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The Oldest Tool in the Modern Acoustician's Tool Box

You Can DIY!
The Cathedrals Speakers
By Ken Bird

elysia xfilter 500
Surface-Mount Magic in a Small Format

Audio Electronics
Build a Sound Level Meter and Spectrum Analyzer
By Ron Tipton

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Contents

Features

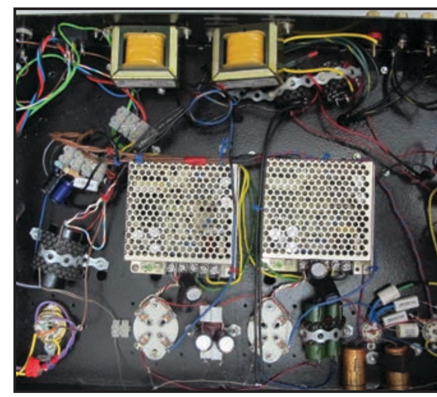
24 elysia xfilter 500
Surface-Mount Magic in a Small Format
By Miguel Marques
The xfilter 500 is an EQ in a 500-series format that offers a precise stereo image based on computer-selected, stepped potentiometers and low tolerance film capacitors.

44 The Cathedrals
By Ken Bird
This article details the construction of a pair of speaker enclosures with a gothic architectural theme. The enclosures were built to house four 8” Peerless woofers, a 4” midrange, and a 1” dome tweeter.

50 Tips to Resurrect a Classic Speaker or Design a New System (Part 2)
A Viable Solution to Speaker Sensitivity Problems
By Thomas Perazella
The second in a three-part series, this article describes how to repurpose vintage speakers and provides a solution to a common sensitivity issue.

56 Build a Sound Level Meter and Spectrum Analyzer
By Ron Tipton
The Model 527 is built using a Velleman audio analyzer kit. The instrument can be used to measure sound pressure level in noise pollution studies, especially for industrial, environmental, and aircraft noise.

Volume 44 – No. 12 December 2013



Columns

STANDARDS REVIEW
14 **Audio for Mobile Platforms: The New DAWs?**
From Audiobus to Apple's Inter-App Audio
By João Martins

PRACTICAL T&M
18 **Designing for Ultra-Low THD+N (Part 2)**
Resistors and Capacitors
By Bruce Hofer

SPEAKERS
31 **The Lowdown on Woofers, Subwoofers, and Bass Shakers (Part 2)**
Reinventing Low-Frequency Devices to Fit Compact Sizes
By Mike Klasco and Steve Tatarunis

QUESTIONS & ANSWERS
34 **DIY Audio Appeals to Applications Engineer**
By Shannon Becker

SOUND CONTROL
38 **The Oldest Tool in the Modern Acoustician's Toolbox**
By Richard Honeycutt

Departments

6 **From the Editor's Desk**
7 **Client Index**
8 **What's News**
63 **Member Profile**
64 **Industry Calendar**

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From Broadcast to Home Recording to Digital Networks—Where the New *audioXpress* is Going

For readers seeing this "second" issue of *audioXpress* since we introduced our new format and layout last month, I feel I should explain the concept a little more. Our target deadline for this relaunch was decided some time ago and I couldn't think of a better place to introduce our "new" magazine than the AES convention in New York City!

I can summarize our concept with a few words: more (of what our readers expect), electronics (our roots), and audio innovation (our focus).

We are proud of our heritage as *Audio Amateur*, *Audio Electronics*, *Glass Audio*, and *Speaker Builder* magazines. Those titles were born in a time when amateur radio was still developing hand in hand with electronics and radio technology. And that is precisely why *audioXpress* is a part of the electronics publication portfolio of Elektor International Media (EIM).

But you may be wondering about *audioXpress*'s evolution and what to expect in the future.

It's important to clarify that we will not continue to be a "home electronics" or consumer application-focused publication. We believe we should share the most interesting audio stories in the industry, independent of their application areas—consumer or professional, music or broadcast oriented. Hence, the innovation focus.

The most important consumer technologies often start with those developed for professionals. So, we will follow audio electronics innovations, together with the all-important disciplines of electroacoustics (and, needless to say, software, digital audio, networking protocols, and audio synthesis).

We believe that a publication such as *audioXpress* cannot focus only on the "home approach," which is still appeals to many enthusiasts and hobbyists. Some of us clearly remember the 1960s, when live concerts used "consumer" amps and speakers, before there were guitar amps and large speakers. At Woodstock, there were McIntosh amps (now a purely home audio brand) and the PAs were early versions of the JBL speakers (today both a pro and a consumer brand). Five years later, all the big "pro audio" brands in live sound, such as Electro-Voice and JBL dominated that market (in the US at least). During this time, things were different in the recording studios. There, technology was first "borrowed" from radio and TV broadcast. This is long before we had "home studios" using computers—and where exactly did that come from?

In the era of the Internet, blogs, and social networks, many magazines have disappeared. But, we know a magazine can flourish. In addition to its content and its readers, a magazine must also have a purpose. It must provide a sense of community. More importantly, it needs to offer readers content they can't find elsewhere. It does not matter if our readers are professionals, students, or enthusiasts. Our common interest unites us, whatever the platform: print, online, Facebook, Twitter, e-mail newsletters, or mobile apps.

We want to build a better *audioXpress* with more content, representing the common interests of the audio community, while also reflecting the industry.

João Martins
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All Electronics Corp.	63	Mundorf EB GmbH	29
Analog Emporium	33	OPPO Digital, Inc.	11
Audio Precision	47	Panasonic Industrial Devices	59
Avel Lindberg, Inc.	59	Parts Connexion	2
Crown International	53	Parts Express	68
Front Panel Express	21	Silicon Ray	43
Hammond Manufacturing Co.	3	Solen Electronique, Inc.	23
Home Theater Shack	49		
KAB Electro-Acoustics	63		

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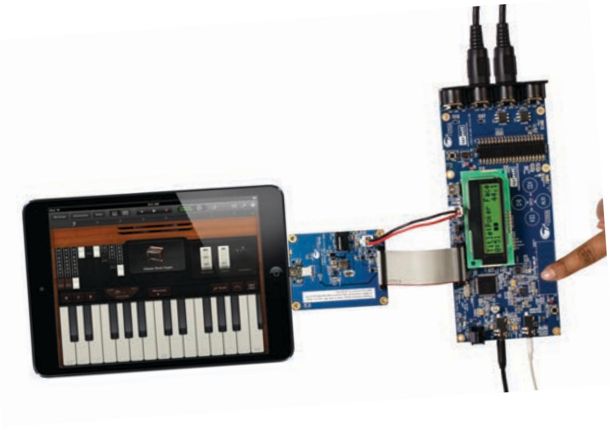
COLUMNISTS

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.



Cypress's PSoC 3 MFi digital audio solution manages the interface between audio accessories and any iOS device via Apple's connectors, now updated to Lightning.

Cypress Development Kit for iOS Devices

Cypress Semiconductor announced the availability of a new development kit that speeds design of digital audio accessories for Apple's iOS products. The CY8CKIT-033A PSoC 3 MFi Digital Audio Development Kit for Lightning includes license-free reference hardware, firmware, and iOS app software, providing the tools to develop Made for iPod/iPhone/iPad (MFi) digital audio accessories that are fully compatible with Apple's all-digital Lightning connector. The connector is used for iPhone 5, iPad mini, and other new Apple products. The comprehensive reference design enables the quickest path for MFi program licensees to develop consumer products, such as speaker docks, headphones and game controllers, and music creation devices including: keyboards, mixers, guitar interfaces, microphones, Musical Instrument Digital Interface (MIDI) controllers, and DJ turntables.

