from and only four slots, we need to select what we want in the central window. Since each slot has its own specialized DSPs, only those and the "Bread and Butter" ones are shown for each selected slot.

When we open the first level, we notice two main blocks, named "devices" and "flow." One of the blocks is dedicated to processing, and the other is dedicated to routings (but we have connection routings everywhere). Double clicking in the "devices" block, we have another panel (we are just in the second level) with several new blocks that are neatly placed and clearly labeled. We immediately noticed the first one, called "Audio." This controls the signal processing. Another double click takes us to the third level. Here the number of blocks greatly increases, but there are several immediately recognizable names: Slot0, Slot1, Slot2, and Slot3. There are also several other modules. And this is where the core cells are located. Basically, each of the blocks already have core cells. Some of them are just routers and selectors. When we double click each of the processors (inside their respective slot-block containers), we find core cells with additional core cells inside. We had to stop there, because the code is not accessible.

Anyway, the inner workings are so complex that we won't go into any more of it here. However, all the ensemble's parts are accessible for those who have the full version of Reaktor and are brave enough to try to find their way through it.

Modulations And Morphing

Modulation is one of Molekular's key words. Most of its parameters can be modulated (and we can even morph between modulations, in real time, manually, or automated). The modulators control panels are located in the control panel's upper-left section. This is where we assign and record modulations, as well as adjust each modulation source's parameters. Molekular features 16 modulation sources divided into four categories—LFOs, step sequencers, envelopes, and logic processing units. There are four of each of the first two, three Envelopes (the fourth is actually an envelope follower) and although we can only use four Logic units, things are much more complex, because these units perform some math operations using two modulation inputs as operators.

Besides all the complexity that these units already carry on their own, we can use them in any combination, or even all at once. We can even modulate other modulators, a characteristic commonly known as cross-modulation, which is another powerful Molekular feature (which it owes to its modular concept).

At the center of the Molekular graphic panel interface is the black square with the powerful Morph Window. The square (the morphing field) usually displays a moving point that leaves a trace representative of the movement (something very common in Reaktor ensembles). Of course, the point is not always moving, and the kind of movement and its speed varies a lot from a preset to another. When the point moves automatically, it's because it is being modulated, and relates to the kind of modulations and the paths traced from the base position to the offsets. Angle and Radius controls have an important role in this definition, and the morphing speed is controlled with the Morph Quantization, accessible by clicking in the little dotted circle at the upper right corner of the right panel.

Molekular is a signal processor, so we tried it as a VST inside Cubase, processing several kinds of sounds, from rhythmic to melodic and from synth sequences to pads.

DSP Modules

Molekular comes with a collection of single-DSP example patches, which demonstrate the capabilities of the individual algorithms. It also comes with an empty "init" patch. These patches have just one DSP slot, use one algorithm, and have some modulations in place. However, that's enough to show things can go really far by just using a single processor coupled with some modulations.

Some processors are subtle, and work more or less as expected. The generic ones fall in this category. The Dual Delay works as expected, except that the modulations can make it go wild sometimes. The Equalizer has smooth filters that respond well to sweep through modulation. The Filter fell into the smooth category, and even with drastic values, didn't dip much into the wild side. Metaverb is a good reverb if we consider it's primary purpose—coupling with other modules to create lush atmospheres. Lush would actually be the word we would choose to characterize it, not in the usual,

The Morph Quantization enables users to define how Molekular's Morph function moves from one point to another. This traveling sometimes draws interesting figures in the graphics panel.

With the Pitch Quantization function control panel activated, Molekular can "play" similar to a synthesizer. The notes and patterns are defined in the main graphic window.

Modular Multi-Effects Processor with Modulation and Morphing

Molekular is perfect for performing artists (e.g., live electronic musicians and DJs). In the studio, the obvious users are producers who work with all kinds of contemporary electronic genres, as well as film composers and sound designers.

Molekular is available in stand-alone format (inside Reaktor or Reaktor Player, minimum version 5.9) or in a plug-in format. All the formats supported by Reaktorare available, including VST2, VST3, AU, AAX Native (Pro Tools 10), and 64-bit AAX (Pro Tools 11).

System requirements (these apply to Reaktor, since Molekular runs in Reaktor) for PCs include: Windows 7 or Windows 8 (latest Service Pack, 32/64-bit), Intel Core 2 Duo or AMD Athlon 64 X2, 2 GB RAM (4 GB recommended). System requirements (these apply to Reaktor, since Molekular runs in Reaktor) for Macs include: OS X 10.7, 10.8 or 10.9 (latest update), Intel Core 2 Duo, 2 GB RAM (4 GB recommended). For information, visit www.native-instruments.com.

About the Author

Fernando Rodrigues began studying music and technology in the 1970s. His goal was to marry his two passions: music and computers. As a student, he helped assemble the electronics music studio at the music college in Porto. Later, he directed the technology department at one of Portugal's major distributors while pursuing a career teaching musical analysis and composition techniques. He now concentrates on research and writing about music and technology, sharing his own perspectives about music and sound.

more realistic reverb way but in the density of the sound it created.

After testing them with very different audio material, we drew some conclusions. First, the processors should not to be judged by themselves, since Molekular is not a "regular" audio processor. What makes these processors really shine are the complex modulations that can be drawn and controlled either manually or automated via the host. That's what places Molekular in a category of its own, and what makes it a kind of "synthesizer for audio."

Molekular comes with a generous amount of presets divided into categories including Abstracts and Soundscapes capable of completely transform the sound. That's something not usually found in an audio processor, but this is more a "synthesizer for audio" than an audio processor. It's no surprise that, besides the generous amount of presets, some sound designers already launched third-party packages of presets for Molekular, which is a good testimony to the variety of results this ensemble has to offer.

Results

Molekular is an innovative product. However, it may not be everyone's cup of tea. In fact, those who want regular audio processing or something that always gives them expected trusted results will most likely not be seduced by Molekular. On the other hand, those who seek new sonorities, like experimenting with audio, venture into unexplored territory where sound and noise cross, and view music as a reality in constant redefinition will find that Molekular is a genuine piece of gold.

As we said, it works very much like an audio synthesizer, capable of completely redefining the sound input and sometimes making its own sound. The characteristic quantization of pitched parameters to user-defined and automatable pitch grids, as well as time-quantization for all time-related parameters contribute to this and is a feature that Native Instruments considers key to the product.

Finding a preset that suits our composition requires a generous dose of patience and commitment. It also requires us to master the modulations included in the ensemble since these are the heart of its sound, more than the processors themselves, in our opinion. Of course, the processors, especially the frequency shifters and the spectral shapers, help a lot in these tasks.

The Molekular's documentation comes in a 159-page PDF. It covers, one by one, all the DSP processors, as well as the inner works of the modulators, Macro controls and the Pitch Quantization. However, some things are not very clear (or they simply are difficult to understand). Therefore, a step-by-step tutorial can be of great help. Native Instruments offers four online tutorials— Overview, Modulation, Morphing, and Effects—that were created by Dan Herbert from Point Blank Music School.

Watching them greatly helped us. We particularly recommend watching the second one, Modulation, and the third one, Morphing, since these aspects are necessary to fully understand the Molekular's potential and extract the very best of it.

One powerful feature of Molekular is its modularity. The Routing control panel enables users to interconnect the modules in several ways.

An Interview with Denis Goekdag, the Creator of Molekular

***audioXpress*: You founded Zynaptiq with Stephan Bernsee, formerly of Prosoniq. Can you describe the Zynaptiq "style" and how Denis Goekdag fits in it?**

DENIS GOEKDAG: Our aim at Zynaptiq is to develop products that are based on technology that hasn't been invented yet or that isn't in widespread use in our particular field. We draw from Stephan's almost 30 years of unique research to create products that are easy to use, despite being based on a new approach. I have a strong background in professional audio production, including about a 100 record releases as well as many sample libraries (for example, the TRANSISTOR PUNCH and PULSWERK Machine Expansions, as well as factory content for pretty much all of their other products). Because of this, I know all the existing tools inside out, which gives me a very good view on the market and on what makes a product good. So generally, Stephan handles the signal processing research and development, and I specify and design the products. As with all start-up companies though, there's obviously overlap. Stephan might have a good product inspiration or workflow concept, or I might have a suggestion on how to fix a specific audio issue. Also, on strategic topics we typically develop the vision together, and we take it from there—but in general, he does the DSP and I do the rest.

***audioXpress*: Who came up with the idea of Molekular and how did it appear?**

DENIS: Molekular work started before Zynaptiq was around, so I actually designed the DSP myself (and some bits came from Native Instruments, such as the Metaverb, for example). I came up with the idea for Molekular as well as an "ugly draft." From there, I worked on an excellent team with Native Instruments people to evolve it to what you see now.

***audioXpress*: Did the structure of the modular synthesizers somehow inspire the conception of Molekular?**

DENIS: I've been using modular systems of various types since I was about 15 years old, so that did, of course, have an influence. The design goals were to make a creative multi-effects processor that allows for an immediate, hands-on workflow as well as for detailed editing, and to make sure that locking to tempo and key/harmony is fast and easy at all times.

***audioXpress*: How do you compare the more "regular" processors to the average professional dynamic processors? You have some "regular" processors on Molekular, but somehow, it seems difficult to get "regular" results (maybe because there are so many modulators, and as soon as we start using them, nothing is "regular" anymore)**

DENIS: Exactly. It's easy to leave the "regular" territories. But also, Molekular isn't intended to replace your mixing dynamics processors or Lexicon reverbs, hence, we didn't plan on competing with them. That said, Molekular is very cleanly coded in terms of audio quality, and it is very flexible, so you can actually get rather elaborate and high-quality "studio" processing if you set it up that way. But doing unique, new sounds is the main purpose, and that is what we made easiest.

Denis Goekdag demonstrates his Zynaptiq technology.

***audioXpress*: How do you position Molekular in the "audio processing" territory?**

DENIS: Well, *users* of Molekular and other multi-effects processors I've designed (e.g., DEEP FREQ and DEEP RECONSTRUCTIONS) often want "a fresh, unique sound and don't want to spend hours on editing." There are many releases by well-known electronica artists where my effects have contributed a lot to the final result. At any rate, it is definitely a "creative" type processor, as well as a "performance" type concept.

Fresh from the Bench

Blue Microphones Mo-Fi Active Headphones

Reimagining Headphones

The Mo-Fi headphones are Blue Microphones first entry into a crowded product category and have quite a few unique attributes. The Mo-Fi headphones are a circumaural (full-sized over-the-ear rather than on-the-ear) closed-back configuration with an onboard headphone amplifier.

By
Mike Klasco and Nora Wong
(United States)

Headphones are a completely new category for Blue Microphones but with the Mo-Fi, it seems the company is not trying to follow any of the big brand names in that product segment. Instead, Blue Micropones is trying to create an entirely new category with an active solution for the professional studio environment and audiophiles. In doing so, it also tried to rethink every aspect of headphone design—from the multi-jointed design to earcups (shaped like ears)—completely reimagining form and functionality. Blue Microphones states the Mo-Fi feels “less like a headphone, and more like a high-quality instrument we can wear.”

In our opinion, the Blue Microphones’ Mo-Fi headphones look more like a science-fiction device that could have been designed by H. R. Giger. It is easy to imagine these headsets on the heads of space pilots and that only helped stimulate our interest.

We liked the statements made in the product announcement, heard some impressive opinions expressed by first reviewers, and naturally also had to experiment to determine for ourselves if the Mo-Fi headset was something completely revolutionary.

Blue Microphones

Blue Microphones was founded in 1995 and is recognized worldwide for its truly unique microphones. True to its technical innovation and cutting-edge design roots, Blue Microphones has created a family of inspiring audio tools, from the Bottle and Blueberry professional studio microphones to the project-studio-oriented Snowball USB mic and newer consumer desktop solutions (e.g., the Yeti Pro and the Mikey Digital portable recorder for iOS devices). Now, Blue Microphones’ portfolio also includes headphones, and the company seems to be determined to make a grand entrance in the new product category.

According to Blue Microphones, the Mo-Fi headphones have a built-in 240-mW amplifier, matched to high-powered, 42-Ω precision drivers.

The Mo-Fi headset has a "heavy metal" design (physically, not music genre) that uniquely features a well-constructed multi-jointed hinge. The hinge enables uniform cushion pressure for various head shapes while keeping the ear-cups parallel to the ears.

The intent is to provide the best performance and sound quality from every device—from professional studio gear to laptops, tablets, and even smartphones—providing a new active solution that reduces the audio workload on those devices.

The powered design also enabled Blue Microphones to use high-quality drivers that accurately and dynamically reproduce sound when combined with a lot of power (± 5-V rail for the amplifier), even with high-resolution audio sources and for studio monitoring use. The custom-designed 50-mm dynamic drivers are some of the largest on the market. Blue Microphones says it provides "the best balance between efficiency and accuracy when matched to the built-in amp. Unlike other headphones that rely solely on the teeny amplifier built into your phone or computer, Mo-Fi's drivers and amplifier are custom-matched to each other so they always reproduce audio with the greatest accuracy and dynamic response."

Probably Mo-Fi's most striking aspect is the carefully crafted design, which also contains unique concepts that rethink form and functionality. Mo-Fi uses a multi-jointed earcup design. An adjustable tension knob that enables users "to dial-in the perfect 'Goldilocks' setting –not too tight, not too loose, but just right" complements the earcups.

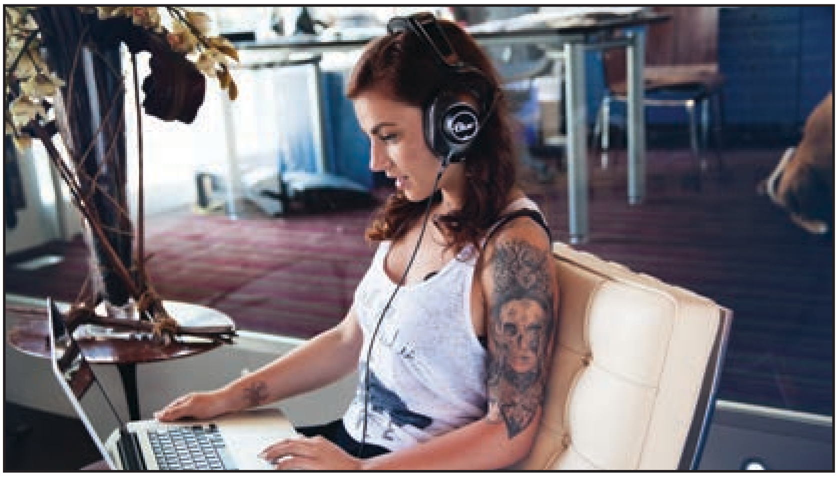
For the headband design, Blue Microphones avoided using the traditional "spring-loaded" approach. Instead, it opted for a multi-jointed headband solution that the company says was inspired by the finely tuned suspension of open-wheeled Formula One race cars.

The result—apart from being comfortable and adaptable to a variety of head shapes and sizes—creates a better seal, resulting in solid bass response (+4 dB bass boost at 62.5 Hz), improved isolation, and a reduction in sound bleed. The headband also allows for different height adjustments of the earcups, using adjustable pivoting arms that also enable angle adjustments.

Mo-Fi's ear-shaped earcup design also plays an important role in how the drivers reproduce sound, contributing to better ambient noise reduction while still enabling the driver to breathe.

Blue Microphones decided to complement that "acoustic" headroom in reproduction by giving the user the ability to adjust three analog amplifier profiles: the standard active mode, a low-frequency enhancement circuit for more bass, and the "off" passive mode. The passive mode enables users to connect the headphones to high-performance studio gear that may already have amplification for headphones (or for use when the battery is at the end of its charge).

Since Mo-Fi uses an integrated amplifier, there's also an integrated rechargeable battery for 12 hours of listening. The battery completely charges in 3 to 4 hours, with an intelligent power management function that detects when the headphones are not in use. If the battery runs down, Mo-Fi converts to a passive system, which enables further listening.



The earcups have huge well-engineered "shaped-like-ears" cushions. These provide an excellent seal, resulting in consistent bass response (very useful in monitoring), improved isolation, and a reduction in sound bleed.

The system will automatically power on and off when headphones are opened and closed. We specifically asked Blue Microphones about battery replacement (which eventually will be necessary) and received confirmation that this is purely "serviceable by Blue Microphones."

Ears On

When we received our Mo-Fi sample, we sent it to our on-duty headphone reviewer—a demanding audiophile with a no-compromise attitude toward music listening who also has an open mind to anything new that we recommend.

To our surprise, after a few listening sessions our reviewer returned the headphones stating he was unimpressed with the experience. He felt the Mo-fi was too "bass heavy" for his taste. Our reviewer is into extremely high-end headphones and owns an extensive collection of the best the industry has produced. His verdict: the balance and clarity were off and the bass control went from too much to over the top. He called it "a DJ headphone!"

From our viewpoint, after a few listening sessions, we felt the bass was clean, tight, and maybe a bit much, even in the normal active mode. Yet, the bass was solid, effortless... and unlimited! But, we could understand our "audiophile" reviewer's problem. There is no sense pushing a great steak on a vegetarian.

Blue Microphones enables the Mo-Fi user to choose from three response balance profiles: an enhanced bass mode, along with active and passive settings. Unlike most other powered headphones, if the battery runs out, Mo-Fi will continue to operate in passive mode (assuming your driving headphone amplifier is up to the task). The Mo-Fi headset is charged using a USB connection.

Next, we sent the Mo-Fi headphones to someone with a completely different background—an audio engineer with extensive experience using headphones for mix and mastering, who was also familiar with the DJ market. We thought the clear definition on the midrange and strong accentuation of the bass would appeal to that user profile and we would be able to obtain a nice review.