Cypress's reference design connects PSoC 3 to Apple devices with Apple's Lightning and 30-pin dock connectors, using Cypress's patent-pending USB audio clock synchronization and recovery scheme. PSoC 3 also ensures bit-perfect audio with minimal external components at multiple sample rates (at standard intervals up to 24-bit audio, 48- or 96-kHz sampling rate).

Development examples that already use Cypress's PSoC 3 MFi Digital Audio solution include the new Avid Fast Track Solo and Duo recording interfaces and the Tascam iM2, iM2X, and iXJ2 microphones (TEAC's professional division).

The CY8CKIT-033A kit works with any Core Audio or Core MIDI iOS app, including Apple's GarageBand, and music streaming apps such as Pandora and Spotify. To jump start development with the Apple External Accessory (EA) framework, the kit comes with Cypress's EA Console example iOS app, enabling bidirectional communication between apps and accessories attached to an Apple device. For example, the EA console can be used to deliver firmware upgrades from the Apple device to a connected accessory. The PSoC 3 MFi Digital Audio solution also integrates Cypress CapSense solution for capacitive touch-sensing buttons, sliders, and proximity sensing to enhance the end products' user interface and aesthetic appeal.

Cypress Semiconductor Corp.
www.cypress.com/go/cy8ckit-033A

Closed-Loop I²S 20-W Audio Amplifiers

Texas Instruments (TI) introduced a family of four digital-input, closed-loop I²S audio amplifiers, delivering a new benchmark in performance and solution size. The TAS5760*xx* devices feature high switching frequency up to 1.2 MHz, reducing electromagnetic interference (EMI), which enables designers to use smaller output filter components to reduce overall solution size and bill of materials (BOM) cost.

The TAS5760*xx* family is ideal for mid-power stereo audio applications including TVs, soundbars, portable docking stations, Bluetooth speakers, and aftermarket automotive audio products. The devices stream 20-W continuous power per channel without a heatsink and deliver the industry's best performance specifications, including THD+N of 0.03% at 1 W, −92-dB crosstalk, and 66-µV idle channel noise. The devices support a wide voltage range from 4.5 to 26.4 V, enabling enough battery and AC line operation to satisfy the majority of mid-power applications. The TAS5760*xx* family also has superior thermal performance for today's consumer audio products that operate at high ambient temperatures. The devices offer flexible configuration options via software (I²C) and hardware (GPIO pins) control modes support multichannel system designs. Amplifiers can be combined with TI's PCM1808 stereo ADC, featuring 99-dB SNR to create a high-performance audio signal chain.

The new closed-loop I²S amplifiers are available in 8-mm × 12-mm, 48-pin thin shrink small outline packages (TSSOPs). The TAS5760*xx* are available in four pin-compatible, 48-pin versions: the TAS5760MD and TAS5760LD with headphone amplifiers/line drivers, and the TAS5760M and TAS5760L non-headphone versions. For projects requiring a smaller footprint, the TAS5760M and TAS5760L are also offered in 6-mm × 9-mm, 32-pin packages.

Texas Instruments, Inc.
www.ti.com/tas5760md-pr

The Texas Instruments TAS5760MD is a stereo I²S input device that includes hardware and software (I²C) control modes, integrated digital clipper, several gain options, and a wide power supply operating range to enable use in a multitude of applications. The device has an integrated DirectPath headphone amplifier/line driver to increase system-level integration and reduce total solution costs.

Focal Professional Launches Spirit Professional Headphones

Focal's Spirit Professional headphones are one of the great surprises of the year in terms of monitoring tools for professional use.

"Listen to your music, not to your headphones!" embodies the "spirit" for Focal's new product entry. The Spirit Professional headphones are the direct result of Focal's historical expertise in the high-performance transducer design, promising neutrality and transparency for total control. In line with the French manufacturer's monitoring loudspeakers, the Spirit Professional headphones ensure high-quality control of monitoring activities without exception, freeing professionals from the acoustic constraints related to their workstation's size.

Focal chose the circumaural design for the Spirit Professional headphones to optimize the acoustic coupling to the ear. This monitoring tool, typically used in professional recording studios, is also suitable for broadcasting or home studios, where the noise level is much higher.

The frequency of use and amount of time used are two of the most important aspects to consider when designing professional headphones. Comfort optimization was essential for Focal. The choice to use memory foam for the ear pieces and the headband ensures a perfect fit regardless of user's morphology. What's more, the ear pieces' dimensions have been increased to reduce the pressure level. The size increase guarantees insulation and prevents fatigue. Finally, the black textured finish gives the headphones a simplistic look and a shock-and-scratch-resistant quality.

According to Focal, the Spirit Professional headphones use mylar/titanium alloy drivers to achieve strong dynamics without distortion and neutrality of tonal balance combined with high definition of audio signals, even at low listening volumes.

Finally, work on the acoustical load of the transducers has resulted in articulated bass worthy of open-style headphones, but with the advantages of insulation. The low-impedance OFC coiled cable (13'/4 m) also enhances the sound transparency, an important concept for the brand. Focal also provides a straight cable (4.6'/1.4 m) with a built-in track selector and microphone for receiving calls as an accessory to ensure mobility. The Spirit Professional headphones cost approximately $300.

Focal Professional
www.focal.com

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iZotope Receives Engineering Emmy award

iZotope, a Cambridge, MA, innovative audio technology company, received an Emmy Award for Outstanding Achievement in Engineering Development for its RX audio repair technology from the National Academy for Television Arts and Sciences. The Engineering Emmy is presented to an individual, company, or organization for engineering developments so significant that they materially affect the transmission, recording, or reception of television.

"We are incredibly proud to be recognized by the Academy for the development of iZotope RX," said Mark Ethier, iZotope's CEO. "iZotope's customers have helped shape RX into what it is today. Their feedback has been invaluable as we've refined RX's workflows and strengthened its processing power. We are excited to be recognized as a game-changer in the television and broadcast industries and are sincerely humbled by this award."

For audio professionals in the television industry, RX quickly became a standard tool for rescuing audio from the cutting-room floor. With remedies for noise, clipping, hum, buzz, crackles, unwanted reverb, and more, iZotope's RX visual audio repair technology makes it possible to clean up dialogue that has been spoiled by weather conditions, microphone issues, equipment buzz, and sudden noises (e.g., dog barks, sirens, etc.).

The iZotope RX 3 Advanced audio repair suite enables fast dialogue clean-up with the all-new real-time Dialogue Denoiser.

The RX has a powerful spectrogram that exposes details that are hidden in a standard audio waveform. By revealing a rich visual display of audio frequency over time, RX enables engineers to detect and isolate problems with innovative selection tools. In addition, RX features intelligent, cutting-edge modules that are carefully designed to tackle specific audio problems.

Recently upgraded to version 3, the audio repair software suite, now works up to six times faster thanks to processing enhancements and a redesigned user interface. RX 3 Advanced also received a new Dereverb module, which helps remove or reduce reverb from existing audio files.

Audio engineers can also use their eyes and ears to identify and fix problems within a spectral audio editor. They can gain unlimited Undo history, which is automatically saved with audio data into the new RX document format.