This reviewer, who has an extensive background in magazine reviews and field tests, also returned the Mo-Fi headset saying it didn't compare to his reference headphones. Also, he thought the complex headband construction would be awkward for many DJ styles of cueing.

Our final and definitive resource was our own in-house team, who are probably more accustomed to evaluating products for engineering than anything else, and prone to their favorite speakers in place

Blue Mo-Fi Specifications

Amplifier

- Output power: 240 mW
- THD+N: 0.004%
- Frequency response: 15 Hz–20 kHz
- SNR, self noise: less than 105 dB
- Noise: less than 20 uV
- Battery capacity: 1,020 mAh

Drivers

- Type and size: 50 mm, fiber-reinforced dynamic driver
- Impedance: 42 Ω
- Frequency response: 15 Hz–20 kHz
- Enclosure details: Sealed enclosure with tuned damping materials
- Weight: 466 g (16.44 oz)

The Blue Mo-Fi headphones ship with a soft case, which has a cable storage pocket, a 1.2-meter audio cable with Apple iPhone/iPad controls and an extra 3-meter audio cable, plus a USB charging cable, an AC charging adaptor, a 3.5 mm to 0.25" TRS adaptor, and a two-prong airplane connector. The Mo-Fi headset costs $349.99.

The Mo-Fi audio cable is removable, which allows for easy field replacement. It is a sensible feature that adds a bit of cost but users can avoid having to send the headphone back for repair and can keep a spare cable handy.

of cans. Anyway, at least one of them had extensive experience with headphone listening for pure pleasure and could provide us with more extensive feedback.

Unfortunately, our third reviewer also gave the headphones a "thumbs down." An extract of the short report says, "this product misses the mark... they tried a lot of innovative stuff but lack experience to change the rules... The cushions are comfortable but the weight is way too heavy for long-term comfort. How is a mixing engineer going to use these all day long?"

He concluded that "the Mo-Fi is pitched at, and voiced for DJs, more than studio reference monitors." The problem is, we had already received feedback from an experienced DJ, and he didn't seem to agree.

The Mo-Fi's radical design seems to be the most positive aspect, although it is still subject to criticism: "The unit is comfortable—but it is very heavy and long-term listening is arduous with this weight."

So, apparently Blue Microphones' first effort in the headphone market, like its design, seems to be "out of this world." So much so, that we couldn't find the right application for it.

Blue Microphones' first effort in the headphone market, as with its design, greatly differs from everything else currently available.

Mo-Fi's Positives

audioXpress readers involved in recording already know and respect Blue Microphones for its professional and prosumer microphones. Many of them also know the brand doesn't design its products like every other manufacturer. We could expect that, much like it does with uniquely styled microphones, the Blue Microphones team would explore new directions when venturing into headphones.

While most studio microphones sit on a mic stand or a boom, headphones are quite another species, and of course, are worn. Shoes, gloves, and other worn accouterments have additional crucial design requirements, including the way they look on the wearer as well as ergonomic comfort.

Most distinctive is the headband design. Blue Microphones succeeded in the multi-jointed headband solution and adjustable tension knob, avoiding using the traditional "spring-loaded" approach where the pressure varies with head size and the angle of the earcups.

The result allows for different height adjustments of the earcups, using pivoting arms that also provide angle adjustments. The headband has a tension knob that is self-calibrating but adjustable, which might change during a long listening session. Yet, the execution of the multi-joint hinge system is a mixed blessing. It is complex and huge. It also results in just about the heaviest headphones we have ever experienced.

Regarding the unconventional design decision to use an on-board headphone amplifier, in its dedicated Mo-Fi website Blue Microphones says:

“Frankly, it’s surprising no one ever thought of this before.” This seems a very odd statement, hopefully coming from one of its marketing guys and not the engineering team! While self-amplified headphones are a special category and have some unique advantages over conventional (passive headphones) this is not all that new. Of course, all active noise-canceling (ANC) headphones have on-board amplification and they were introduced by Sennheiser, Bose, and NCT Noisebuster decades ago.

However, not all amplified headphones are about ANC. For example, the Dr. Dre Beats Studio is the best-known powered equalized headphones (its ANC is a secondary feature). The headphones were developed around 2006 and launched in 2007. A more recent take on self-powered headphones is JBL’s Synchros S700 signal-processing headphones with spatial imagining as well as equalization.

And while the Blue Microphones team may have reinvented the light bulb, the question is still why powered? The open-back earcup type is easier to get a good sound balance acoustically, but the music will leak out and outside noise will leak in. If the Mo-Fi headphones are to be used for monitoring, the outside ambient noise cannot intrude. The Mo-Fi headset tries to provide much of the best of both approaches—it achieves the sound balance through active equalization, which is made possible due to the onboard amplification but with closed-earcup construction.

And, Blue Microphones decided to complement that “acoustic” headroom in reproduction by giving the Mo-Fi user the ability to adjust the response balance profiles.

The bass can be described as authoritative, powerful, and tight. Not quite boomy but in the enhanced mode it is too much bass for our taste. However, it is easy enough to switch to the normal bass mode or passive mode.

Using an iPhone 6 Plus as the signal source, we found the (powered) On switch as the best sounding mode and the same when using the OPPO HA-1 headphone amplifier. With the On+ the big deep sounding bass was too intrusive. The headphones even shake at higher levels. In the passive mode (Off switch), the bass was just not as clean. It was a bit fuzzy and distorted, probably due to our listening levels and the iPhone 6+’s lack of amplifier headroom.

Getting away from the bass, the localization and spatiality were excellent, and the dialog had exceptional clarity as long as the On+ was not used.

The Mo-Fi’s sound is powerful and high sound levels can be achieved without strain. Typical full-size headphones use 40-mm diameter drivers, while the Mo-Fi headset boasts 50-mm drivers. Usually a 50-mm (2” diameter) driver might lack top-end and yes, the Mo-Fi’s balance tends toward bassy.

Mo-Fi’s ear-shaped earcup design also plays an important role in how the drivers reproduce sound, contributing for better ambient noise reduction while still enabling the driver to breathe (through venting).

Comparing them with several other headphone choices (some of which are three to four times the price of the Mo-Fi headphones), the Mo-Fi was comparable to the Beats Dr. Dre Studio, which is another powered headphone. The Mo-Fi is definitely not for the Beyer T1 or the Sennheiser HD 800 “enhanced-clarity” crowd.

Blue Microphones gets points for attempting a lot of innovative design choices but perhaps lacks enough experience to change the rules.

Overall, it is an impressive first headphone entry for Blue Microphones. At $350, Mo-Fi is a lot of headphone for the price. We expect the next generation will show further refinement on weight and sound balance.

Resources

Blue Microphones, www.bluemic.com.

H.R. Giger, www.hrgiger.com/frame.htm.

Mo-Fi Headphones, mofiheadphones.com.

Nordic Audio Tales (Part 5)

Peter and Dorit Larsen's LOUDSOFT

This Nordic series article focuses on the Danish company LOUDSOFT and its founders Peter and Dorit Larsen. In fact, many of the tweeters we use today grew from the imagination of Peter Larsen.

By
Mike Klasco
(United States)

Affectionately known in the speaker industry as "Tweeter Peter," a moniker dating back to the 1990s and his days at JBL, Peter Larsen and his wife Dorit are business partners in a company called LOUDSOFT, which develops speaker design software and speaker test instrumentation. The company is located in Horsholm, just north of Copenhagen in Denmark. Horsholm was developed in the 1730s, but several of the communities that make up the municipality have even older origins. LOUDSOFT's offices and lab at LOUDSOFT are located in a science park, readily accessible by the train system.

LOUDSOFT's headquarters are located in Horsholm, just north of Copenhagen, in Denmark.

Peter Larsen

Peter Larsen's 40-year career and innovative contributions to speaker engineering began at SEAS in 1974. Next, he was Chief Engineer for Vifa-Speak from 1979-1987, and then he went to Dynaudio, where he worked from 1987–1990 as chief engineer.

While at Dynaudio, Larsen developed the D260 Esotec, and the 15W75 and the 20W75 cast-aluminum basket speakers. Larsen also developed a special production method, which made production more efficient and gave the products a higher and more stable quality. At the same time, the workers received more interesting jobs, a "must" in Denmark, which is a pioneer for good employee working conditions.

Next, Larsen worked at JBL in the US until 1993. After 1993, he became an independent consultant for leading loudspeaker factories worldwide, including Audax in France, KEF Audio in the UK, Goldmax in China, Vifa-Speak in Denmark, Peerless Fabrikkerne of India, NXT in the UK, Microlab in China, Apple in the US, and many others.

While Larsen has significantly contributed to dome tweeter design, at first glance they all appear very similar. In the tweeter world, more noticeable and unique is his ring radiator design. According to Larsen, he created it one Friday afternoon out of coincidence, curiosity, and skill. While working on a dome tweeter project at Scan-Speak, things were not going as intended and repeated efforts did not yield the desired results. Being resourceful, he took extreme measures, which consisted first of consuming a couple of beers (it was the end of the week after all). Larsen then decided to punish this tweeter that would not behave and stuck a pin in it! Driven by curiosity, he found it sounded better. The tweeter measured even better when clamped in its center! This became the ring radiator, a design favorite copied by many Asian factories (although Tymphany is a legitimate vendor) and is used in thousands of loudspeaker systems.

Later, as an independent consultant, Larsen specialized in in-depth analysis of loudspeakers and manufacturing techniques, which include research on new components and materials, advanced Acoustic Finite Element modeling, new measuring methods, and many novel speaker design concepts. During the same period, Larsen also developed and globally marketed the LOUDSOFT speaker development and design programs. Larsen has also written several papers and given many workshops at the Audio Engineering Society (AES) conventions and the Association of Loudspeaker Manufacturing & Acoustics (ALMA) association events. He has also written articles for *Voice Coil* and other publications.

The LOUDSOFT Team

Dorit Larsen first became involved with the loudspeaker industry in 1974 when her husband began his career in loudspeaker engineering. In the late 1980s, Dorit helped Dynaudio obtain substantial support for export and development from the Danish state, which greatly assisted Dynaudio's ongoing growth. Since 2001, Dorit has spent most of her time with LOUDSOFT, where she is in charge of administration, sales, and marketing. Dorit was elected to ALMA's Board of Directors in 2007. As the Europe Vice President of ALMA, she has organized its industry events at the Frankfurt Prolight + Sound shows.

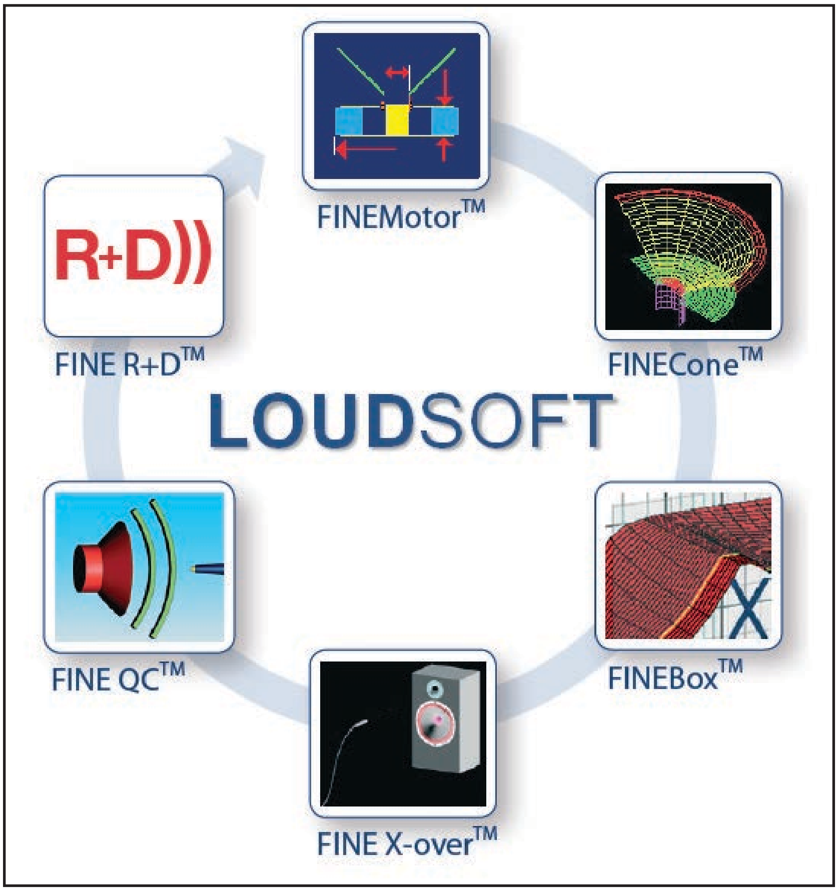
Dr. Yu Luan ("Spark") is Manager of R&D and specializes in advanced mechanical and acoustical development. He received his PhD from the Technical University of Denmark (Copenhagen) in 2011. His thesis "Modeling Structural Acoustic Properties of Loudspeaker Cabinets" was done in cooperation with Bang & Olufsen (B&O).

LOUDSOFT is essentially the sales vehicle for Peter Larsen's creative work: from consulting to intuitive software programs to acoustic test systems. Over the years, he has developed a considerable amount of software and hardware.

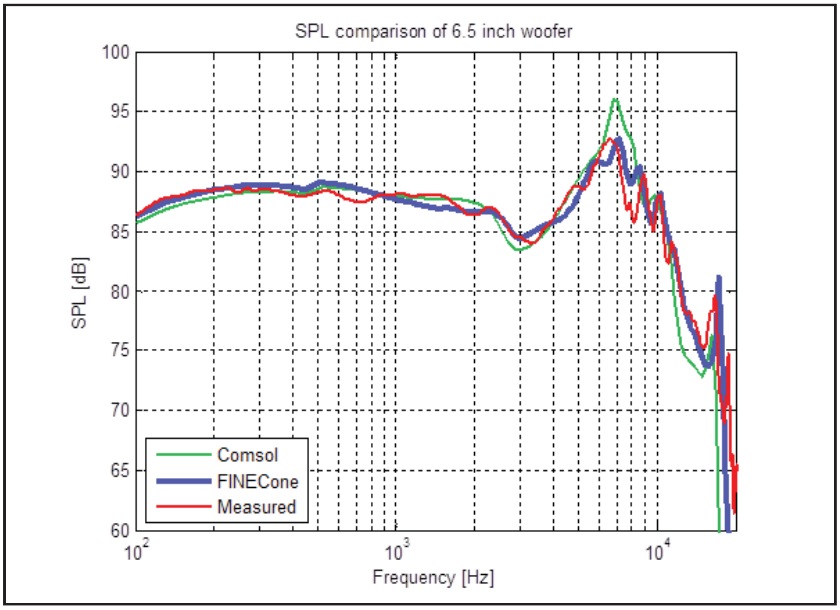
FINEMotor

When Peter Larsen joined Dynaudio in 1988, he faced the challenge of designing woofers with extremely large voice coils. As he was unsure how to calculate the basic parameters for these drivers, he decided to develop some software specifically for this purpose.

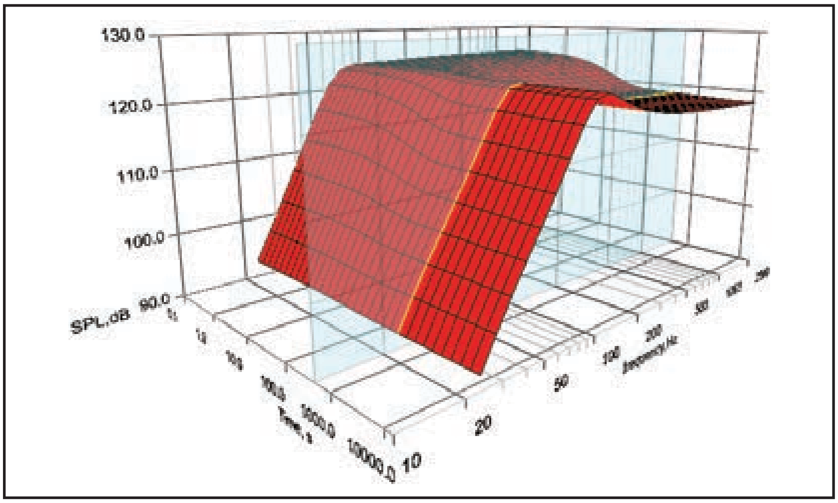
Wanting to be able to optimize the voice coil and the magnet for the best performance, he calculated the sound pressure level (SPL) and the Thiele-Small (T-S) parameters for each available wire diameter using a fixed direct current resistance (DCR) and magnet size. That was a lot of data, so to gain a better overview, he plotted the SPL and Q_{TS} vs. the



The FINE Circle connects all the design software and measuring systems together to form a total development and test system for loudspeakers, headphones, and micro speakers.



The FINECone program is used to troubleshoot the design of cones, domes, surrounds, dust caps, and more, as well as optimize the driver.



The FINEBox is a nonlinear high-power box design program for hi-fi, PA, and micro loudspeakers.

voice coil winding width for each solution. Using the plotted graph, it was easy to select the solution that provided a satisfactory SPL plus X_{MAX} and Q_{TS}.

In 1994, Peter Larsen asked Bob Harris from KEF in the UK to program this in C++ as WinMotor, which could run on Microsoft Windows platforms. Later, the program was renamed FINEMotor and is still based on the main principle by plotting all parameters for each wire diameter and storing them in an editable database.

FINEMotor is used as an application specific program to design magnet systems and voice coils easier and quicker than the general purpose magnet simulation programs. The latest version optimizes pole pieces and predicts BL(x) curves and voice coil offset. Another new capability is for simulation of microspeakers with rectangular voice coils and motors. In 2014, FINEMotor passed 1,000 licenses as the industry standard for loudspeaker motor design.