RX 3 and RX 3 Advanced can be used as a stand-alone audio editor or as plug-ins in any industry-standard editors, integrating seamlessly with any post-production workflow. Supported plug-in formats include 64-bit AAX (Pro Tools 11), RTAS/AudioSuite (Pro Tools 7.4-10), VST, VST 3, and Audio Unit.

RX Audio Repair Technology was honored along with YouTube, Aspera, Digital Dailies, and Lightcraft Technology on October 23, 2013 at the Lowes Hollywood Hotel in Hollywood, CA.

iZotope, Inc.
www.izotope.com/rx3
www.emmys.com

FIRmaker Focuses on Precision

The Ahnert Feistel Media Group (AFMG) introduced its FIRmaker software during the Prolight+Sound Show in Frankfurt (Germany). The results were felt across the audio industry. FIRmaker is a revolutionary sound optimization tool for next-generation sound systems. FIRmaker enables users to tune DSP finite impulse response (FIR) filters to every possible venue within seconds.

Berlin-based AFMG is also responsible for the industry-standard EASE and EASERA software for acoustic simulation and measurement as well as its related products—EASE Focus, AFMG SysTune, EASE Address, EASE Evac, AFMG SoundFlow, AFMG Reflex, and EASE SpeakerLab.

AFMG presented FIRmaker during the Prolight+Sound 2013 show. FIRmaker can calculate the best sound configuration for a specific sound system using venue acoustic model information. The results can be loaded into any DSP system.

Technically speaking, FIRmaker represents a highly sophisticated optimization algorithm that is designed for computing optimal FIR coefficients tailored to each venue. Based on input data describing location geometry and sound sources, AFMG FIRmaker computes optimal filter transfer functions that can be automatically converted to FIR filters. FIRmaker provides exact sound system coverage and sound pressure level (SPL), offering easy on-site tuning within minutes with any loudspeaker array, existing or new.

The FIR Approach

With the increase in DSP processing power, FIR filters are used by loudspeaker manufacturers to design system-specific filter sets. FIR filters enable extensive and very fine-grained system tuning and, more importantly, they enable independent frequency response and phase control. This has not been possible with other prior filters.

Manufacturers usually invest large amounts of time and energy into their system specific filter set designs. However, such filter sets are specific to each system and do not incorporate the venue demands since this can be different from day to day.

FIRmaker changes the FIR filter design process, applying highly sophisticated algorithms that take seconds to calculate optimal filter sets for a system. It calculates a linear frequency response as well as a custom target curve, so sound can be electronically steered directly into the audience while maintaining the same frequency response throughout all the audience areas.

The system's complete radiation behavior can be electronically changed without touching the rigging or setup with the simulation of the system response. This provides an immediate preview of the results, even before the filters are generated and loaded.

Users remain in control of all parameters, make the final decisions on which filters to implement and, afterwards, can continue to use all their previous tuning instruments.

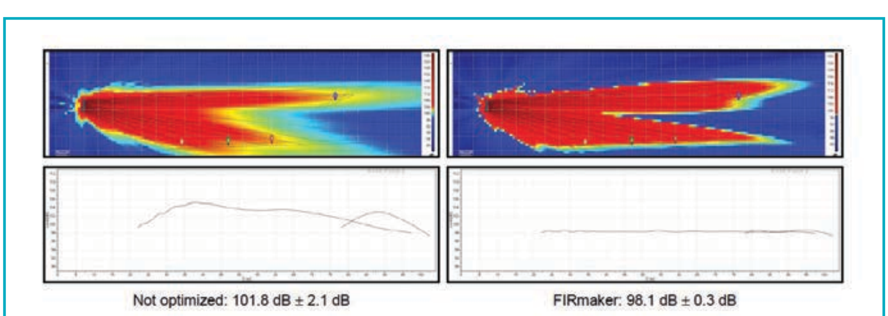
While FIRmaker algorithms are complex, its optimization approach is straightforward, simple, and easy to use. All that is needed for a FIRmaker optimization is the system manufacturer's high-quality loudspeaker "GLL" data; the exact system setup configuration provided using, for instance, the EASE Focus 2 software; a detailed layout of the venue's audience areas, sections to exclude; and priorities for system SPL or venue coverage, which can also be provided by the user with EASE Focus 2. FIRmaker will calculate the system's optimal FIR filters and provide a preview of the resulting system response for the user to evaluate and base his decision.

Industry Adoption

Since the new FIRmaker technology was introduced, several companies in the industry have already adopted it. A key AFMG partner is Lab.gruppen, which combined this technology with its powered loudspeaker management (PLM Series) platform. Italian amplifier manufacturer Powersoft integrated FIR filters directly into Powersoft's flagship product lines, the K series amplifiers and the DSP4 module for powered loudspeakers.

Another Italian manufacturer, K-array, successfully completed its first live tests with FIRmaker after being the first company to evaluate and license the technology. VUE Audiotechnik and ESS Professional Audio from Huanan, China, also decided to license FIRmaker for use with EASE Focus 2.

The number of FIRmaker-enabled DSP platforms is rapidly growing. In response to customer interest, German manufacturer AllDSP announced support for AFMG FIRmaker. In cooperation with AFMG, AllDSP plans to enable external control units and integrated DSP boards for use with FIRmaker. Apart from dedicated platform integration, FIRmaker is generally able to export FIR filters in almost any file format (e.g., CSV, XML, or WAV) enabling conversion to compatible formats with solutions such as Biamp Tesira, BSS SoundWeb, Electro-Voice IRIS-NET, or the Symetrix Solus.



FIRmaker's most impressive feature is the precision of the results with the calculated models showing ±1-dB differences in the complete space and tolerances of 1 to 2 dB compared with real measurements. For instance at 2,000 Hz the conventional setup's directional pattern breaks up into multiple lobes. This results in a fairly uneven SPL distribution. FIRmaker, however, provides a flat spatial response with only a small reduction in average level.

AFMG Techonolgies GmbH
http://firmaker.afmg.eu
www.afmg.eu

Klotz Cables Comes to North America

The Munich (Germany)-based KLOTZ Audio Interface Systems (A.I.S.), renowned makers of KLOTZ Cables appointed TruNorth Music and Sound as its exclusive US and Canadian distributor for its music industry (MI) retail product segment. An agreement was signed on July 30, 2013 between Dieter Klotz, KLOTZ A.I.S.'s Managing Director and TruNorth's President Jeff Sazant.

KLOTZ Cables is made its North American presentation at the 135th International AES Convention in New York, and plans a major launch of the brand at Winter NAMM 2014 in Anaheim, CA.

"We have been looking intently at the North American MI market for a couple years now to find a suitable partner and we are very pleased to have found TruNorth Music and Sound as our dedicated distribution partner for both the US and Canadian MI retail markets. With TruNorth's Jeff Sazant and his team, we have appointed a young and dynamic distribution company, which is strongly devoted and focused on its fine selection of best-in-class Music and Audio brands" I strongly believe in TruNorth's distribution abilities and it's full service offer. Their strong dealer network will make KLOTZ Cables a great success story in North America", said KLOTZ Cables Director of International Sales, Frederic Kromberg.

"We are elated to be partnering with KLOTZ Cables in North America!" states Jeff Sazant, TruNorth's principle. "KLOTZ is the perfect manifestation of a complete cable range that has already proven 'best in class' in every market they serve. TruNorth is extremely fortunate to have been chosen by KLOTZ to bring their superb products to the North American market. Our research reveals that KLOTZ is well positioned to compete by offering a stunning, high quality cable line with strong marketing elements and excellent dealer margins. We believe that we are offering the best cable brand never yet seen in our markets."

KLOTZ A.I.S. is a globally renowned German manufacturer of professional audio, video, multimedia, and fiber-optic cables and cable solutions with a strong presence in more than 100 countries, already established throughout Europe, the Middle East, Africa, Asia, and Asia Pacific.

TruNorth Music and Sound, Inc.
www.trunorthmusicandsound.com
KLOTZ Audio Interface Systems A.I.S. GmbH
www.KLOTZ-ais.com

KLOTZ's "Made in Germany" hallmark of quality includes a diversified catalog of pre-made cables and bulk solutions for audio installations and cables for 10Gbase-T Ethernet audio applications that comply with the Power over Ethernet (PoE) standard.

Switchcraft Receives US Patent for Phantom Lift Technology

Switchcraft, the Chicago, IL, specialist in interconnect solutions for audio, video, and data, received a US Patent for its Phantom Lift. The Phantom Lift technology is currently incorporated into its 900 Series Instrument Direct Boxes.

The 900 Series boxes, which are geared toward live sound applications, are the world's only direct boxes with a ground lift switch that can be remotely controlled using phantom power from almost any standard audio console. In the past, any unwanted noise on a line utilizing a passive direct box meant someone would have to run on stage to flip the ground lift switch on the direct box. Phantom Lift enables the engineer to remotely activate the switch by simply engaging phantom power on the console.

Switchcraft
www.switchcraft.com

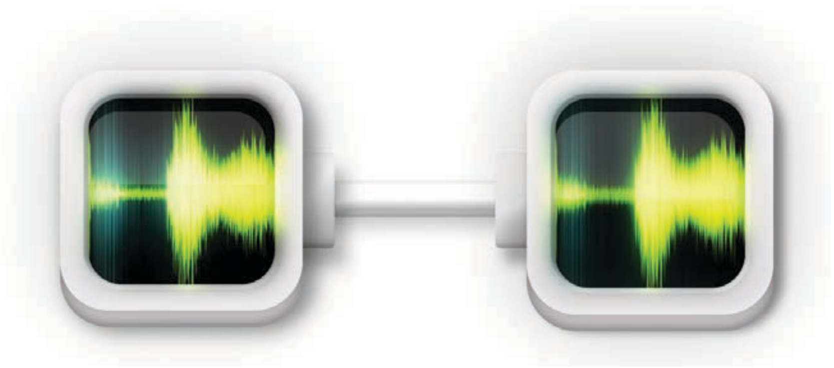
The Switchcraft 900 Series Instrument Direct Boxes have a ground lift switch that can be remotely controlled using 48-V phantom power or directly using the integrated toggle switch.

Audio for Mobile Platforms: The New DAWs?

From Audiobus to Apple's Inter-App Audio

By
João Martins
(Editor-in-Chief)

It's no secret that Apple's iOS devices have been globally adopted for music, both as players and for music recording and production. Especially with the iPad, the iOs systems' music and audio recording capabilities have achieved a level no one had anticipated thanks to the combination of CPU power and an Apple-designed development-friendly ecosystem.



Audiobus is one of the most exciting developments for mobile platforms ever in terms of audio technology.

The result was a much larger than anticipated adoption by musicians, producers, DJs, and audio engineers worldwide, who were lured by the convenience of all those capabilities in a mobile platform. That success attracted even more manufacturers and developers to the iOS environment. This created a new generation of apps that are no longer trying to emulate desktop software and instead concentrating on creating new music-making experiences.

Since March 2013, Audiobus (based in Melbourne, Australia) has created a whole new technology dimension, creating audio communication channels between multiple apps running in the same device, much like a virtual routing system.

The revolutionary inter-app audio routing Audiobus Software Development Kit (SDK) is available for all iOS developers. When the Audiobus app was released, more than 100 compatible applications were in development or already available. And in the weeks immediately following, Audiobus-compatible apps were introduced by the "big names" such as Steinberg, Korg, Moog, IK Multimedia, and Propellerhead, quickly updating their existing apps recognizing the importance of this technology.

Suddenly, it became possible to play a Moog synthesizer, process it in the (IK Multimedia) AmpliTube rack, and record it in (Steinberg) Cubasis—with one iPad! As of August 2013—six months after SDK availability—there were already more than 250 Audiobus-ready apps. Amplitube Fender for iPad was the 250th.

Holy Grail

When the first-generation iPad became available, no one anticipated such a device could become popular in the recording studio or as a musician's tool. But the convenience of the mobile platform and the incredible touchscreen quickly inspired developers to create versions of existing sheet music software as well as virtual instruments and sequencers. With the pro audio community also embracing the iPad and the iPhone as remote controls for the latest generation of digital mixers and audio networks, recording and processing became the natural next step.

Then, something extraordinary happened. The iPad gained a functionality that placed it one step further from the traditional digital audio workstations (DAWs) on PC and Macs: its ability to internally route audio and Musical Instrument Digital Interface

(MIDI) signals. Suddenly, the idea of working "in the box" evolved to a whole new level.

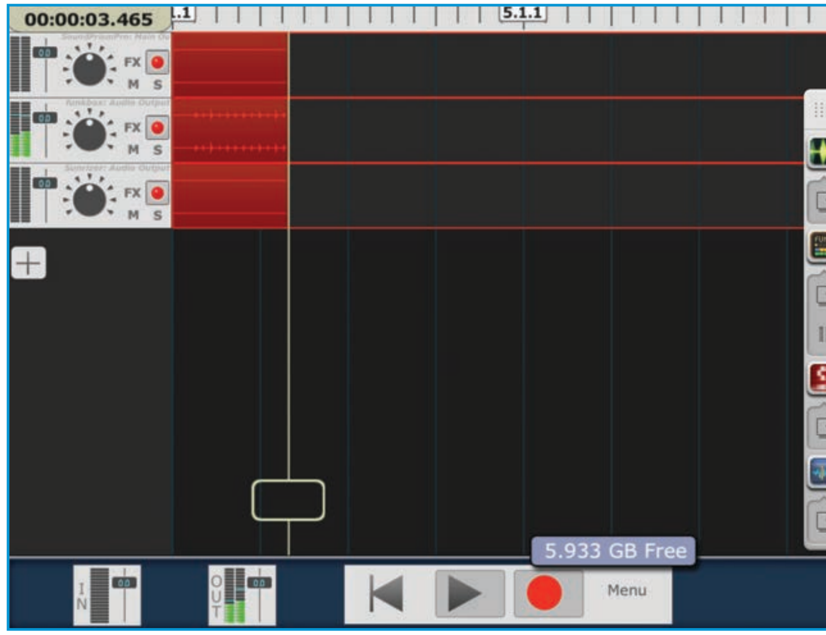
The whole idea of virtual routing audio signals and even MIDI within different software applications that run on the same computer has always been one of the industry's "Holy Grails," since computers were introduced in the studios. Somehow, because both CPU limitations and the fact that streaming audio technologies were not that advanced, software developers never insisted on that route. The natural workflow for the studio became a constant input/output flow of audio signals to and from the computer and audio interfaces. Looking at the most popular DAW systems' entire architecture and plug-ins, it's clear that routing audio and MIDI is always done within that single-host application concept, mainly because the existing OS structure only enables the audio engine to be controlled by one application a time.