FINECone

In 1995, after listening to a presentation by Dr. Henrik Vollesen of his PhD thesis "Analysis of the Mechanical and Acoustical Properties of a Loudspeaker Unit with Controlled Directivity," at the Danish Loudspeaker Society (The "Coffee Club"), Larsen and Vollesen decided to collaborate on the development of a practical Acoustical Finite Element software suitable for loudspeaker analysis. The design would be based on Dr. Vollesen´s doctorate work.

Dr. Vollesen´s software did not include all the relevant components and did not use common loudspeaker terms, and so forth, so there were many adjusts. After six years and more than 6,000 hours of programming, experimenting, verifications, and frustrations, FINECone was finally presented in 2001 as the first real analysis and design tool for loudspeaker engineers. Since then, Larsen has gained a deep understanding of cone break-up in all kinds of woofers, midranges, dome tweeters, headphone drivers, and micro speakers.

FINECone has become the industry standard, because it is extremely fast. It is also much simpler and more effective than other finite element analysis (FEA) software. The program is designed for troubleshooting the design of cones, domes, surrounds, dust caps, and more, as well as optimizing the driver. FINECone animates cone/dome break-up and predicts the SPL response including fine details.

FINEBox

FineBox is a nonlinear high-power box design program for hi-fi, PA, and micro loudspeakers. The software provides simulations of voice coil temperature and compression at high power in closed box, reflex, ABR, bandpass and inter-port alignments. All nonlinear T-S parameters and thermal data can be imported directly from FINEMotor. The latest version can also simulate micro speakers and very small volumes, including headphone rear cavities and holes.

FINE X-over

FINE X-over is the ideal program for optimizing a loudspeaker system. The user can scroll through standard E24 components and immediately see the responses, including impedance and acoustical phase changes.

The Intelligent Optimiser will find the best flat response while keeping the minimum impedance and calculating the real power in all components. The program imports curves from FINE R+D and most other professional systems, and can optimize up to seven on- and off-axis responses.

FINE QC

In 2007, the FINE QC End of Line Test system was introduced. Having worked in the loudspeaker

FINE QC is a quality control system for loudspeaker testing, using a standard PC with Windows XP. The intended use is end-of-line testing. The system includes the FINE QC hardware, which has USB control and soundcard built-in as well as a 25-W amplifier.

industry since 1974, Larsen was frustrated over the lack of good quality control (QC) test solutions. The existing systems were all designed by electronic engineers and had interesting features, but lacked the necessary features required in loudspeaker production.

Inspired by MLSSA, FINE QC was developed using the Farina Sine sweep method. The Fast Fourier Transform (FFT)-based system can exclude reflections, eliminating the need for an anechoic chamber. The system also introduced a number of firsts: Automatic Golden Average Speaker finding, which saves days of work measuring and comparing samples; FINE Hardware, with microphone plus analog-to-digital and digital-to-analog converters; a power amplifier, providing simultaneous SPL and impedance test without switching; and the unique FINEBuzz algorithm. This is based on Danish research, which found a new principle for Rub & Buzz detection using a combined harmonics and transient method. The latest version has replaced human testing of micro speakers in China. A complete test, including Rub & Buzz can be done in 0.5 to 1 s.

FINE QC is growing in popularity because all measured data is automatically saved for Statistics. Interestingly, most of these features (except FINEBuzz) were later been adopted by the competition.

FINE R+D

Based on customer reactions, Larsen realized in 2011 that he could expand the measuring system to include all the daily lab loudspeaker measurements. The intuitive FINE R+D system uses the same FINE hardware as FINE QC, providing an R&D system with all advanced features such as up to 16 anechoic FFT measurements, T-S parameters, SPL, and impedance with phase, harmonic distortion, waterfall, real-time analysis (RTA), and numerous curve splice/averaging options. The latest version even saves the time data so the engineer can change the FTT window and other settings later, giving unseen flexibility.

FINE R+D is the last element in the FINE Circle, which connects all the design software and measuring systems together to form a total development and test system for loudspeakers, headphones, and micro speakers.

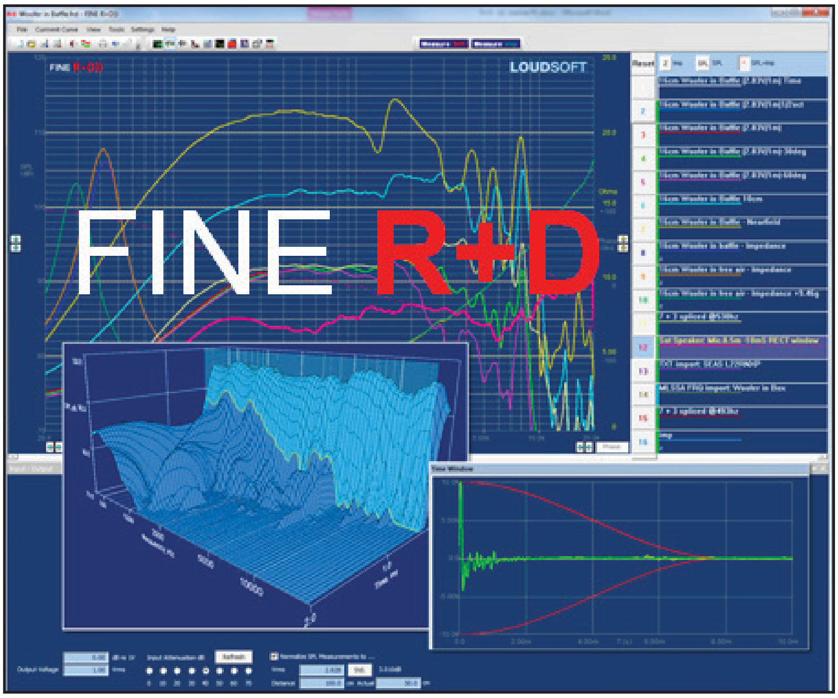
Larsen Continues the Nordic Audio Tradition

ALMA's Titanium Driver Award recognizes a specific technical contribution, accomplishment or expertise in the loudspeaker industry. Larsen received this award at the annual ALMA symposium in Las Vegas, NV, in January 2012. Specifically, he received the award for his constant efforts and expertise in driver development and his hunt for better speaker designs for the loudspeaker industry worldwide.

Larsen is still very active as a consultant in all kinds of loudspeaker and headphone projects, which helps him to understand the practical problems and find the best software and test solutions to address them. In this way, customers are the most valuable resource by requesting new special features and inspiring further interesting solutions.

For more information, visit www.loudsoft.com. LOUDSOFT, Ltd., is located at Agern Alle 3, DK-2970, Horsholm, Denmark.

FINE QC is an end-of-the-line quality control test system for active loudspeaker systems and drivers, headphones and microspeakers.



FINE R+D measures loudspeaker, headphone, and microspeaker development.

Headphone and Earphone Design Considerations

This month, we examine the headphone and earphone product development process. In future articles, we will discuss the many aspects of headphone product development—from driver designs to the secrets of ear cushions.

By
Mike Klasco and Steve Tatarunis
(United States)

Most speaker system engineers know how to develop a product for the audio market. Price points are selected, target products defined, the brand's DNA, the product series, and the buyer's expectations are known. All these variables drive the product definition in some very familiar ways for established original equipment manufacturers (OEMs) and original design manufacturers (ODMs) vendors. But over the past few years, most speaker brands have added headphones and earphones to their product mixes and have found themselves in unfamiliar territory.

Applying traditional speaker system product development procedures to earphone and headphone devices has been challenging for even the most experienced development teams. While painful for the manufacturers, this means job security for consulting companies such as Menlo Scientific (which is owned by article co-author Mike Klasco). Brand executives and engineering teams can be more receptive to outside help when they are exploring unfamiliar territory. On the other hand, attempting to salvage projects that are in trouble are more difficult.

Let's start with the initial questions the team (e.g., management, marketing and sales, and engineering) needs to consider.

Brand DNA and Brand Permission

Know where your existing products are positioned in the market. For example, Menlo Scientific's Eric Kowalski handles business strategy and merger and acquisition details for the company's clients and he uses a four-quadrant chart to help define their brand "self-awareness."

Figure 1 shows The lower left quadrant is the commodity product with no style or sizzle. The right lower quadrant details products with style, but no technology. The upper left quadrant is for products that have technology, but no style, and the upper right quadrant is the marketer's dream product, which offers style and performance. The "marketer's dreams" usually become inspirational and iconic products that come with premium prices (and premium profits).

In the automotive world, Aston Martin and Tesla have well-deserved spots in the upper right quadrant.

Of course, most of the larger brands have many product lines with different DNA. Consider Harman, which offers dozens of headphone models—from JBL Lifestyle to AKG's professional and audiophile models to Harman-Kardon's high-style and high-grade construction. Even within those brands, there is a wide span of price points as well as a huge range of target customers. So on our four-quadrant chart, we would be more specific (for at least the larger brands) and indicate each brand type or separately list each sub-brand on the chart.

Reinvent and Position a Brand

For those who grew up in the US, we know the market position of many well-established brands. For example, Best Western and Holiday Inn are mid-level brands that can greatly differ depending on location. Before staying at one of these motel chains, read the online reviews. The quality and consistency greatly vary because many of the franchises are independently owned. Yet in China, the Holiday Inns are consistently a 5-star hotel brand, and in Europe, there are many boutique and long-established hotels under the Best Western brand. While this is geographic brand "re-invention," a speaker company can also select a new brand direction. Launching a headphone and earphone product line is one place to start.

A company could define its new headphone or earphone devices for sports fitness, high style, technology, or some combination of enhanced attributes and innovations. One game-changing success story is Monster. This cable power strip and accessory brand skillfully teamed with Interscope for the Dr. Dre headphones and earphones, and now dominate this market (see **Photo 1**). Monster's hijacking of the headphone market from the legacy leaders (e.g., Sony, AKG, Beyer, Sennhieser, etc.) was a combination of skill, talent, and being in the right place at the right time. Once the new headphone and/or earphone product line positioning is targeted, it is necessary to figure out how to achieve your goal.

Off-the-Shelf or a Clean-Sheet Design

Often, all established loudspeaker companies want are a couple of earphones and headphones to enhance a product line. When nothing out of the ordinary is needed, tooling budget and engineering time are limited, and time to market is short, then the ODM solution is viable. A few headphone factories offer ready-to-go tooled models. Usually the last 20% of product tuning to fit a specific brand is an option. This can be cushions, the headband, fabric or paint finish, upgraded cables, packaging, and so on. The X-02

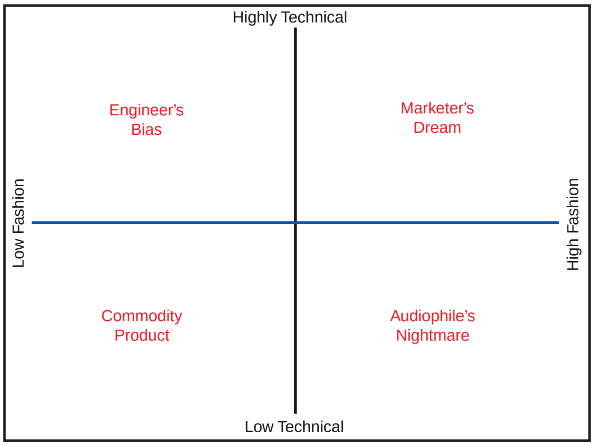


Figure 1: This four-quadrant chart can be used to define brand self-awareness. (Image courtesy of Menlo Scientific, Ltd.)

Photo 1: The original Monster Dr. Dre Headset is used worldwide. (Photo courtesy of Monster)

Photo 2: Yoga Electronics' X-02 is a customizable high-end ODM design. (Photo courtesy of Yoga Electronics)

Photo 3: The Sennheiser HD-800 premium "from scratch" design is a perfect example of a quality design. (Photo courtesy of Sennheiser Electronics)

from Yoga Electronics is an example of a high-end ODM design that can be customized (see **Photo 2**).

"From scratch" headphones can require a dozen tools or more, and tooling can be $5,000 per tool or more. Aside from tooling costs, there are product development risk assessment considerations. Clean-sheet designs can be risky, especially if this is the "first headphone" project for the engineering team and the industrial designer. For headphone development, the mechanical engineering, especially the ergonomics, are challenging. Designing headphones that are comfortable, yet have good retention, require following design rules that are trade secrets of the most experienced headphone factories and headphone engineers. Headphones, like gloves or shoes, must look good and fit a wide range of users. With headphones, we are stuck with a "one-size-fits-all" approach. In addition to designing comfortable headphones, they must also be robust enough to survive store displays where they are handled all day (see **Photo 3**).

Photo 4: The LCD-3 is Audeze's flagship headphone. (Photo courtesy of Audeze, LLC)

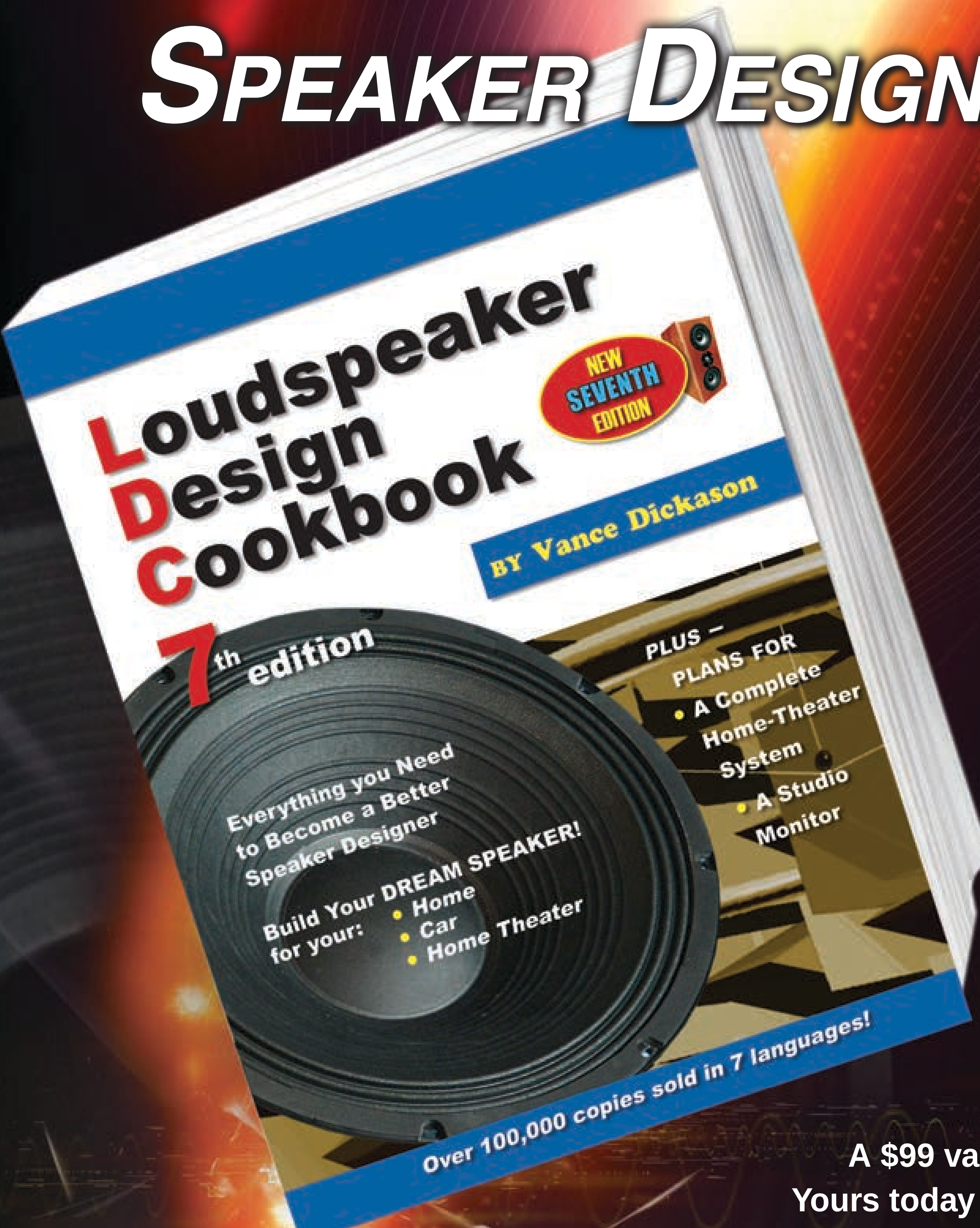
Of course, tooled plastics are not always the best approach. In recent years, boutique brands such as HiFiMAN and Audeze have entered the high-end of the market with ribbon planar headphones that have received great acceptance, yet have very limited tooling investment. CNC-machined wood earcups, stamped-metal yoke-hinged headbands and other aspects provide a handcrafted appearance (see **Photo 4**).

Innovation has its benefits and risks, especially in headphone mechanical construction. When the original Dr. Dre headphones were launched, the headband design was elegant, fit well and was comfortable. However, the mechanical stress of putting it on and taking it off resulted in premature failures—primarily at the retail displays— and this sent the wrong message. So, good mechanical construction is not limited to the headphones just being individually worn.

One professional microphone brand recently launched its first headphones, which feature a unique headband and earcup hinge configuration, the resulting design manages to perfectly fit thin to wide skulls, but the product's appearance and weight are not appealing. This last aspect, weight, is a huge factor in headphone design. Well meaning designers may be tempted to replace ABS plastic and use something unique such as stainless steel, but these "heavy metal" headphones have been a disaster as anything weighing more than 300 grams (without a cable) is too heavy for most users.