Ironically, as a single closely integrated platform where hardware and software work in tandem, maximizing and optimizing performance at all levels, the iOS platform suddenly enabled the creation of the first "universal audio bus" between apps from different developers. Audiobus was not the first effort in that area, but it was the first available SDK with the ability to interchange audio among a sound generator app, an audio processing app, and a third recording app, without requiring the audio to be removed from the device itself. Audiobus also operates independently from external audio peripherals.

And since we can expect the iPad's memory and processing power to exponentially grow every year, it's obvious that this is going to be big in the near future. Just remember that in September 2013 Apple introduced a "desktop class" 64-bit platform on its iPhone 5s A7 chip with more than 1 billion transistors in a 102-mm die size. Apps optimized for the A7's new 64-bit architecture, with its ARMv8 instruction set, compilers, and development tools within Xcode, immediately reflected important efficiencies, effectively enabling users to implement new features and effects. It also doubled the speed of audio processing while processing graphics four times faster.

In 2011, the idea of using the iPad (or worse, an iPhone) in a professional recording studio seemed outrageous. Just two years later, this is a reality—albeit with severe restrictions in terms of track count and sampling frequency. With this same A7 architecture reaching the iPad, suddenly, some of the real-time audio operations recently achieved on a workstation level platform can become a reality on mobile (computing) devices. Real-time audio convolution on mobile devices anyone?

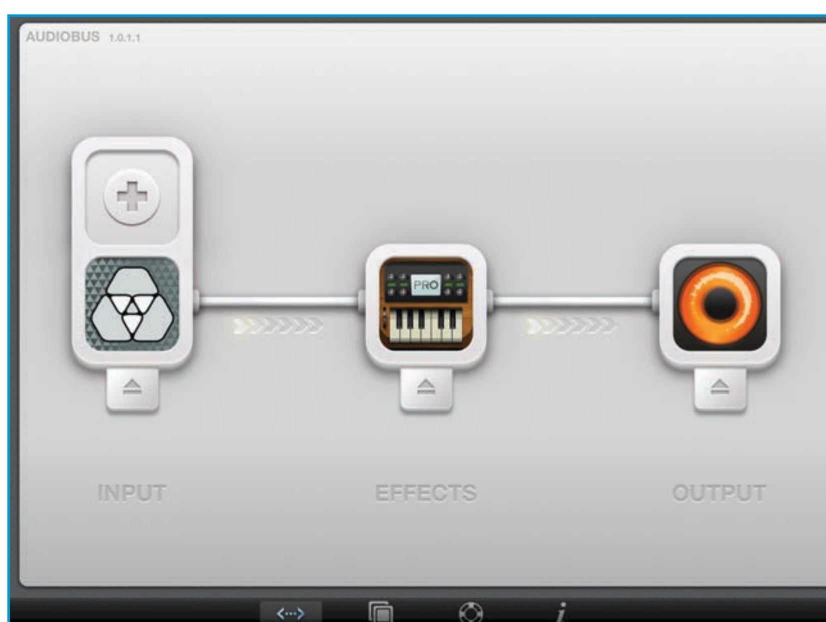
That is why technologies such as Audiobus are so important for the audio community.



A simple multitrack recorder app on iPad showing the Audiobus communication lateral menu that shows which apps are actively communicating.

Fruitful Initiative

Audiobus resulted from a joint venture between British-based A Tasty Pixel, an iOS audio app developer, and German-based Audanika, previously known for creating the sequencer and recorder SoundPrism and SoundPrism Pro iOS apps. Both companies quickly discovered that the Loopy apps from A Tasty Pixel



Audiobus was formed when two companies—A Tasty Pixel and Audanika—discovered their apps could create a powerful complementary system if they interchange audio.

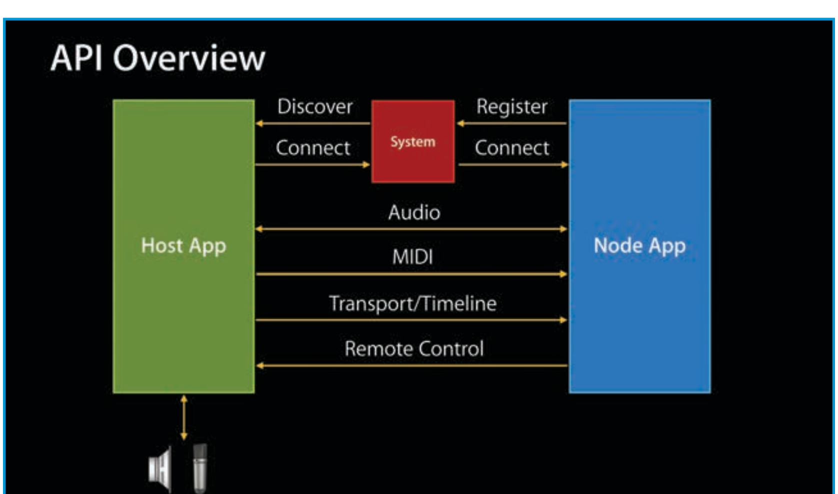
Audiobus basically enabled "virtual cables" to connect different apps running on the same (mobile) platform. Users can have a synthesizer, a microphone, an instrument app, a sequencer, a radio receiver, a sampler, or a voice processor, send audio to an effects processor and receive it in another, such as a multitrack recorder or a mixer. This was never possible between applications on a standard PC/Mac since the operating systems never allowed control of the audio engine by more than one application at a time.

together with SoundPrism Pro could create a powerful complementary system if theycould interchange audio. After 12 months of work, they found a way for the iOS system to enable audio streaming between apps. The results were so promising, the two companies created Audiobus as a separate business and made the technology available to developers as an SDK.

Envisaging a huge impact on the market, Michael Tyson, the founder and main Audiobus developer, stated "Audiobus technology will fundamentally change the way people are able to make music with an iPad and iPhone."

Basically, Audiobus was implemented as though there are "virtual cables" connecting different apps running concurrently in the same iOS device. It enables different apps to not only exchange audio streaming but also talk to each other in terms of basic transport functions. After installation and activation, the technology is essentially transparent to the user, becoming a feature of the device itself as long as the Audiobus-compatible apps are running. A synthesizer can play in real time through an effects processor and a multitrack recorder registering the results and it all simply works. All the leading names in music software were among the first 25 developers to release Audiobus-compatible apps, while 750 other developers were still testing the Audiobus SDK.

With its Inter-App Audio implementation in iOS7, Apple basically integrates the same level of functionality previously only available on Audiobus, adding MIDI integration together with audio streaming between apps, at the system level.



Software instrument developers (e.g., VirSyn) introduced new improved versions of apps (e.g., the Addictive Synth), not only with AudioBus support but also with MIDI Clock sync and lower latency. VirSyn also launched AudioReverb for iOS with Audiobus support, an algorithmic reverberation app (algorithmic- and convolution-based) that deeply enhances the listening experience with any iOS audio apps, including iTunes.

Other examples of Audiobus implementation include the IK Multimedia AmpliTube apps, which enable reception of three Audiobus "slots" for effects and amp modeling, generating sound as a drum machine and recording all the results in multiple tracks.

In March 2013, Apple announced support for the Audiobus technology on its GarageBand app, which is in fact the reference app implementation for every other audio app developer. Apple confirmed it would support Audiobus following the SDK release.

Audiobus and Apple

The work of the amazing Audiobus team is lead by Tyson and Sebastian Dittmann, who is one of Audanika's founders. And, the team is not stopping now that they have caught Apple's attention.