A good headphone design road map includes a parts list with component weights so when the final product is assembled, there are no surprises. Techniques such as engineering plastics, magnesium aluminum alloys, metal-injection molding are all powerful tools in keeping the headphone design lightweight and user friendly. Next month, we will examine the contributions 3-D CAD and 3-D printers have made to product design.

BECOME A BETTER SPEAKER DESIGNER!



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Handheld Acoustical Measuring Systems from NTi Audio

By
Richard Honeycutt
(United States)

Acoustical measuring instruments can be classed as handheld systems, desktop or laptop computer-based systems, and personal-device-based systems. In this column, we will examine the NTi model AL1 and XL2 handheld acoustical measurement systems.

When Wallace Sabine made his first acoustical measurements, he used human-blown organ pipes as sound sources. For many years, broadband reverberation time (RT) was measured using an impulsive source such as a bursting balloon or a starter's pistol. Speech intelligibility was a much later metric, but it was originally measured using human talkers carefully pronouncing selected syllables from standardized lists, and human listeners writing down what they heard. Measurement of noise levels, or levels produced by a sound-reinforcement system, awaited the development of measurement microphones feeding calibrated amplifiers and meters.

Today, there are many instruments available for measuring sound—so many, in fact, that we shall make no effort to discuss them all. Most of these instruments are limited to measuring sound levels, either broadband or in fractional octave bands. We will limit our focus to systems that also measure two quantities of great importance in architectural acoustics, noise control and community noise: RT in octave bands and the Speech Transmission Index (STI), which is a key number for quantifying speech intelligibility.

The NTi AL1

The NTi AL1 Acoustilyzer (see **Photo 1**) is similar to the ML1 Minilyzer introduced a few years earlier. The ML1 was designed as a combination real-time analyzer (RTA), speaker polarity tester, sound level meter, RT measuring device, digital oscilloscope, root mean square (RMS) level, and total harmonic distortion + noise (THD+N) measuring device. Both frequency-swept and time-swept measurements are possible on the ML1. The AL1 offers improved RT measurement, fast Fourier transform (FFT) analysis, delay-time measurement for speaker setups, and optionally, STI-PA speech intelligibility measurement in accordance with the 2003

International Electrotechnical Commission (IEC) 60268-16 standards release.

For acoustical measurements, the ML1 and the AL1 accept the MiniSPL self-powered measurement microphone, which can be plugged directly into the analyzer or connected via an XLR cable. This combination creates a Class 2 instrument. Other measurement microphones can be connected through the use of appropriate power supplies and connecting cables. The MiniSPL is pre-calibrated for a sensitivity of 20 mV/Pa.

The AL1's control pad consists of the four-arrows-and-an-ENTER-key arrangement familiar to users of DVD players. Upon powering up the analyzer via the oval button at the upper right, the user can press the ENTER key and select the measurement function from the pop-up menu (see **Photo 2**).

If the SPL/RTA function is selected, the operator can then use the arrow keys to highlight the second field in the top menu line. Pressing the ENTER key brings up a choice between a sound pressure level (SPL) measurement—designated as 123—or the RTA function—designated by a row of vertical lines representing a bar graph. By highlighting the word SET on the left side of the menu's second line, the user can choose 1-octave or 1/3-octave resolution for the RTA. Various fields surrounding the graph display permit adjustment of the vertical and horizontal axis calibration, and enable the user to lock the numeric level display for a specific frequency band. **Photo 3** shows an octave-band RTA display. See the sidebar for more information about RTA and FFT analyzers.

SPL measurements are made by selecting the 123 function on the second field of the top menu line, after SPL/RTA has been chosen. The SPL display shows instantaneous SPL and Leq (effective equivalent level of sound averaged over time). Five frequency weightings are available: Flat, A, C, X-CRV (for cinema installations), and RLB (for evaluating loudness of broadcast material). Either fast or slow response can be selected. **Photo 4** shows the SPL measurement screen.

Measuring RT requires five steps. First, the AL1 is automatically set to measure the background noise in the room in each octave band, after which it sets two pointers for each octave band. The lower pointer represents the noise level in that band. The upper pointer represents the minimum signal level needed for a valid RT measurement. The second step is to set a pink noise source so that the bar graph display crosses each of the upper pointers. Step 3 is to highlight the START field on the screen. Step 4 is to stop the sound. Step 5 is to choose

Photo 1: The NTI AL1 is a complete handheld acoustical measuring instrument. (Photo courtesy of NTi Audio)

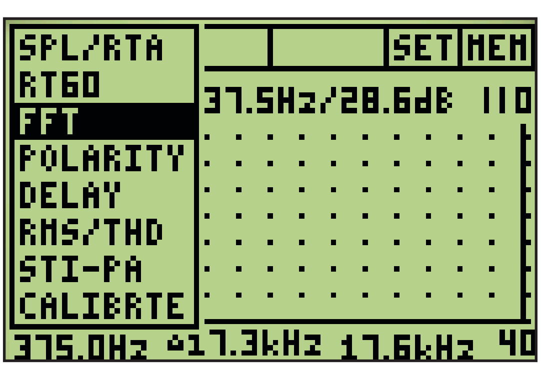

Photo 2: The AL1 screen displays the measurement functions at the left.

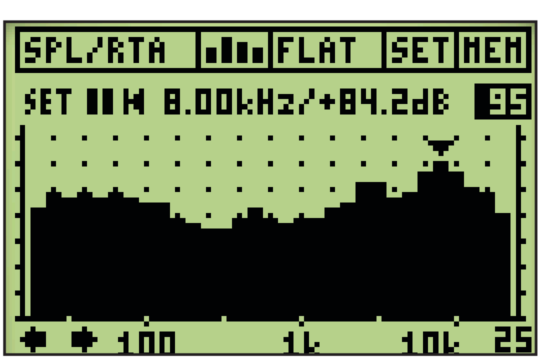

Photo 3: This real-time analyzer (RTA) measurement screen shows the sound pressure level (SPL) is highest in the 8-kHz band.

the 123 option in the second field of the top menu line. The RT values will be displayed, along with the correlation factor for the measurements. A high correlation factor indicates a reliable measurement, and requires averaging, which the AL1 performs automatically. **Photo 5** shows the results of an averaged RT measurement.

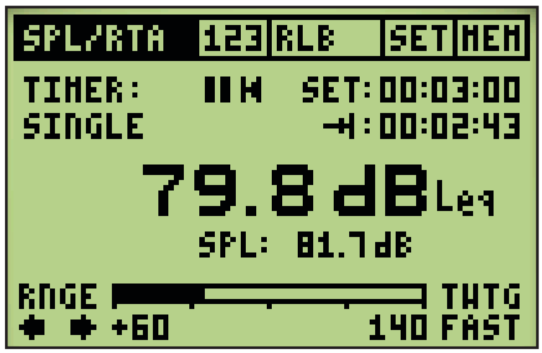

Photo 4: The sound pressure level (SPL) measurement shows both instantaneous SPL and L_{EQ}.

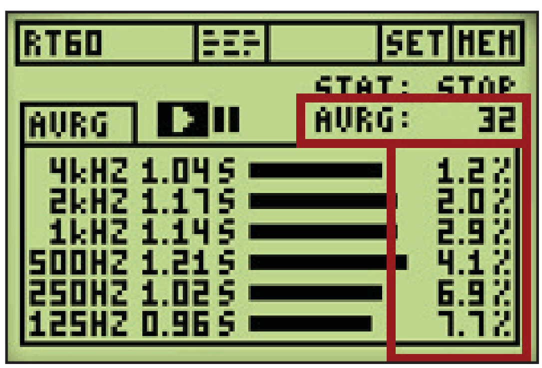

Photo 5: This screen displays reverberation time (RT) results and correlation (shown in the red box).

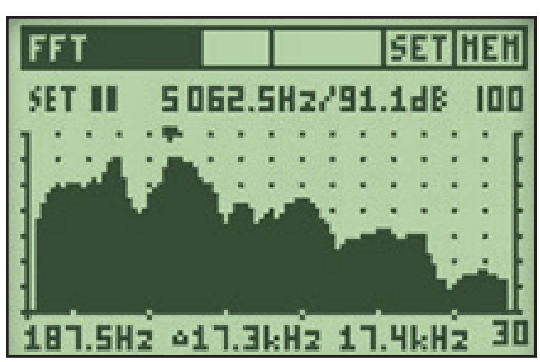

Photo 6: The Fast Fourier Transform (FFT) function can provide much finer resolution.

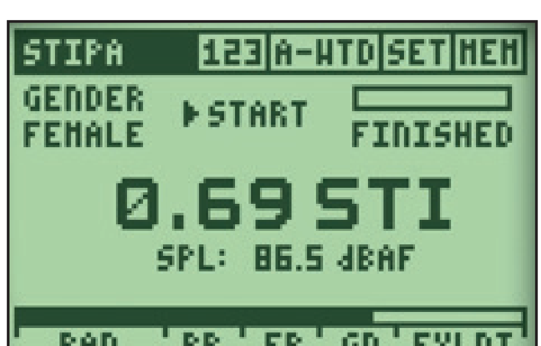

Photo 7: The Speech Transmission Index for Public Address systems (STI-PA) display shows the STI speech intelligibility value, and categorizes it as bad, poor, fair, good, or excellent.

FFT analysis provides a 187.5-to-0.73-Hz resolution over the frequency range of 10 to 20,000 Hz. This function provides information on narrow-band features such as comb filtering. The results of an FFT measurement made on the AL1 are shown in **Photo 6**.

STI-PA measurement requires a special test signal provided on a CD. This signal must be played at a minimum of 60 dB SPL, but preferably at the nominal level of the program material of emergency announcement (e.g., 85 dBAS). The STI-PA function of the AL1 is selected, and the user presses START. The measurement requires 15 s, during which the word RUNNING appears under the on-screen progress bar. When the measurement is completed, FINISHED is displayed (see **Photo 7**).

The AL1 is supplied with NTi's MiniLINK software, which provides a USB computer interface for the instrument. This enables users to remotely operate the instrument while viewing the screen and to download stored screenshots.

The XL2

The XL2 (see **Photo 8**) operates much like the AL1. With the M4260 self-powered measurement microphone plugged in, it is a Class 2/Type 2 analyzer. Other microphones can be used to provide a lower noise floor, or when Class 1-level measurements are required.

In addition to an easier-to-read screen, the XL2 offers a few advantages over the AL1:

- Sound level measurements can be viewed as actual SPL, Lmin, Lmax, Lpeak, Leq, or gliding Leq. Frequency weighting of A, C, and Z (=flat) can be displayed simultaneously. Time weighted values of fast, slow and optional impulse can be seen simultaneously. Correction values k1 and k2 can be entered, and sound levels exceeding a preset limit can be made to illuminate as red, orange, or green, facilitating live event monitoring. There is a digital I/O interface for the control of external peripherals. Optionally, Sound Exposure Level LAE; Percentile statistics for Lxy (x = A, C, or Z, y = F, S, or EQ1): 1 – 99%; calculated levels of LCEQ – LAEQ, LAIEQ – LAEQ, LAFT5EQ – LAEQ; and clock-impulse maximum level "TaktMax" according to DIN 45645-1 can be available.
- Time weighting of fast, slow and (optional) impulse; and percentile statistics (optional) can be available for spectrum analyzer measurements.
- Logging to an SD Card is possible in several modes: all levels simultaneously, short-time levels including spectrum for 1 second or longer, .wav

file for listening (compressed), data logging in 100 ms intervals (optional), linear .wav files with 24-bit, 48-kHz resolution (optional).
- Remote measurement acquisition can be performed (optional).

Available options for the acoustic analyzer include 1/12 octave analysis, pass/fail measurements, and noise curves (NC, RNC, RC, PNC, and NR).

The STI-PA Analyzer is available with these options: speech intelligibility measurement IEC60268-16 (edition 4.0, 3.0 and 2.0), ISO 7240-16, ISO 7240-19, DIN VDE 0828-1, DIN VDE 0833-4; ambient noise correction; automated averaging; and display of modulation indices. The STI-PA Reporting Tool creates measurement reports according to the IEC 60268-16 and VDE0833 standards.

The XL2 is used to measure ambient noise during the day and STI at night. The software enables this data to be directly imported from the XL2. The report combines these and displays the speech intelligibility value that would be expected in real-life conditions. The corresponding speech intelligibility STI or Common Intelligibility Scale (CIS) values are shown. The STI-PA Reporting Tool is free to download at the XL2 Support website.

The XL2's electrical audio analyzer functions include: RMS Level, THD+N, frequency, digital oscilloscope, spectrum analysis (1/3, full octave and optionally 1/12 octave), FFT analysis, and pass/fail measurements (optional).

These additional capabilities may be especially useful in the fields of industrial or community noise or of sound-system work. In addition, many of the systemic improvements in basic design between the AL1 and the much newer XL2 do not appear in a function list. These include many operational, memory, power supply, performance, and other platform changes that make an XL2 much more than an "AL1 +."

Both the AL1 and the XL2 are powered by a trio of AA cells, and the firmware of both instruments can be upgraded free from NTi Audio (www.nti-audio.com). The ML1 and AL1 use essentially identical hardware, and each model can be cross-graded to the other via a firmware change. For more information, visit www.nti-audio.com.

Photo 8: The XL2 is an advanced acoustic analyzer. (Photo courtesy of NTi Audio)

RTA and FFT Analyzers

Real-time analyzers (RTAs) and Fast Fourier Transform (FFT) analyzers belong to different classes. An RTA displays levels in fractional octave bands, which makes it a constant-percentage-bandwidth analyzer. This type of analyzer is handy because the human ear/brain system also functions on constant-percentage bandwidth.

An FFT displays levels in "bins" of a certain number of hertz (Hz), which makes it a constant-bandwidth analyzer.

So, a 1/3-octave RTA has a bandwidth of 3.7 Hz (17.8 Hz–4.1 Hz) in the 16-Hz band, and 3,700 Hz (17,800 Hz-14,100 Hz) in the 16-kHz band. In both cases, the bandwidth is 23% of the center frequency. An FFT analyzer with a 10-Hz resolution has that same resolution at any frequency within its operating band.

Forte Makes a Name for Itself

An Interview with Forte Founders Scott Chen and Travis Blane

Travis Blane is one of Forte Audio's three founders and is the least shy of the three.

By
Shannon Becker
(United States)

SHANNON BECKER: Tell us about your company Forte (myforteaudio.com).

SCOTT CHEN: The research for our audio products originated as conservation research to extract the sound of a bullfrog from a noisy recording of the surrounding forest mountain and waterfall. Dr. Jianping Pan developed a signal processing method to be able to extract the clear and uncontaminated sounds of the frog. He was even able to identify from the sound that there were two frogs, one large and one small. Unfortunately, we cannot reveal the details of the research method. Dr. Pan's research depended solely on fundamental physics and math to extract the authentic sound of a tree frog, but while talking to our founding engineer, they quickly discovered that this research is also the perfect solution to a critical issue that headphones companies are desperately trying to solve.

Our lead engineer worked in the consumer electronic industries for 30 years, and he brought the cutting-edge technology to us while some of us were still engineering students and before we started the company. We brought our love for music and turned that into the passion to bring the best audio product to our fans.

So really, it all started about one and a half years ago, with the breakthrough research that created our Harmonic Interference Free (HIF) Technology. We saw the potential of this innovation to create the audio product of our dreams—headphones that deliver amazing sound quality without the hefty price tag—and started designing our first product that implements the HIF Technology. At the beginning of 2014, with the completion of our first product line, Impact, we officially launched Forte Audio.

We named our company Forte because we aspire to be the strong driving force of a revolution that

Forte's headphones (Impact, Life, and Life S) use Harmonic Interference Free (HIF) Technology, which is solely based on fundamental physics principles and advanced mathematical calculations to reduce and better eliminate harmonic interference.

demand affordable, high quality products in the headphone industry. This is a quest, and we have only just begun on this journey!

TRAVIS BLANE: Forte is really just in the beginning stages and is currently based in southern California. Currently, 10 of us work for Forte, but some of us work from Los Angeles and some of us work from San Diego.

SHANNON: Who are the founders and what are your roles and educational or professional background?

TRAVIS: Forte has three main founders. Scott Chen leads our manufacturing process, Angela Lai is head of creative design, and I am in charge of branding, sales, and marketing.

SCOTT: The core founders and members of Forte coincidentally all have an educational background in science or engineering, and some of us have been engineers for a few decades.

Forte Impact's modern, simple design offers a strong and smooth bass, combined great soundstage and mid range for powerful vocals. The metallic casing is the key for its signature solid and impactful sound and an excellent noise-isolating effect.

SHANNON: What makes Forte different?

SCOTT: Music is a beautiful thing that goes beyond any languages and boundaries. To experience the best of it, an audio product with supreme sound quality is essential.

We believe that superior audio quality should not be a reserved or locked away luxury. Forte is centered on this mission to make high-quality headphones available to everyone.

SHANNON: Describe the new harmonic interference technology that your company uses?

With Forte Life, the company's Harmonic Interference Free (HIF) Technology is wrapped in a minimalist plastic housing and available in six vibrant colors (white, black, blue, green, red, yellow).

SCOTT: Harmonic Interference Free (HIF) Technology was the result of some very cutting edge signal process research. Harmonic interference is the natural interference that occurs between sound waves of different frequencies, and audio tracks without interference processing will sound fuzzy and muddy. HIF Technology uses physics principles and calculations to reduce the harmonic interference between audio tracks.

As a result, the audio produced by our headphones are a lot more crisp and clear. Instead of approaching the issue by employing costly materials or multiple drivers, we went on a completely different route and attacked the interference from the root with a scientific and efficient solution, and ultimately avoiding putting the cost burden on the consumers.

Originally, the research was conducted to extract the sound of a forest frog from all other surrounding noises in a recording. Imagine all the work put into extracting that quack from the sounds of the wind, of a nearby waterfall, and the rustling of the bushes. That's the power of HIF Technology.

SHANNON: Tell us about the headphones you have designed.

SCOTT: Currently, we have two lines of headphones: Forte Impact and Forte Life.

Forte Impact, our first line of the Forte headphones, is a bass-enhanced model that's our signature powerhouse. It has a strong boost in bass and an extraordinary soundstage, giving you the impact you want from your rock, hip hop, and bass-heavy music.

Forte Life, our newest model, is designed to be versatile and fit into your daily life and musical needs. The sound profile is well balanced between all frequencies with a slight kick in the bass, making it your ideal companion if you listen to a variety of different music genres. It also comes in both metallic colors and bright colors to cater to our fan's stylistic needs.

TRAVIS: We make the only high-performance headphones you can buy at this price point ($40 to $70). You have to be different, especially considering all the competition that offer similar products to one another and battle it by offering sharp marketing. We give people the best audio quality headphones at the best price point period. Just wait for when we release our over-ear prototypes. They will blow away your Beats, Bose, Skullcandy, and Sennheiser.

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

SCOTT: The biggest challenge is to keep the price low for all the music lovers out there but not sacrificing the quality. Thanks to HIF Technology, we are able to effectively achieve clean and crisp sound without adding costly components, and create headphones with a big ticket audio quality but not a big price tag.

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

SCOTT: The Kickstarter community embraces new ideas and concepts, it demands high-quality products, and it is very supportive of innovations. We chose to run our campaign on Kickstarter because our mission is in line with the Kickstarter spirit, and we believe these headphones that we put so much effort in creating will not let the community down. Along the way, we have been very humbled by all the support and invaluable feedbacks we have received. The success has reassured us the demand for great sound quality at an affordable price is indeed high, and our effort to make our goal a reality is valid and recognized. The Kickstarter community has helped us take the most important first step of our quest to rediscover sound, and we couldn't be more grateful!

SHANNON: Has the product started shipping yet, and what kind of response has it received?

SCOTT: Forte Life, which was launched on Kickstarter, started shipping at the end of October. We are very proud of these headphones. We love them, and we are confident our backers will love them too!

With Forte Impact, we have received many positive feedbacks and reviews from professional DJs and Tech bloggers (such as Geoff Morrison from Forbes)! These reviews have helped us improve our products to provide the best to the world!

SHANNON: What's next for Forte?

SCOTT: We are on a quest to make more genre specific models to fit individual musical tastes! Our eventual goal is to create a complete lineup that seamlessly caters to all your diverse needs and promises a perfect listening experience. We are currently working on a model for acoustic and classic tracks. I can't say much about it now yet, but I'll say this without spoiling the surprise it's beautiful in looks and sound!

Forte Life S is designed to provide crisper sound and is available in champagne, silver, and gunpowder.

You Can DIY!

Arduino-Based Tube Power Amplifier Controller

In a nod to the Maker Movement, this project uses a common open-source microcontroller module, the Arduino, to build an automatic power controller for a tube power amplifier.

By
Mark Driedger
(United States)

I was fortunate to grow up in an era when it was easy to experiment with electronics. Consumer audio and TV products were a mix of vacuum tube and discrete transistors. It was possible to do a lot (and learn a lot) with a soldering iron, a voltmeter and, if you were lucky enough to have one, an analog scope. Construction was point-to-point or single-sided PCBs. If you did not have a schematic for something, you could usually trace one out with a pencil and paper.

Early computer technology was more complex for the DIY community. But, you could still build a computer from scratch. Technology quickly advanced, and we benefited from some incredible capabilities. However, it seemed that the DIY community suffered. We saw the demise of *Popular Electronics* (1999) and *Radio Electronics* (2003). Useful surplus electronics parts were harder to find. Thanks to multi-layer PCBs, surface-mount packages, and a lot of software, it became impossible to troubleshoot anything but the most basic faults in consumer electronics products. Schools spent more time on SPICE (Simulation Program with Integrated Circuit Emphasis) simulations than real world labs.

Recently, there has been a DIY resurgence described as the Maker Movement. This trend is fueled by technologies such as 3-D printing, robotics, social networks, open-source software, cheap sensors, and inexpensive microcontrollers.

With the Maker Movement in mind, I thought that I would use the Arduino—a common open-source microcontroller module—to build an automatic power controller for a tube power amplifier.

The resulting controller is very simple—a $25 Arduino module, a couple of dual op-amps, a few other components, and a 5-VDC power supply plus some opto-isolated switches (see **Photo 1**).

The controller detects an audio signal's presence and automatically powers on the amplifier, stepping through a timed sequencing of filament, driver, and output stage power supplies. It turns off the amplifier several minutes after the audio signal is lost, helping to increase the tube life and preventing unnecessary power cycling. The controller has an override switch to manually turn the amplifier off or on, independent of the audio signal.

How It Works

Figure 1 shows how the Arduino audio controller connects to the tube amplifier. The controller consists of an audio detector and a state machine. The state machine has three outputs that drive the switching circuits. The P0 signal switches AC to the amplifier's power transformer using an opto-isolated, zero switching relay. The P1 and P2 signals drive the opto-isolated MOSFET switches that I will describe later in the article. The P1 signal enables the driver, while P2 enables the output stage.

You do not need to use all three of the switches. If you have a solid-state amplifier just use P0. If you don't want to separately control the driver and output stages, then don't use P2.

Audio Detector

Figure 2 shows a block diagram of the audio detector. The left most portion is implemented in hardware, the remainder in software. The left and right audio signals are independently passed through precision half wave rectifiers. On the negative half cycle, the rectifier behaves like a normal inverting amplifier with gain –RA/RB except that the op-amp has to drive its output higher by the voltage drop of the diode.

When the input is positive, the op-amp output goes to ground and the diode is reverse biased. The op-amp and diode are effectively out of the signal path. There is some feed through of the input through RA and RB but without any gain. This isn't useful, but neither is it harmful for our purposes.

The rectifier outputs are sampled with the Arduino's built-in 10-bit ADC. The left and right channels are compared to a threshold and then logic OR'd, providing an instantaneous indication of whether there is an audio signal. The OR'd signal is passed through a digital low-pass filter and is again compared to a threshold to provide some averaging and protect against spurious noise.

You can adjust the circuit's sensitivity by changing the gain of the precision rectifiers or by changing Threshold A and Threshold B in software. Since we don't care about the signal's fidelity, it does not matter if the rectifier saturates on large input signals. The gain can be set fairly high to provide good sensitivity.

State Machine

Once we have an indication that audio is present, we use a software state machine to implement the power sequencing. When audio is detected, the P0 output is enabled, and we start a counter that waits 8 s before enabling P1 and then a further 1 s before enabling P2. Once audio has been absent for 120 s, we disable all three outputs. Manual override switches are provided. When in the manual off position, the amplifier instantly turns off regardless of the presence of audio. When in the manual on position,

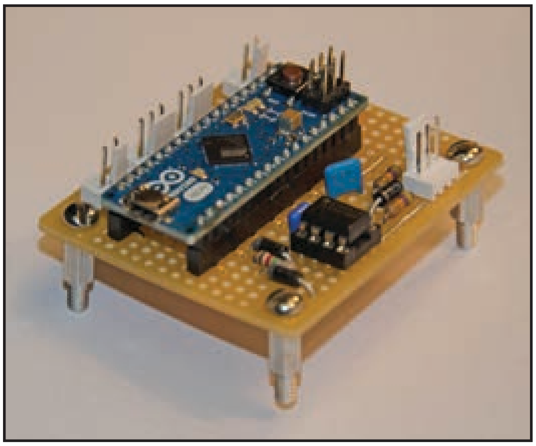

Photo 1: The Arduino audio controller is inexpensive and easy to assemble.

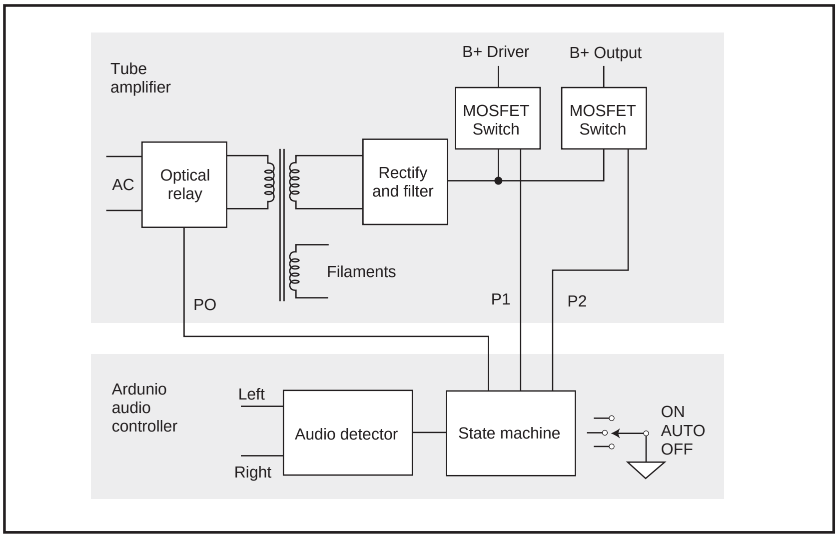

Figure 1: The block diagram details the Arduino audio controller's connection to the tube amplifier.

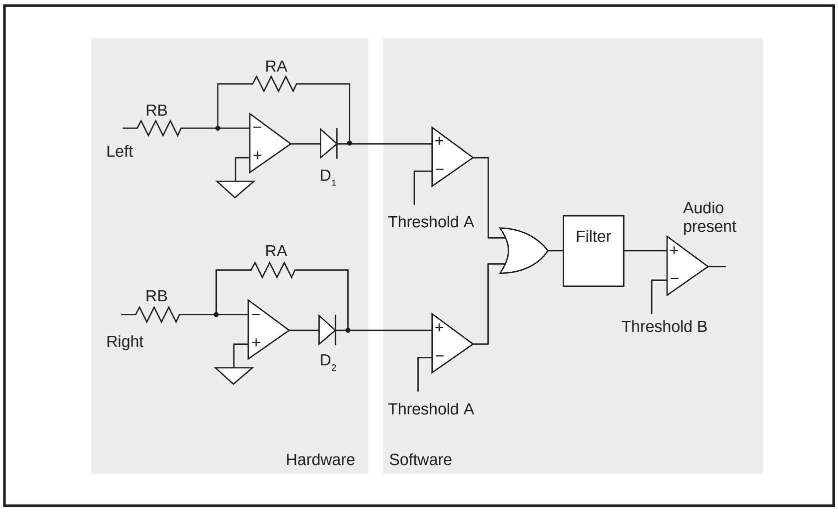

Figure 2: The audio detector's block diagram details the hardware and the software.

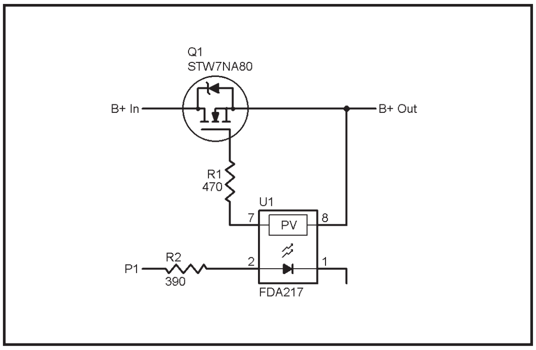

Figure 3: The MOSFET switching circuit is operated at 10-mA forward current through the 390-Ω series resistor, R2.

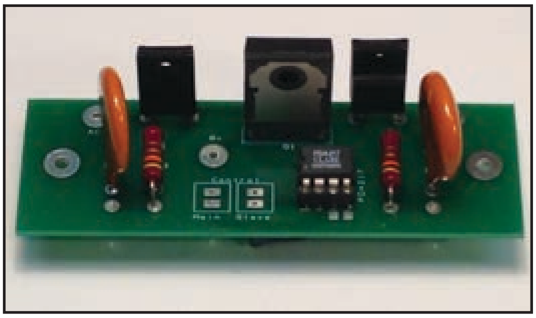

Photo 2: This design uses a FDA217 dual photovoltaic MOSFET driver. The MOSFET switching circuit is shown on a PCB with a B+ rectifier and a snubber.

the amplifier turns on and goes through the normal power-up sequencing.

Power Amplifier Control

To switch the AC power (P0), I used an opto-isolated "relay." It has a zero crossing feature that reduces turn on transients, which is especially important with toroidal transformers. The control input does not need an external current limiting resistor and is directly connected to the P0 Arduino output port. To switch the B+ supply, I used power MOSFETs with an opto-isolated driver.

When I started this project, I searched for a driver that was simpler than the discrete circuit I had previously designed. That yielded the FDA217 driver chip from IXYS. Quoting from its datasheet:

"The FDA217 is a dual photovoltaic MOSFET driver. Each independent driver consists of an LED that is optically coupled to a photodiode array. The driver output is controlled by means of the highly effective gallium-aluminum-arsenide (GaAlAs) infrared LED at the input. When the input current is applied to the LED, the light emitted activates the photodiode array, and generates the voltage at the output.

The photodiode array is capable of generating a floating power source with voltage and current sufficient to drive high-power MOSFET transistors. Each photodiode array contains an integrated turn-off circuit that discharges the external MOSFET gate when LED current is removed. This eliminates the need to use external components to facilitate the discharge. The optically coupled technology provides 3,750 V_{RMS} of input to output isolation."[1]

At $2.50 each, the FDA217 chips make extremely simple and cost effective drivers.

Without heatsinking, Q1 can safely pass a few hundred milliamps. The FDA217 is operated at 10-mA forward current through the 390-Ω series resistor, R2 (see **Figure 3**). The 470-Ω resistor R1 is a gate stopper and should be as close as possible to the gate of the MOSFET. The MOSFET is an N-channel enhancement mode device.

I used surplus STW7NA80s with an 800-V drain-to-source voltage rating. You can substitute other devices that are compatible with your amplifier's

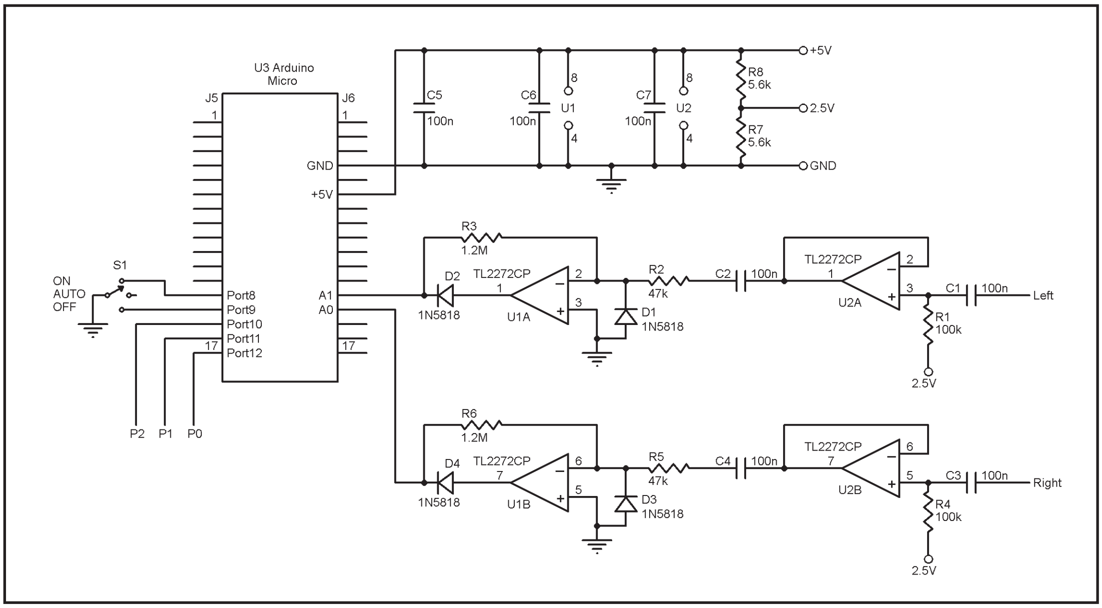

Figure 4: This is the schematic for the Arduino audio controller.


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Controller Parts List

Designation	Part Number	Description
U1, U2	TLC2272	rail-to-rail op-amp
U3		Arduino microcontroller
D1-D4	IN5818	Schottky diode
R1, R4		100 kΩ, 1/8-W resistor
R2, R5		47 kΩ, 1/8-W resistor
R3, R6		1.2 MΩ, 1/8-W resistor
R7, R8		5.6 kΩ, 1/8-W resistor
C1-C7		100 n
S1		3PST rotary switch or center off SP2T toggle switch

MOSFET Switch Parts List (per switch)

Designation	Description
Q1	ST W7NA80 N-channel MOSFET
U1	FDA217 opto-isolated MOSFET driver
R1	470R, 1/8-W resistor
R2	390R, 1/8-W resistor

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.

Project Files

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

Reference

[1] IXYS, Integrated Circuits Division, FDA217 Dual Photovolatic MOSFET Driver datasheet, www.ixysic.com/home/pdfs.nsf/www/FDA217.pdf/$file/FDA217.pdf.

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70S2-04-B Opto-isolated relay
Magnecraft (now Schneider Electric) | www.schneider-electric.com

TLC2272CP Op-amp
Texas Instruments, Inc. | www.ti.com

voltage and current needs. Follow normal high-voltage safety precautions when working with the B+ circuits. Power off and ensure all capacitors are safely discharged before working on any high-voltage circuit (see **Photo 2**).