In fact, Tyson and the team are also working on overturning the current audio limitation in iOS by developing The Amazing Audio Engine, which is a sophisticated framework for iOS audio applications that is developer-ready and built on the efficient and low-latency Core Audio Remote I/O system.

The Amazing Audio Engine starts where Audiobus stopped, enabling users to manage "audio objects"(e.g., audio units or loops) by carrying within those objects information about the audio signal itself as well as timestamps, stereo panning, filters, and effects, effectively creating a framework for a true digital audio workstation environment.

By including things such as support for interleaved and non-interleaved files, full core audio and core audio remote I/O, the device provides a synchronization between audio apps in iOS.

Tyson is also aware that its achievement with Audiobus faces some competition. Among the competition is JACK for iOS, an app created by Christian Schoenebeck of Crudebyte, that provides another communication environment inside iOS to enable audio streaming between apps and support full MIDI implementation. With an available SDK, the JACK Audio Connection Kit for iOS focused on the exchange of MIDI messages and sync between apps, but never achieved the same acceptance level as Audiobus.

Until recently, it seemed that the Audiobus technology was proven, endorsed by Apple, and ready to evolve in other directions, eventually becoming a de facto standard. That is, until Apple announced its own plans for iOS 7 and confirmed it was working on technology comparable to Audiobus.

Apple's strange but familiar announcement lead to many unconfirmed rumors, including the possibility that Apple would license the Audiobus technology. Meanwhile, Apple launched iOS 7 as scheduled and implemented its Inter-App-Audio feature, effectively competing with the Audiobus solution for streaming audio from multiple apps. According to Apple, "With Inter-App Audio, apps can register their audio streams to share with other apps. For example, a series of apps could publish audio streams of instrument tracks while another uses the combination of these streams to compose a song. Inter-App-Audio also provides MIDI control of audio rendering, remotely launching other registered Inter-App-Audio apps and more."

With a system-level implementation, Apple did something similar to what we had already seen from Audiobus and JACK, in this case adding the MIDI control of apps.

Apple's GarageBand app for iOS, shown here, was one of the reference implementations for Audiobus. Apple has already updated it using the Inter-App-Audio technology, without losing Audiobus support.

In response, Audiobus posted on its blog, "We're excited about the great work the Core Audio team has done on the new functionality, which has some terrific features that only a team at Apple with system-level access could achieve. It validates the work we've been doing with Audiobus and will pave the way for some great user experiences to come."

Recently, Audiobus confirmed it was working with Apple on new developments and that its technology would "continue to be a valuable member of the community, bringing music apps and devices together."

Whatever the results of this collaboration, I believe Audiobus and the Inter-App-Audio implementation can evolve in specific ways. Currently, it is even possible to make use of both systems concurrently in the same iOS device using different apps. This will create an amazing framework on which many new capabilities for audio applications in mobile devices are possible. With the Apple's new 64-bit architecture, there will be much to look for in the next two years.

Resources

The Amazing Audio Engine, http://theamazingaudioengine.com.

Apple Developer Forums, http://devforums.apple.com.

Audiobus, www.audiob.us.

Audiobus Developer Center, http://developer.audiob.us.

AV Foundation Programming Guide, iOS Developer Library, http://developer.apple.com/library/ios/#documentation/AudioVideo/Conceptual/AVFoundationPG.

M. Tyson, "Thirteen Months of Audiobus," *A Tasty Pixel*, December 2012, http://atastypixel.com/blog/thirteen-months-of-audiobus.

Designing for Ultra-Low THD+N (Part 2)

By
Bruce Hofer
(United States)

Resistors and Capacitors

The first article in this series described how factors beyond the schematic can effect an audio circuit design's real-world performance. This article examines the most common passive components on a circuit board—resistors and capacitors. This article's focus is not what makes the "best" audio amplifier nor do I make any audible quality claims. I shall leave those subjective matters to others. Instead, I will discuss what we at Audio Precision have been doing for many years—understanding what is in fact measurable and what mechanisms contribute to the imperfections we find.

Resistors in Analog Design

Resistors come in many forms, incorporating several different technologies, and they are available at many different price points (see **Photo 1**). The primary types used in audio include: carbon composition, thick film, thin film or metal film, metal foil, and wire wound.

To one degree or another, all of these constructions exhibit two primary types of nonlinearity: voltage coefficient and power (thermal) coefficient.

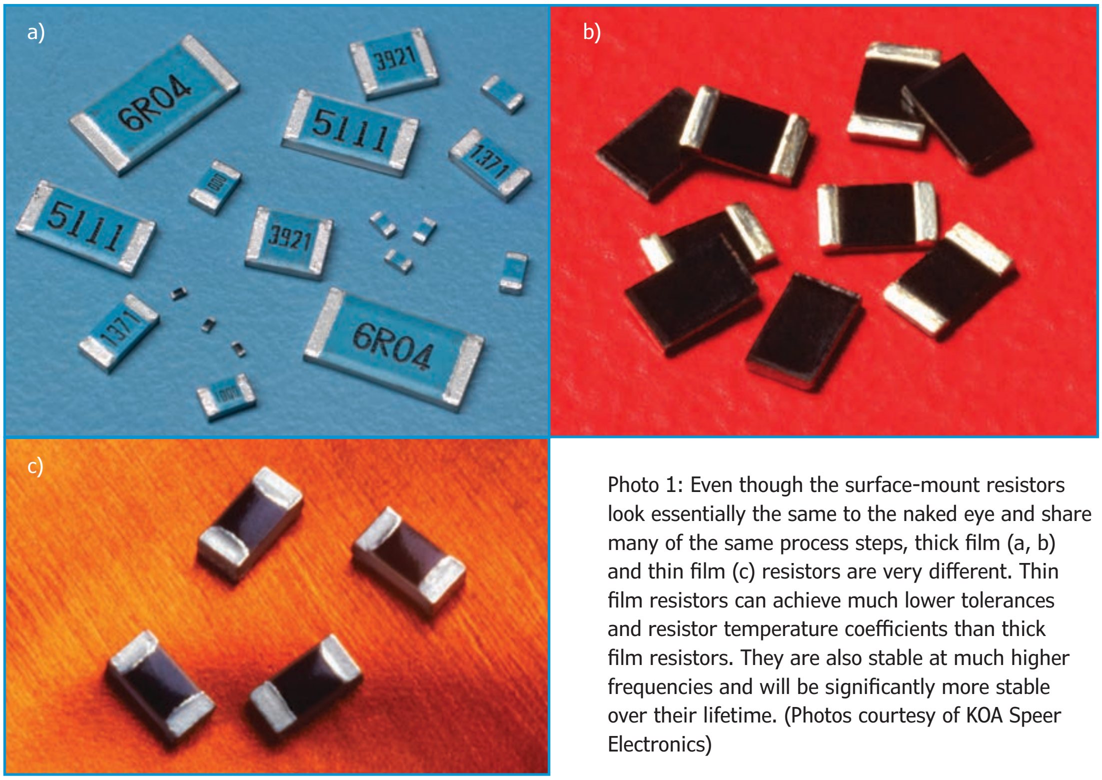


Photo 1: Even though the surface-mount resistors look essentially the same to the naked eye and share many of the same process steps, thick film (a, b) and thin film (c) resistors are very different. Thin film resistors can achieve much lower tolerances and resistor temperature coefficients than thick film resistors. They are also stable at much higher frequencies and will be significantly more stable over their lifetime. (Photos courtesy of KOA Speer Electronics)

I'll discuss how each can be modeled and how this can affect ultra-low THD+N designs.