Putting It Together

Figure 4 shows the schematic for the Arduino audio controller. You can buy the Arduino from multiple online sources. There are several poor quality copies for sale. I recommend using a reputable supplier and paying a couple more dollars. I used the Arduino Micro version for its small size (4.88 cm × 1.77 cm). I have provided pin numbers for the Micro but you can use any version. Power for the controller is provided by a small 5-VDC cellphone charger.

Op-amp U2 is an AC-coupled unity gain buffer. It isolates the amplifier audio input from the precision rectifier. The rectifier's input impedance dramatically varies for the positive and negative inputs, which could cause distortion if it was directly loading the audio input.

Op-amp U1 is the precision rectifier with a gain of roughly 25. Schottky diodes D1 and D3 provide input protection for the op-amp. The op-amps are rail-to-rail devices suitable for 5-V single-ended operation. To aid in debugging, the built-in LED on the Arduino board (port 13) is turned on when audio is detected, and a square wave is output on port 7 at the sample frequency (2 kHz).

Software

In addition to a soldering iron, you will need a computer to upload the software into the controller. Arduino has an excellent website and online forum if you need additional help (see Resources). Here are step-by-step instructions.

- Follow the instructions at http://arduino.cc/en/guide/windows to download the Arduino Integrated Development Environment (IDE) software. Then connect the Arduino to your computer and install the device drivers. On my machine, I needed to tell Windows 8 to disable driver signature checking to get the driver to properly install.
- In the IDE, under the Tools menu, select the model of board you are using. Then, choose the first entry from the Serial Port menu (you may need to repeat the serial port step each time you reconnect the Arduino to the PC).
- Download the "sketch" (software) for the audio controller from the Supplementary Materials section of the *audioXpress* website.

- Open the sketch using menu commands File, Open...
- Upload the sketch to the Arduino using menu command File, Upload
- Remove the USB cable and install the Arduino in the amplifier
- If you want to implement different delays then simply change the definitions in the "DelayP0", "DelayP1" and "DelayOff" constants before you do the File, Upload.

Taking It Further

The Arduino controller can easily be extended to other functions. In a *audioXpress* future article, I will explain how to add amplifier-over-temperature shutdown protection and how to incorporate a software version of my earlier bias and balance meter design. That article is expected to run in the April issue of *audioXpress*. The article will also include a PCB design for the controller.

Arduino Software Basics

The Arduino uses the C programming language. There are many good books, including the classic "K&R" reference (*The C Programming Language by* Brian Kernighan and Dennis Ritchie, Prentice-Hall, 1978) and many newer books and websites focused on C for Arduino. If you are familiar with other programming languages, you should be able to comprehend most of the code. Here are some pointers for a few of the less intuitive parts:

The ++ and - - operators increment or decrement the adjacent variable
Logical operators are OR (||), AND (&&) and NOT (!)

Each Arduino program ("sketch") consists of two main functions. The function setup() is performed once at power up. The function loop() is repeated thereafter. I've used the setup() function to program how I use the input and output pins and to initialize variables. In the loop() function, I first deal with the audio signal:

- Read the audio signal and implement the first comparator and OR function
- Pass through a digital single pole filter
- Then implement the second comparator to get the AudioPresent signal

You can see how a couple of lines of code replace a lot of analog circuitry.

Then, I implement the state machine. The state machine transitions between four different states (OFF, P0, P1, and P2) based on the condition of the inputs from the override switch (ManualOn, ManualOff) and the output from the audio detector (AudioPresent). Counters are used to implement the time delays, advancing once for each time through the loop() function.

Figure 1 shows the state machine with the lines illustrating the condition of the inputs and timers that cause it to transition from one state to another. You can learn more about state machines at http://en.wikipedia.org/wiki/Finite-state_machine.

At the end of the code, there is an extra delay to set the sample rate to about 2 kHz (500 usec each time through the loop() function).

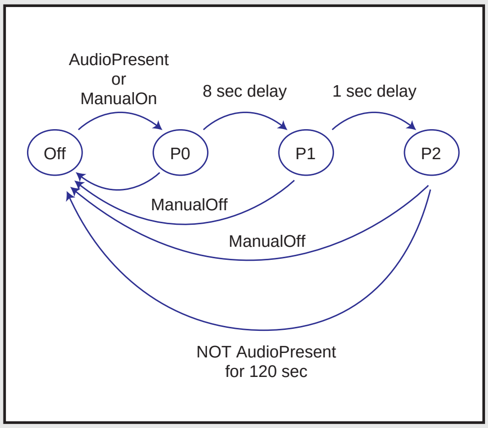


Figure 1: The state machine logic shows the condition of the inputs and timers that cause it to transition from one state to another.

Tube Microphones

Historical tube condenser microphones that are 30, 40, and even 50 years old still rank among the most coveted studio equipment. Anyone who owns such a jewel will not keep it in a glass cabinet. In times when technology is good and bad characteristics are made audible through high-resolution digital transmission, tube microphones are the measure of all things.

(Photo courtesy Neumann.Berlin—Bplus—Gruppe, Berlin)

By
Thomas Görne
(Germany)

Bringing the 1950s into the Digital Studio

Jazz from the 1950s and 1960s lives because of musicians such as Horace Silver, Herbie Hancock, Jimmy Smith, and Wayne Shorter. It also lives because of the legendary productions made at Blue Note, a label founded by Alfred Lion and Francis Wolff, featuring their audio engineer Rudy van Gelder.

In the summer of 1957, trained ophthalmologist Rudy van Gelder set up the Manhattan Towers studio in New York for a recording session with Hammond organist Jimmy Smith. Back then, van Gelder hardly had any equipment at his disposal other than a rustic mixing desk, an old tape recorder, and a few tube condenser microphones made by the Berlin-based manufacturer Neumann. And, he still recorded one of the best-sounding productions in Jazz history in just a few hours. In the opinion of many experts, the fabulous sound quality of these recordings—warm, transparent and clear—is only surpassed by the recordings van Gelder made in his living room, which he converted into a recording studio.

Music Production and Microphones: Then and Now

The sound quality of some of the recordings from that time is legendary. Paradoxically, thanks to lossless data memory (e.g., CDs and high-end loudspeakers) the beauty of such productions can still be appreciated today. So it is no wonder that after a period in which the transistor dominated the 1970s and 1980s, many audio engineers still considered tube microphones the measure of all things.

However, the evolution of studio microphones is remarkably short. In the 1920s, radio broadcasting fueled great innovations in audio technology. In 1923, Berlin engineer Eugen Reisz introduced his famous carbon microphone. Only five years later, Georg

Neumann, one of Reisz's employees, presented the first, and for a long time, only condenser microphone, the CMV3 produced in series.

The successor of this "Neumann bottle" marked the big bang in studio microphone technology—the Neumann U 47 (see **Photo 1**). The Neumann U 47 was built between 1949 and 1965, and still ranks among today's best and most expensive microphones (see **Photo 2**). The Austrian manufacturer AKG provided a worthy equivalent, the C 12, which was introduced in 1953. And, in 1960, Neumann consolidated its international reputation with models such as the M 49 and the U 67, which defined the standard of professional microphone technology (see **Photo 3**). Although the transistor microphone—built since the mid 1960s—features better technical data, only the sound matters in music production. And in this case, the tube devices are often clearly superior to their transistorized mates.

Photo 1: The Neumann U47 tube microphone's capsule is a double-gradient transducer.

The Condenser Microphone: Mechanics, Acoustics, and Electronics

The classic tube microphone is based on the capacitive converter in a low-frequency circuit (see **Figure 1**). The capsule consists of a massive, perforated brass electrode covered by a conductive membrane from both sides. In the early years, the membranes were made of gold-plated nitrocellulose. Later, they were made of gold-coated PVC and polyethylene, aluminum, and nickel. The membrane and the counter electrode together form a plate capacitor, the capacity of which is inversely proportional to the distance of the capacitor plates. At a typical membrane interval of some hundredths of a millimeter, there is an idle capacity of 50 to 100 pF.

The perforation of the electrode ensures that the sound reaches both sides of the membrane. The detour of the sound between the membrane's front and the back leads to a frequency-dependent sound pressure difference, which is the driving force for moving the membrane. Such a capsule is called a pressure-gradient transducer or pressure-difference transducer. With a suitable acoustical design, it has a cardioid and, therefore, unidirectional characteristic. The symmetrical capsule design featuring two membranes turns it into a double-gradient transducer. Its directional characteristic can be adjusted with the electrical summation of the partial signals of both capsule halves.

The microphone capsule is charged to a typical direct voltage of 60 to 200 V (capsule bias). A large charging resistance of 10^8 to 10^9 Ω ensures that the capsule circuit operates in quasi-idle mode. A deflection of the membrane leads to a change in capacity and, therefore, in voltage in proportion to the distance change. When the membrane oscillates, the direct voltage is overlaid by a low alternating voltage that can be tapped as signal. You only need to decouple the signal of the direct voltage offset of the capsule bias by a capacitor wired in series.

For minor membrane deflections, the capacitive transducer works in linear mode. The change in voltage over the capsule

Photo 2: Every studio that still owns a functioning U 47 looks after it as one would a valuable treasure. This legendary tube microphone's full and warm sound is still in very high demand. Frank Sinatra famously refused to sing at a studio without "his U 47." The Beatles also loved this microphone, a fact which is documented in many films recorded in the studios. Technically speaking, it was the world's first switchable capacitor microphone. In the history of Neumann microphones, it was the second crucial milestone after the invention of the "Neumann Bottle."

Photo 3: The Neumann M 149 tube microphone uses a capsule derived from the legendary U 47 and M 49 models.

About the Author

Thomas Görne studied acoustics and communication technology in Berlin. After graduating, he worked, among other things, as a movie sound engineer. He was also a professor of theory of music transmission at the College for Music in Detmold (Germany). He is currently a professor of audio design and audio systems at Hamburg University of Applied Sciences. His books *Mikrofone in Theorie* and *Praxis* (*Microphones in Theory and Practice*) and *Studiotechnik, Hintergrund und Praxiswissen* (*Studio Technology, Background and Practical Knowledge*) were published by Elektor. *Monitoring* was published by PPV. In spring 2006, the handbook *Tontechnik* (*Audio Engineering*) was published by Hanser.

is proportional to the incoming acoustic signal. In practice, there aren't any acoustic signals that would make the microphone capsule distort. Even when it gets loud, the condenser microphone's membrane would only move a little more than a few millionths of a millimeter!

Although the condenser microphone's high-impedance quasi-idle circuit is excellently suited for converting sound, it is not suited for signal transmission. The source impedance in the megohm-up-to-the-gigaohm range, together with the microphone cable's capacitance, would lead to substantial quality losses. Therefore, the capsule's output impedance needs to be adjusted to the input impedance of common mixing desks by means of an impedance converter inside the microphone.

Impedance converters are amplifier circuits with a voltage amplification of about 1, with a high current gain, a high input impedance, and a low output impedance. Tube microphones are designed with triodes or pentodes wired as triodes (e.g., the VF 14, the AC 701, or the EF 86) and as cathode follower (see **Photo 4** and **Photo 5**).

The input signal (i.e., the alternating voltage generated by the capsule) lies between the control grid and the anode. The output signal is tapped between cathode and anode. For interference-free transmission on a two-wire shielded cable, the signal is further balanced by means of a transformer inside the microphone. The operating voltages for the tube and the microphone capsule are generated by an external power supply and fed via a cable.

Tubes: Then and Now

In the golden years of tube technology, microphone manufacturers could draw on unlimited resources. Consequently, in the 1940s, there were thousands of different types of tubes from German, British, and American production alone! However, development ended just as quickly as it began following the introduction of the transistor. Since the end of the 1960s, there have not been any tubes manufactured in Western Europe. Tube manufacturing moved to countries in the Eastern bloc or completely stopped.

As a result of the change in consumer electronics from tubes to semiconductors, the range of tube types currently manufactured is limited. The tube types required for near-field technology are produced in series. Interesting tubes (e.g., the ECC 802 S and the ECC 803 S) are being re-issued. Furthermore, new

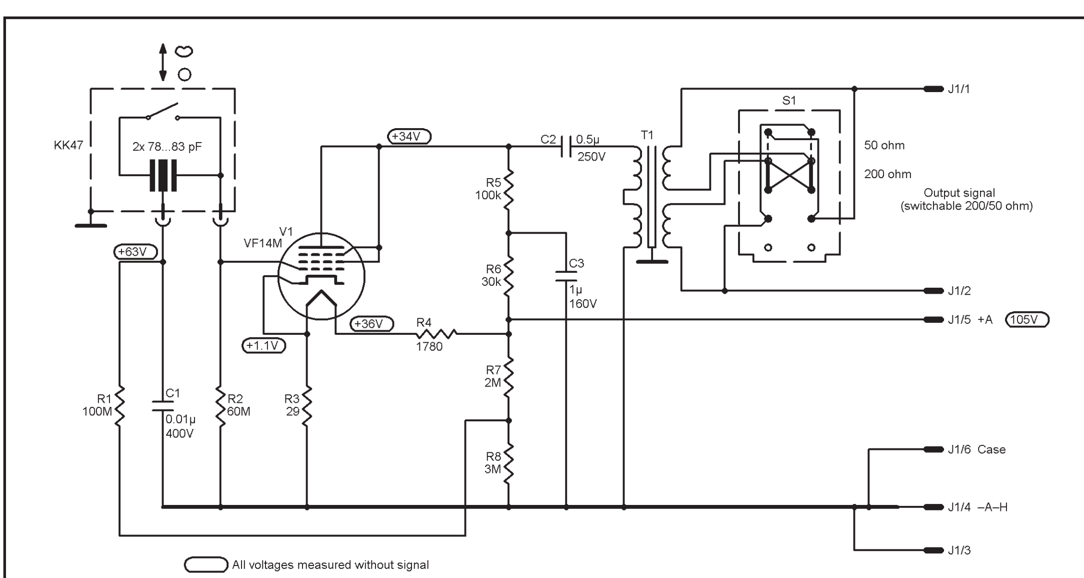


Figure 1: The classic tube microphone's impedance converter is shown with a pentode and as a cathode follower (Original schematic courtesy of Georg Neumann GmbH)

Harmonic Distortion

A nonlinear distortion can change the signal shape. In this way—according to the Fourier approach that states each periodic signal can be broken down to a sum of spectral components—new components are added to the signal spectrum. Depending on the frequency of these additional spectral portions, a distinction is made between harmonic and non-harmonic distortions.

In the case of the harmonic distortion, the new spectral components are integral multiples of the fundamental oscillation, thus harmonic harmonics. They are called harmonic distortion because of their odd sound. Possible causes for harmonic distortions are nonlinearities of the tube characteristics as well as large-signal overdrive of the tube and the transformer, which is driven to saturation because of high modulation. Depending on the cause of distortion, the signal is modulated into a rectangular or triangular shape. In the case of a triangular modulation, even harmonics are formed; whereas with a rectangular modulation, the harmonics are odd.

For total harmonic distortion (THD) k is used as a measure for a system's harmonic distortions. It specifies a percentage of the newly generated harmonics in relation to the total signal, in which all voltages are measured as effective value:

$$k = \frac{\sqrt{U_2^2 + U_3^3 + \ldots + U_n^2}}{\sqrt{U_1^2 + U_2^2 + U_3^2 + \ldots + U_n^2}} \times 100\%$$

Low harmonic distortions are hidden by the instrument's harmonic spectrum. It is clearly audible when the THD reaches approximately 2%. The counting of the harmonics starts with the second harmonic, because the fundamental oscillation is indexed as the first harmonic k1. If the effective value of only one harmonic is put in relation to the total signal, one obtains a partial harmonic distortion (e.g., k2 and k3).

K2 is the musical octave to the fundamental frequency. It is not easy to hear because it is also represented in the harmonic spectrum of most instruments. However, k3, being an octavated fifth, is clearly audible. A symmetrically clipping transistor amplifier evenly produces all odd distortion contents (e.g., k3, k5, k7, etc.). A good tube amplifier produces k2, k3, low k5, and virtually no higher harmonics when driven into saturation. This smooth saturation behavior enables to cut dynamic peaks, which makes the tube circuit act like a pumping dynamic compressor. As a result, one obtains slightly flattened and smoother dynamics, more drive, and subjectively more volume similar to an analog tape recorder.

Photo 4: The original Neumann U 67 tube microphone was manufactured in 1960.

Photo 5: The U 67's interior view shows its construction with an EF 86 pentode. (Photo sourced from Jonathan Low - http://heyjonlow.com/tube-talk)

Photo 6: The Brauner VM1 (1995) was the fulfillment of Dirk Brauner's dream to build the perfect sounding tube microphone. (Photo courtesy of Brauner Microphones)

types are being developed, produced in series, and put on the market. Previously common types (e.g., the VF 14 pentode used in the U 47 microphone) are very rare today and can cost thousands of dollars!

Virtually all modern triodes and pentodes come from tube factories in Russia, China, Serbia, and Slovakia. The operating parameters of some tube series from Asia are often high and the production quality is unsatisfactory. Only rarely do Telefunken, Siemens, or other former tube manufacturers produce tubes on the original machines. However, as with everything else in audio, you get good products for good money. Businesses that produce poor quality are quickly eliminated. Today, you can buy reliable products from good manufacturers.

Photo 7: The Brauner VM1 Klaus Heyne Edition (KHE) was a limited-edition version of the VM1 tube condenser, redesigned and upgraded by Klaus Heyne of German Masterworks. (Photo courtesy of Klaus Heyne)

Martin Schneider, a developer at Neumann, says: "The search for the suitable tube is an important part of development work. The tube should feature the lowest noise and the lowest vulnerability to mechanical rattle. Their transmission features must fit the special circuit design, and the tube must be available in sufficient quantity. The high price of modern high-quality tube microphones is not least caused by the strict selection of the tubes. Only special tubes of an ongoing production satisfy our requirements."

The godfather of tube microphones, Dirk Brauner, decided to go another, less expensive way. The tubes, which are specifically tailored to his requirements are developed and manufactured at the Slovak manufacturer JJ Electronic with the assistance of the German company BTB (see **Photo 6** and **Photo 7**).

About the Sound

There are three decisive elements for the sound of a tube condenser microphone: the capsule, the impedance converter, and the output transformer. AKG and Microtech Gefell, which emerged from the "Elektronisches Laboratorium Georg Neumann Gefell im Vogtland" (Electronic Laboratory Georg Neumann Gefell in Vogtland), offer new products as well as reissues of former tube microphones. The capsule and the circuit correspond to those of their classic models.

The redevelopments by Brauner are fully designed in the tradition of the former Neumann microphones, featuring Spartan circuits and output transformers. Although Neumann builds capsules according to specifications from 1940s, the company uses modern circuits and replaces the output transformers by electronic balancing using a transistor circuit. Apparently, this is not detrimental to the famous tube sound. In the meantime, when it comes to the most

Good and Expensive

Tube microphones are the most expensive studio microphones. One can easily spend $5,000 to $6,000 for a modern microphone or a reissue of an old model from a top manufacturer. Well preserved originals from the 1950s or 1960s may possibly be even more expensive. Copies of such microphones that are sold under different brand names usually originate from China. They are available at a fraction of the price.

modern developments, the measuring devices are left in the cabinet. In terms of sound, the tube circuit is consciously modeled according to the classic models and not according to measurably optimal ones.

Today, new tube microphones from top manufacturers such as AKG, Neumann, Brauner, and Microtech Gefell rank among the most desirable studio equipment.

Microtech Gefell has continued to manufacture tube microphones and re-issued the classic 1950s "mini bottle" CMV563/M7, which won a "best engineering" award in 2014.

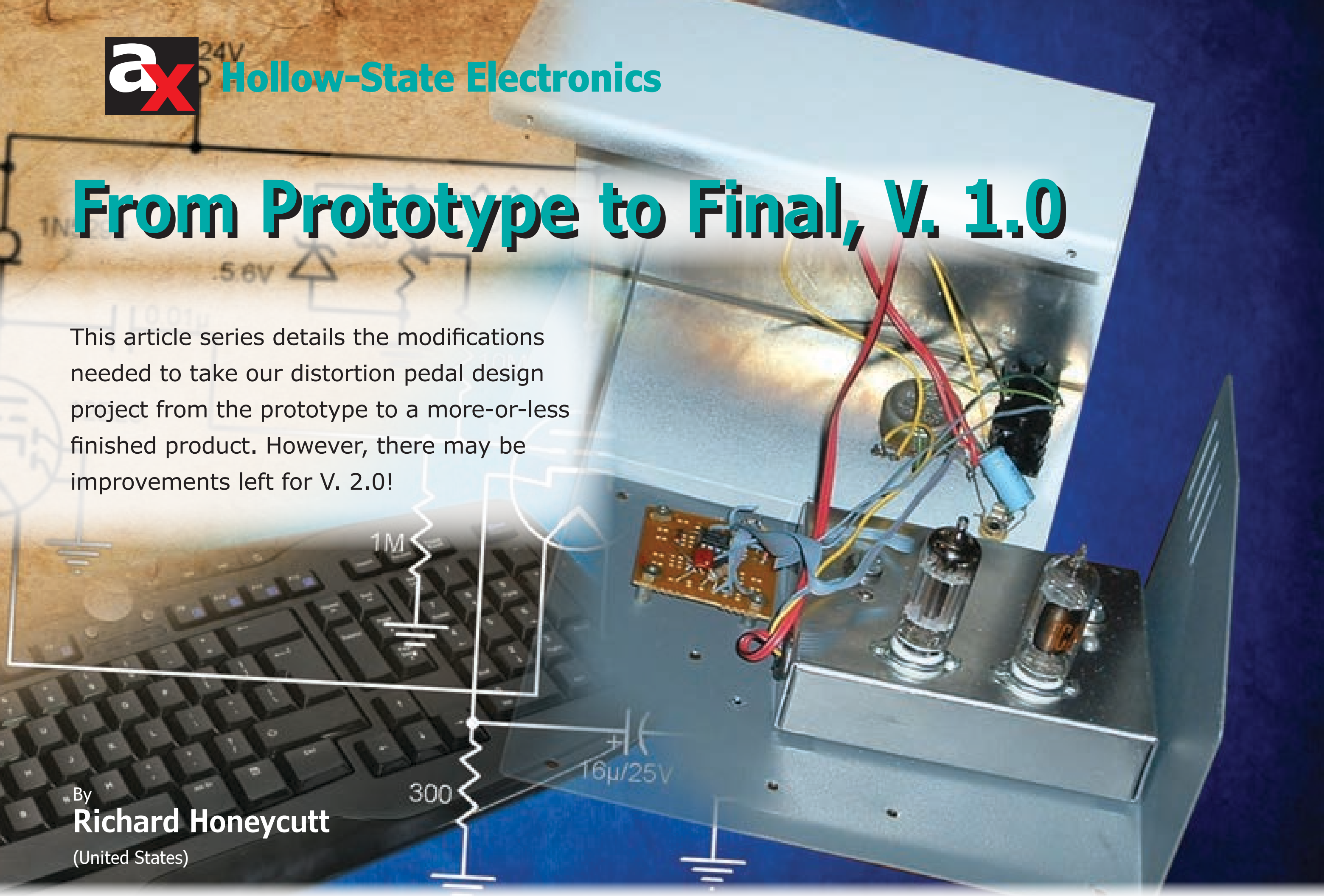
However, historical microphones remain popular, but not just with collectors. Following the lead of van Gelders and Blue Note productions, a significant proportion of pop music is recorded with original Neumann and AKG microphones. Even today, the U 47 and the C 12 (more than 50 years later) are among the most used studio microphones worldwide.

Some great music recorded with these original microphones include Jimmy Smith's "The Sermon," Horace Silver Quintet's "Song for My Father," Wayne Shorter's "Adam's Apple," and all of the Blue Note productions made from 1957–1966.

Transformers

A transformer consists of a ferrite rod made up of sheets, which feature two coils that are electrically insulated from each other. Both coils are penetrated by the same magnetic flux. If an alternating current flows through the primary coil, the magnetic circuit induces a corresponding alternating voltage in the secondary coil. By varying the turns ratio, the voltage on the secondary side can be increased or reduced compared to the primary side. The transformer electrically acts as a series inductance and as a first-order low-pass filter. At low frequency, it also has a limiting effect because of the principle of inductive coupling. A subtle change of the amplitude frequency response is a characteristic feature of many transformers. Above 100 Hz, it slightly increases until it reaches an increase of 3 to 4 dB at 10 kHz. This increase in amplitude is linear. Therefore, the phase response is not affected. The result is a characteristic "brilliant" sound.

The inputs and outputs of classic studio tube equipment are always equipped with transformers. The transformer ensures balancing of the signal to be able to use the interference-free two-wire shielded cables, as well as level and impedance matching. Since the introduction of transistor technology, these tasks can also be managed using inexpensive op-amps used in semiconductor technology. The transformers can be eliminated since they often cost $70 or more. But even though transformers are expensive and costly to manufacture, they provide a well-balanced signal and don't need a power supply. Above all, they ensure the galvanic isolation of the devices interconnected with cables. On the other hand, semiconductor technology can reduce distortions in the system. Furthermore, transistor circuits can better transmit low frequency. Transformerless or so-called "ironless" condenser microphones deliver noticeably more bass.



From Prototype to Final, V. 1.0

This article series details the modifications needed to take our distortion pedal design project from the prototype to a more-or-less finished product. However, there may be improvements left for V. 2.0!

By
Richard Honeycutt
(United States)

While I was visiting the development offices of a well-known software company, my host shared a great story with me. We were passing by the refrigerator in the employees' lounge, and he said, "Well, they finally got it working." I looked questioningly at him, and he told me that several months earlier, the refrigerator had stopped working. The next day, a hand-scribbled note written in Magic Marker was affixed to the refrigerator, with the words, "This is broken." A week afterward, a new note was added: "This is still broken." After a month had passed, a neat new note appeared: "We'll fix it in v. 2.0."

In the past few articles in this series, we have discussed taking a distortion pedal design from concept to prototype. It seems that any modifications to our project, can be done in our own V. 2.0. **Figure 1** shows the circuit after we made the prototype modifications. (A few minor corrections are included.)

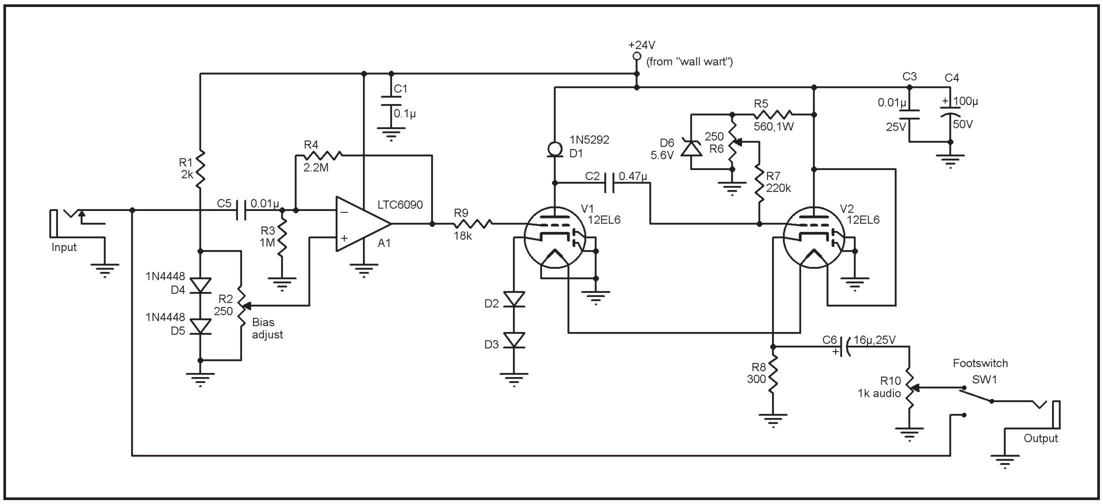


Figure 1: This schematic shows the starting point for our final tweaks.

Modifying the Circuit

In preparing to build the circuit, I found that the LTC6090 IC is not easily available in experimenter quantities. Since the input resistance of the op-amp stage must be about 1 MΩ, an op-amp is needed that has very low bias current, offset current, and offset voltage. Also, the op-amp must have very low noise. The OP27 meets these requirements and is readily available in small quantities. The LTC6090 boasted rail-to-rail input and output voltage ranges; whereas for a 2-kΩ or higher load, the OP27 has a typical span of 1.5 V (maximum 3.5 V) between maximum output voltage and the supply rail. This requires a change in our thinking about how to use the op-amp to set the control-grid bias of V1. Other low-bias-current, rail-to-rail op-amps are manufactured, but I did not find any that are easily available in small quantities. Thus, I selected the OP27.

The op-amp's non-inverting input (shown in the circuit in **Figure 1**) is fed by a DC voltage that can be varied from zero to about 1.2 V. Since this voltage sets the op-amp's quiescent DC output voltage, the same voltage appears at the output.

Using an OP27, we need a quiescent DC output voltage at least 3.5 V more positive than the negative supply (which is 0 VDC in this single-ended-supply configuration). We also need to allow about 1 V for the output voltage swing to avoid clipping in the op-amp stage. Then, the control grid bias needs to be adjustable from about 0 to −0.2 V. Of course, with a single supply, we cannot produce a negative voltage, so we'll have to make the cathode somewhat positive.

In the original circuit, we used two diodes in series between cathode and ground to place the cathode about 1.2 V above ground. Thus, a 1-V control-grid voltage gives us an equivalent bias of −0.2 V. But we can't drive the OP27's quiescent output voltage that low.

Since the forward voltage drop of a semiconductor diode is not well-defined (0.6 V is the approximate value), I experimentally found that by using only one diode between the cathode and the ground of V1 and making the adjustment range of the non-inverting input to the op-amp span from 0.387 to 5.6 V, I could obtain the desired range of control-grid bias. This led me to change the circuit of the op-amp stage to the one shown in **Figure 2**. R1, R2, R3, and D4 give the required adjustment

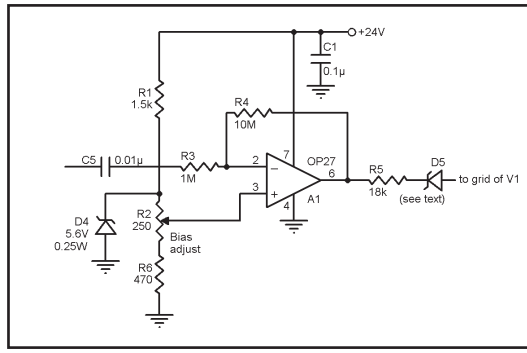


Figure 2: Using an OP27, we must change to this design for the op-amp stage.

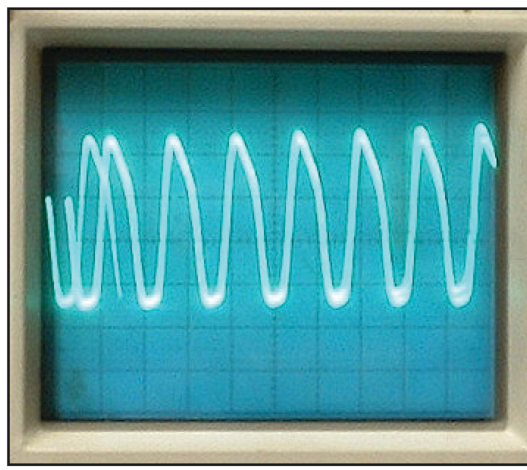


Photo 1: This is the output wave of the distortion pedal after adjustment.

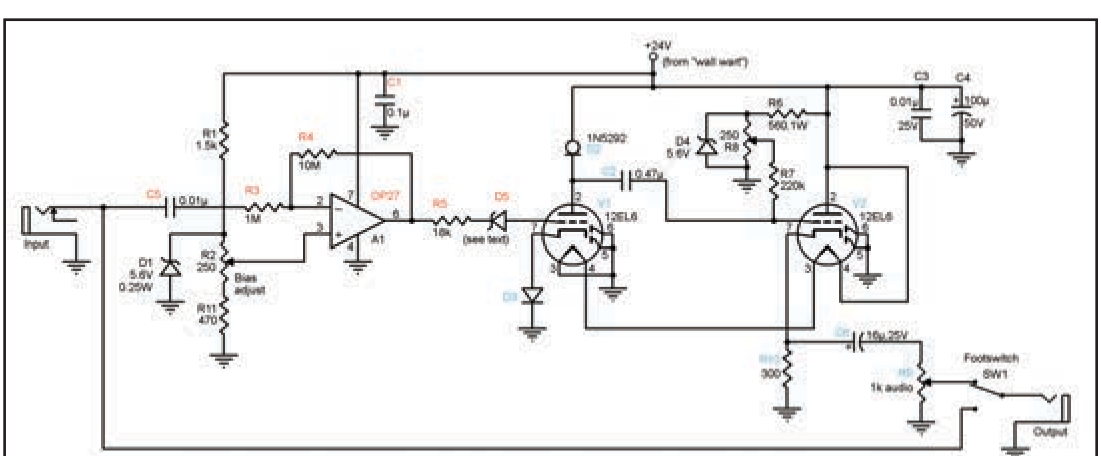


Figure 3: The final schematic uses an OP27 op-amp.

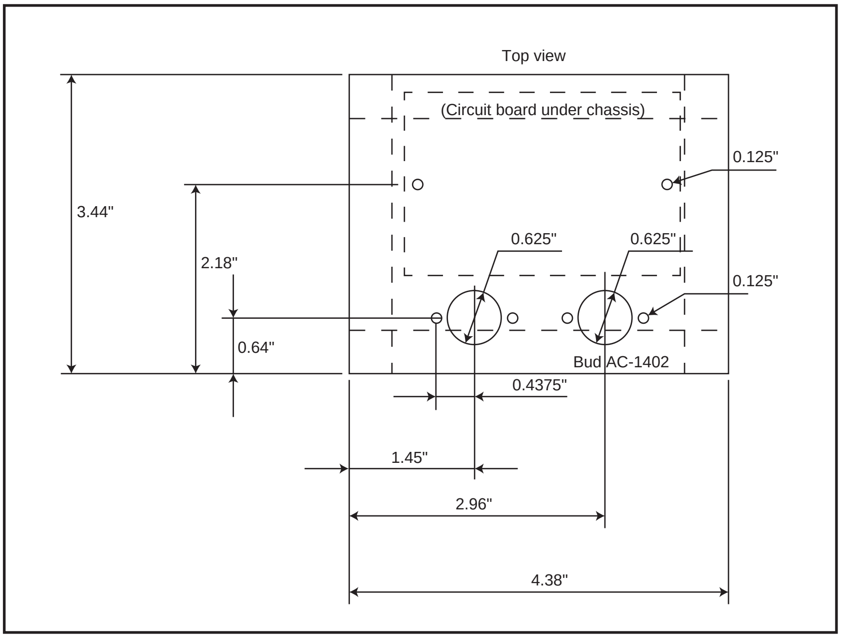


Figure 4: This is the layout for the chassis with tube sockets and other circuitry.