Defining Resistor Terms

Resistor Temperature Coefficient—All resistors exhibit value changes as a function of temperature. The temperature coefficient of resistance (TC_R) describes the magnitude of this sensitivity and is expressed in units of parts per million per degree C (ppm/C).

In addition to the TC_R's static (DC) values, there is also a dynamic component, which is the result of temperature changes propagating through a resistor as a function of time. This function is often referred to as the resistor power coefficient, as it represents the effects of temperature changes caused by the dissipation of energy in the resistor as a result of applied signals.

Resistor Voltage Coefficient—In addition to changes in value due to temperature, a resistor's value may also change as a function of applied voltage. The voltage coefficient of resistance (VC_R) describes the magnitude of this effect for a given resistor.

The value is expressed in units of parts per million per volt (ppm/V). This value is generally positive and results in a decrease of resistance as applied voltage increases.

Resistor Type Summary

All resistors are not created equal. There are many types of resistors, each with distinctly different behaviors.

Carbon Composition Resistors—A carbon composition resistor's resistive element is a compact mixture of carbon and ceramic held together in a resin base.

Experienced readers will recall that prior to the 1970s, carbon composition resistors were extremely common. This is not true today and for good reasons. Carbon composition resistors usually have poor tolerance (typically 5% to 20%); poor TC_R (typically 150 to 1,000 ppm/C)—and worse at lower values. These resistors also have high modulation noise and high VC_R compared to other types

While still useful in some non-audio applications, carbon composition resistors are *not* recommended for high-performance analog designs. One interesting audio exception is the vacuum tube guitar amplifier design, in which the distortion characteristics of early eras are still prized.

Thick Film Resistors—A thick film resistor's resistive element is a conductive film applied to the surface of a cylindrical or rectangular substrate. Resistance is determined by the film composition and the etching pattern.

This type of resistor is currently popular for general-purpose applications. It offers much better performance than composition type resistors. Thick film resistors have good tolerance (0.1% to 2%), good TC_R (typically 100 to 250 ppm/C), adequate VC_R (it varies considerably from brand to brand, but it can be as high as 10 ppm/V), and adequate modulation noise.

Thin Film (Metal Film) Resistors—Thin film resistor's resistive element is a stable conductive film that is sputtered onto the surface of a cylindrical or rectangular substrate. Resistance is determined by film thickness and pattern.

Thin (metal) film resistors offer superior performance when compared to their thick film counterparts, but at much higher cost. Thin film resistors offer excellent tolerance (0.02% to 1%), excellent TC_R (which is typically 5 to 25 ppm/C but it can be as low as 2 ppm/C), excellent VC_R (between 0.1 and 1.0 ppm/V), and excellent modulation noise.

Metal Foil Resistors—A metal foil resistor's resistive element is a special alloy metal foil that is the cemented to an inert substrate. Resistance is determined by foil characteristics and pattern. Trimming is accomplished by opening links in the foil pattern, which is more stable than "L" cut trimming.

Metal foil resistors feature the best performance at DC. They are also the most expensive. Metal foil resistors offer outstanding tolerance (as low as 0.001%), outstanding TC_R (as low as 0.05 ppm/C), outstanding VC_R (typically less than 0.1 ppm/C), and extremely low modulation noise.

However, praise for metal foil resistors comes with a caveat. Their low-frequency modulation distortion can be much worse than expected.

Wire-Wound Resistors—The resistive element for a wire-wound resistor is a wire with a low-temperature coefficient that is carefully wound on a substrate. Its resistance is determined by the wire's composition, length, and thickness.

Wire-wound resistors are typically appropriate only for smaller resistance values, but they can support high peak and average power ratings. They exhibit almost no VC_R, but due to the combination of low values, stray inductance, and other parasitic effects, they are uncommon in modern audio circuits.

Resistor Nonlinearity

Resistors exhibit nonlinearity from two sources: the voltage coefficient and the power (thermal) coefficient. To understand the differences, I will explain

About the Author

Bruce Hofer is a co-founder of Audio Precision and a widely respected analog design expert with a career spanning more than 40 years. His designs rank among the quietest and lowest distortion audio measurement circuits in the world.

how these are modeled.

Resistor Voltage Coefficient Nonlinearity—In most common electronics designs, the VC_R is safely assumed to be negligible. This is not the case when designing ultra-low THD+N devices. Note that while VC_R and the subsequent TC_R are correlated as a matter of practice, they represent wholly separate phenomena, which must be taken into account.

The model commonly used for voltage coefficient nonlinearity is dependent on the applied voltage, as shown below:

$$R(V_S) = R_0 \times (1 - VC_R \times |V_S|)$$

Because this model employs the absolute value of the applied voltage (V_S), use of a Taylor series is not appropriate. However, we can estimate the distortion by taking the fast Fourier transform (FFT) of the product of a sinusoidal signal multiplied by a full-wave rectified version of itself (e.g., the FFT of sin(ωt) × | sin(ωt)|). In this case, it follows that:

$$2HD \approx 0$$

Assuming no significant DC component, and thus:

$$3HD \approx \frac{|VC_R \times V_S|}{5.9}$$

Note the proportionality to V_S and not V_S^2 as might be expected if you used a Taylor series model for nonlinearity.

$$5HD \approx \frac{|VC_R \times V_S|}{41} \approx \frac{3HD}{7}$$

In other words, the fifth harmonic's magnitude is estimated to be approximately –17 dB below the third harmonic. It is clear that nonlinearities due to this mechanism are dominated by the third harmonic's presence.

Resistor Power Coefficient Nonlinearity—The nonlinearities that arise from power dissipation in a resistor are a result of temperature changes, forming a correlation between applied signals and the TC_R. Simply looking at a static model of the power coefficient, we can model this as:

$$R(P_S) = R_0 \times (1 + k_P \times P_S)$$

Where k_P is our power coefficient and P_S is the power being dissipated by the resistor. The power coefficient is expressed in units of ppm/W and may be either positive or negative.

Resistor Thermal Modulation Nonlinearity—The power coefficient model is a static description of the effects of internal heating due to an applied voltage. In ultra-low THD+N analog design, nonlinearities that are correlated with changes in signals can affect the outcome.

Because a resistor cannot instantaneously dissipate heat, the effects of thermal changes due to applied signals are complex functions of frequency that are highly dependent upon resistor size, placement, and construction. Thermal modulation can be modeled as follows:

$$R(V_S) = R_0 \times \left[1 + TC_R \times Z(\omega) \times \left(\frac{V_S^2}{R_0}\right)\right]$$

Where Z(ω) is the device thermal impedance as a complex function of frequency. As ω -> 0, | Z(ω)| -> θR, or the DC thermal resistance.

Thermal Modulation at Very Low Frequencies—At very low frequencies (typically less than 0.2 Hz), the resistor reaches thermal equilibrium as quickly as the signal varies. The model can be used to predict:

$$2HD \approx 0$$

A True Story of Resistance

About 13 years ago, a certain manufacturer changed its network substrate material from ceramic to passivated silicon without notifying customers. Ceramic is brittle. It is also more expensive to process and cut to size, and the company was hoping to save money.

Although the resistor DC parameters remained unchanged, the AC performance proved to be disastrous. The stray capacitance between each resistor and the substrate was higher and nonlinear. It is believed that P-I-N diodes were formed between each resistor and the semi-conducting substrate, thus causing the voltage drop in one resistor to generate distortion products in the other resistors!