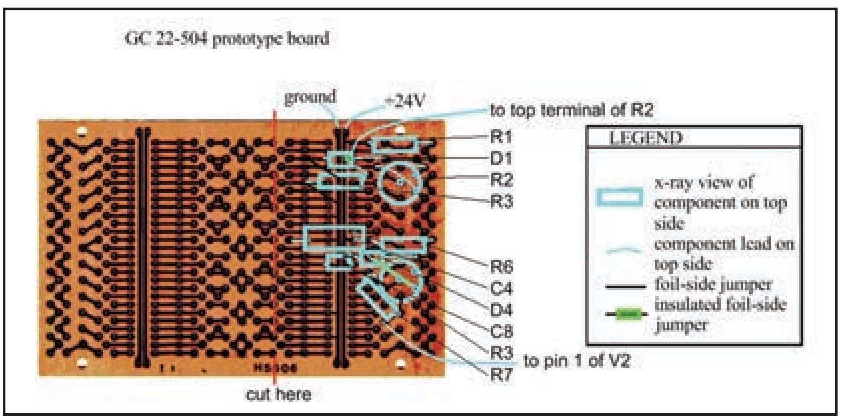


Photo 2: This is a foil-side view of the main circuit board.

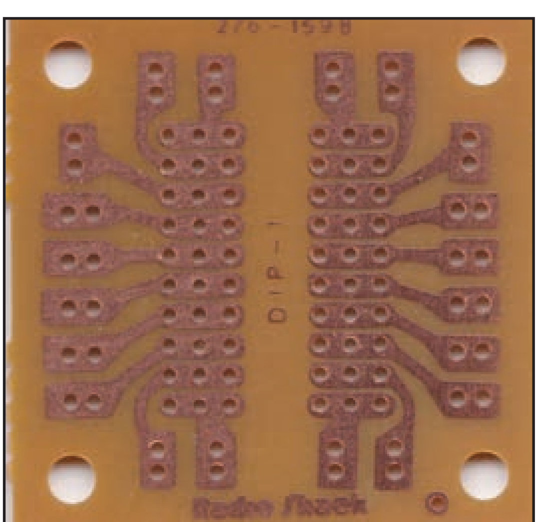


Photo 3: The op-amp and related components are mounted on this board.

range, while D5 acts as a 3.1-V level shifter to bring the DC level down to the necessary range for proper grid bias. I used a Zener diode, but to get the proper level shift at the low current provided by the triode, I had to use one specified at 4.5 V to get a 3.1-V level shift. I could have used a potentiometer to make this adjustment, but that would have also affected the AC level, which I didn't want.

Input Resistance

Knowing from the earlier "paper-to-prototype" work that the input resistance of the cathode follower would load the V1 stage, I examined the output waveform from V1. I found that it required a larger signal than a typical humbucking pickup would produce in order to create the distortion signature I wanted, so I increased the op-amp gain by changing feedback resistor R4 to 10 MΩ. Then, I verified that the op-amp could not be driven into clipping until the triode was already in hard clipping, so that our circuit would provide a tube sound.

Choosing the Power Supply

Figure 3 shows the final schematic. The component designators (R1, D1, etc.) shown in black represent components mounted on the main circuit board. Designators shown in blue represent point-to-point wiring on the tube sockets and/or terminal strip. Designators shown in red represent components mounted on the op-amp circuit board.

When building this circuit, one must remember that the type of power supply is critical. We have previously mentioned using a 24-V wall-wart. However, there are caveats to keep in mind. First, switching supplies can create enormous difficulties in filtering the direct current enough to avoid having buzz in the output. Second, some 24-V wall-warts are designed mainly for battery charging, and have little, if any, internal filtering. With a normal supply, this makes for a lot of hum. With a thyristor-regulated supply, it creates an awful buzz.

These problems are less critical for pedal circuits built from op-amps using split supplies, since the power-supply rejection of an op-amp is much better than that of a tube circuit. If you can't find a suitable wall-wart, one option is to buy one rated higher than 24 V and build a 24-V three-terminal regulator circuit into the pedal chassis. As long as the loaded voltage from the wall-wart does not drop below about 26 V, the regulator will provide excellent active filtering. An LM7824 three-terminal regulator will do the job.

Connecting the Pedal

After completing the construction, including

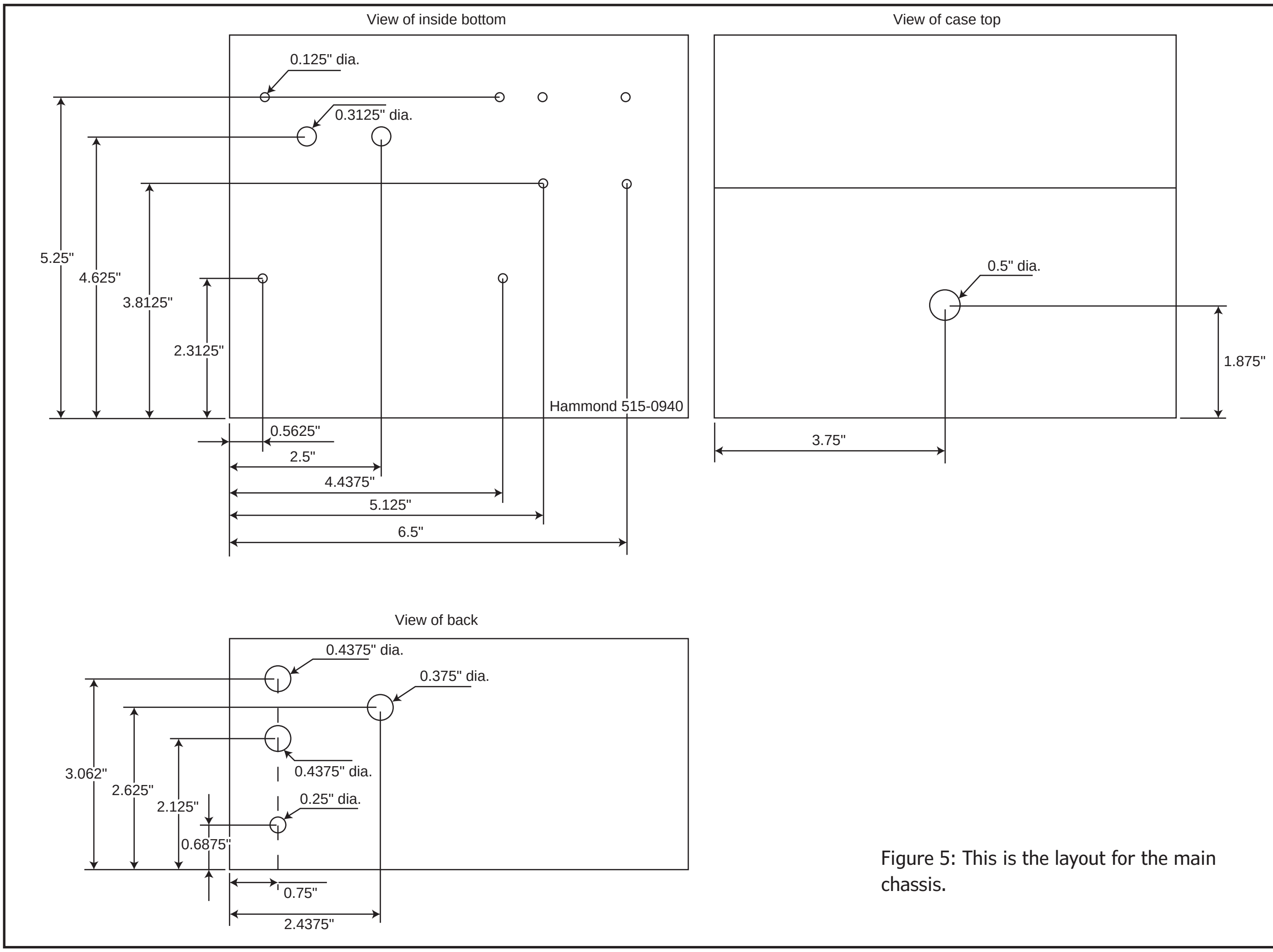


Figure 5: This is the layout for the main chassis.

leaving holes through which the bias adjustment potentiometers for V1 and V2 can be adjusted, I connected the pedal between a dual-humbucker guitar and an amplifier and adjusted the bias for the best (to my ears) distortion sound. **Photo 1** shows the resulting waveform using those settings, with a sine-wave input. (Ignore the double trace at the left-hand side; it is a triggering artifact of the oscilloscope.)

The Chassis

The project was built on a Bud AC-1402 chassis laid out as shown in **Figure 4**, with a circuit board for the tube-circuit components mounted inside the chassis, using 4-40 screws and nuts, with 0.5" spacers

Photo 2 shows the circuit board's layout with the tube-related parts. A GC22-504 board was used and cut as shown. I did not show a separate layout for the op-amp circuit board. I used a RadioShack board. **Photo 3** shows the foil side.

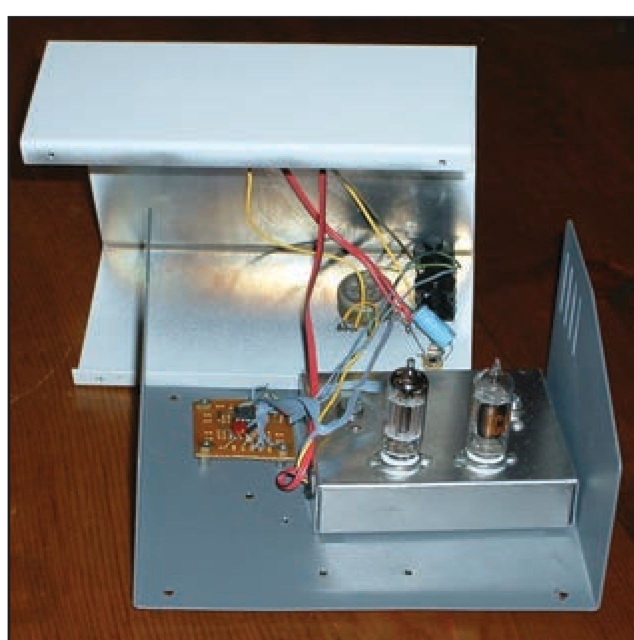


Photo 4: This is the inside of the handmade stomp box.

a) b)

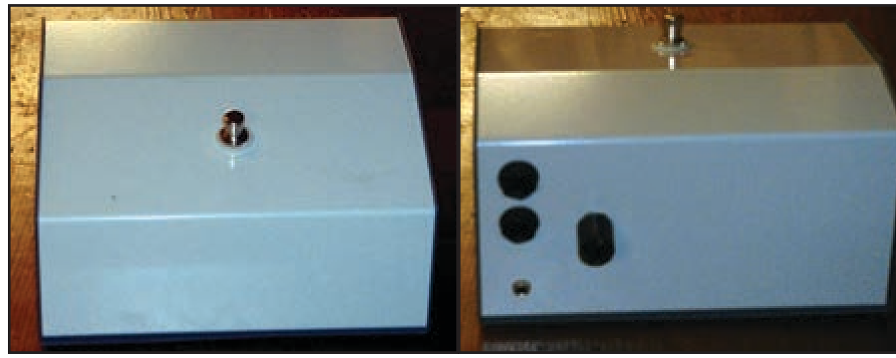


Photo 5: The top view (a) and the back view (b) of the completed project are shown.

Figure 5 shows the main chassis's layout. On the back view, the bottom hole is used for mounting the connector for the +24-V supply. The middle one on the left is for the input jack. I used a nylon-body jack. If you use a traditional metal jack, the hole diameter will need to be 0.375".

The top hole is for the output jack. The hole shown to the right of these jacks is for the distortion level potentiometer (R9). In the drawing of the chassis inside bottom, two 0.3125"-diameter holes near the top left allow distortion adjustments via the screwdriver-adjusted bias potentiometers.

Photo 4 shows the inside of the stomp box. **Photo 5** shows the top and back of the completed project. Possible improvements could include adding labels for the input, output, and power jacks, and perhaps adding a clever name to the top, using dry-transfer lettering.

The Stomp Box

To recap, we have now designed and built a "stomp box" that provides true tube distortion for an electric guitar. The distortion signature can be controlled by means of screwdriver adjustments accessible through holes in the bottom of the chassis. A 1-MΩ input impedance is provided by an OP27 input stage, configured so that the tube distortion occurs with input signal amplitudes in the op-amp's "clean" range. The bias potentiometer that controls the cathode follower (V2) stage will have little effect for most guitars but will have a clearly audible effect if a preamp (such as an Electro-Harmonix LPB-1) is used to boost the input signal to the stomp box.

You may want to experiment with different gains for the op-amp circuit. The gain of that circuit is given by:

$$A_V = \frac{R_f}{R_i} = \frac{R_4}{R_3}$$

More gain will increase the distortion, but too much gain can drive the op-amp into clipping, giving you a buzzy solid-state distortion rather than tube distortion.

Sources

AC-1402 Chassis
Bud Industries | www.budind.com

LBP-1 Circuit
Electro-Harmonix | www.ehx.com

GC22-504 PCB
GC Electronics | www.gcelectronics.com

515 0940 Case
Hammond Manufacturing | www.hammondmfg.com

OP27 Op-amp
Texas Instruments, Inc. | www.ti.com

Member Profile

Member Name:
Bill Reeve

Location:
Santa Clara, CA

Education:
BA in English and Geology, Whitman College, WA; MS in Geology, Colorado School of Mines, Golden, CO; MS in Mechanical Engineering, University of California, Berkeley; and MS in Electrical Engineering, Santa Clara University, Santa Clara, CA

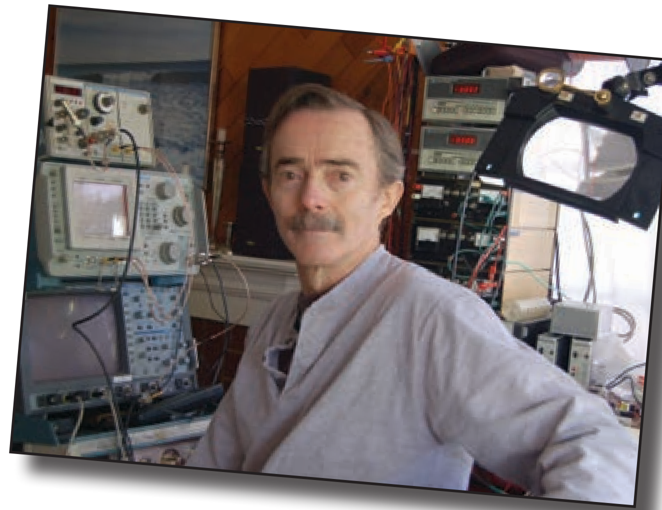
Occupation:
Bill is a technical program manager for Google. Prior to that, he worked as program director at Lockheed Martin and retired from that position.

Member Status:
Bill has subscribed to *audioXpress* for at least 10 years, possible 15 years, but he is uncertain of the exact timeframe.

Affiliations:
Bill is involved with the Society of Automotive Engineers (SAE).

Audio Interests:
He said musically he enjoys Peppino D'Agostino and most recently, Icelandic singer-songwriter Hera Hjartardottir.

Most Recent Audio Purchase:
His most recent purchase was a Tektronix TR503 tracking generator for a spectrum analyzer.



Current Audio Projects:
Bill's recent projects involve designing and building a small desktop and headphone tube stereo amplifier and restoring a 1960 Harman Kardon Trio A244 tube stereo amplifier.

Dream System:
Bill said his dream system would definitely include a restored 1960s Garrard turntable that he completed on his own and a 1970 McIntosh 240 amplifier.

Industry Calendar

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

February 10–12, 2015 — **Integrated Systems Europe (ISE) 2015**
Amsterdam RAI, The Netherlands
www.iseurope.org

Currently one the most successful and fastest growing shows in the world, Integrated Systems Europe (ISE) is promoted jointly by InfoComm and the Custom Electronic Design and Installation Association (CEDIA). This European event is becoming stronger every year, focusing on Audio over IP (AoIP), wireless audio, immersive installations, and the latest in electroacoustic solutions. Due to the convergence of consumer and professional technologies in the electronic systems integration markets and the worldwide expansion of the installation market, ISE has become the largest showcase for Internet Protocol (IP)-based technology solutions, home networks, and building management. The organizers expect a record attendance for 2015, with more than 50,000 attendees and more than 1,000 exhibitors.

Prior to the show—Monday, February 9—the expanded Smart Building Conference will feature parallel tracks for residential and commercial building applications, with full-day workshops.

April 5–8, 2015 — **Prolight+Sound Guangzhou 2015**
China Import and Export Fair Complex, Guangzhou, China
www.prolightsound-guangzhou.com

Benefitting from the alliance between Messe Frankfurt and the Guangdong International Science & Technology Exhibition Company (STE), Prolight+Sound Guangzhou is now China's largest pro audio and lighting show and takes place in Guangzhou, which is China's pro audio and lighting manufacturing hub.

Backed by Messe Frankfurt's global resources and STE's extensive local knowledge, the 2014 show attracted 60,068 visitors from 105 countries. The 2014 show showcased 1,138 professional audio, lighting, and stage equipment suppliers, occupying more than 110,000 m^2 of exhibition space.

Since its inauguration in 2003, Prolight+Sound Guangzhou (formerly known as Sound Light Guangzhou) has earned a reputation as the largest event for professional audio and lighting, providing trade visitors with plenty of manufacturer resources and networking opportunities.

Messe Frankfurt also promotes the Prolight+Sound Shanghai, the International Exhibition for Event and Communication Technology, and the AV-Production and Entertainment (October 14–17, 2015) at the Shanghai New International Expo Centre (SNIEC), Shanghai, China.