The manufacturer quickly added the "option" to specify the original ceramic substrate when told it was about to be disqualified by key customers.

Assuming no significant DC bias:

$$3HD \approx \frac{TC_R \times \theta_R \times \frac{V_S^2}{R_0}}{4}$$

$$5HD \approx 0$$

Contrast this with 5HD ≈ 3HD/7 for the resistor voltage coefficient distortion.

Thermal Modulation at Low Frequencies—Within the 5-to-200-Hz range, the magnitude of Z(ω) is usually (though not always) much smaller than θR, rolling off to near zero above 1 to 5 kHz.

There are exceptions. My recent experiments (which are also corroborated by another individual) show that some metal foil resistors with exceptionally low TC_R behave as if | Z(ω)| >> θR, thus contributing higher modulation distortion than a thin film resistor with a larger TC_R under identical conditions.

Controlling the Effects of Resistor Nonlinearity in Audio Circuits—The mechanisms previously described may have measurable effects on audio circuits.

Resistor Voltage Coefficient Effects—Even low VC_R values can introduce measurable distortion. Consider a single feedback resistor in an amplifier producing a 50-V output, which is not an uncommon value in higher power designs. Even if this resistor provides a respectable 10 ppm/V VC_R, it is now contributing 0.05% distortion, which is very significant.

Now, consider that there are many resistors in the circuit, all contributing in varying ways and degrees. Wisely choosing resistors can make a real difference in your design's quality.

Reducing Voltage Coefficient Nonlinearity—Use thin film resistors for most applications. These resistors are much better than thick film with regard to VC_R by a factor of 5 to 10. Never use carbon composite resistors, unless high amounts of VC_R distortion are desired.

Using multiple resistors in series can significantly reduce problems associated with VC_R. A string of n identical resistors in series reduces the applied voltage across each to V/n, thus reducing the effect of any VC_R nonlinearity.

Resistor Temperature Coefficient Effects—As I discussed earlier, the thermodynamic behavior of resistive elements means that a resistor cannot instantaneously change temperature as the signal across it changes. It acts as a low-pass filter for temperature change.

At DC, there is not a problem with regard to linearity as the resistor is nominally at thermal equilibrium. Likewise, at very high frequencies any changes occur so rapidly that the low-pass filtering effect of the resistor's thermal behavior results in an even temperature distribution across the element.

Problems occur in a middle zone of approximately 5 to 200 Hz. In this range, the resistive element's temperature may vary with time as a function of signal. It may also vary as a function of location within the resistor itself. The result can be significant and measurable distortion at those low—and often audible—frequencies.

Making matters worse from an audio perspective, the temperature coefficient's effects are proportional to the signal's absolute value applied across the resistor. This means the resulting distortion is dominated by third harmonics—a very audible variety of distortion that occurs in a high hearing sensitivity area.

Reducing Temperature Coefficient Nonlinearity—Avoid the common 25-ppm/C characteristic in critical circuit locations. Instead, opt for premium parts that deliver 10-ppm/C or even 5-ppm/C characteristics.

Avoid metal foil resistors. While they have very low temperature coefficients (less than 1 ppm/V), they exhibit poor AC performance. This information isn't found in the datasheets, which typically only show DC values.

For surface-mount technology (SMT) resistors, only use the 1206 size. Smaller resistors exhibit increasingly larger dynamic thermal impedances, thus exacerbating signal-dependent nonlinearity for ultra-low THD+N audio.

Limit the signal across resistors to approximately 20 mWpk or about 3 VRMS (12 dBu) for the lowest distortion. Also, use series-parallel combinations in circuits requiring higher peak power dissipation or higher voltage.

Resistor Networks—Resistor networks

Photo 2: Mechanical resonance is a key feature in audio capacitors, which come in a variety of types and sizes. (Photo courtesy of Humble Homemade Hifi, www.humblehomemadehifi.com/Cap.html)

offer extremely low differential temperature coefficients. They are especially useful in applications that benefit from precise ratio matching.

Ratio accuracy can be as good as 0.01% for thin film and 0.001% for metal foil. Watch for high thermal modulation effects with metal foil resistor networks.

Avoid large ratios of R (e.g., 10:1 or higher). The best performance is achieved when all resistors are of equal value.

The small size of some resistor networks (e.g., SOIC-8/-16) means a higher thermal impedance such as Z(ω). The result is higher thermal modulation distortion than discrete resistors.

Reducing Noise from Resistors—Resistor noise voltage is proportional to $\sqrt{R}$ and to $\sqrt{T}$. Hence, a low-noise analog circuit design implies the use of the lowest possible resistor values that are consistent with power dissipation and distortion requirements and circuit topologies that inherently minimize the value of resistors in the signal path. Use only thin film or metal foil resistors when they must pass significant DC bias currents.

Capacitors in Analog Design

Capacitors are used for AC coupling and signal EQ in analog designs. While many people may think that coupling capacitors contribute little or nothing to an audio signal's quality, this is not true. Poor choices in the selection of coupling capacitors can have a dramatic effect on an analog design's performance.

Like resistors, capacitors are implemented in a variety of ways to meet goals of cost, behavior, and

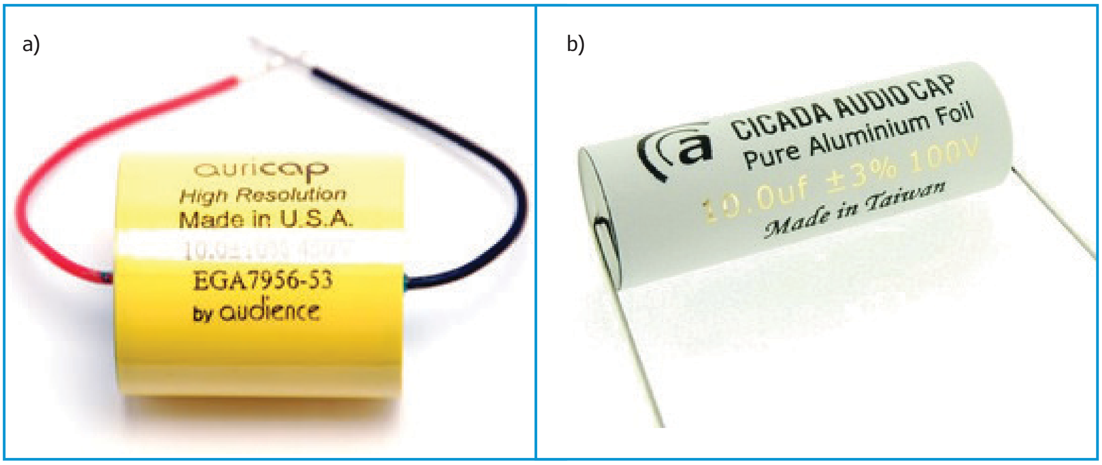


Photo 3a: Auricap metalized Polypropylene capacitors are cylindrically wound with epoxy end fill. b) Cicada Pure Aluminium Foil 100VDC type MKP (Photo courtesy of Humble Homemade Hifi, www.humblehomemadehifi.com/Cap.html)

performance (see **Photo 2**). The common types of dielectric materials found in audio signal paths are: polymer film (PET, PEN, PPS, PP, PS, and PTFE), ceramic (Z5U, X7R, NP0, and Hi-K), mica, and glass.

Polymer File Capacitors—There are many types of polymer film capacitors (see **Photo 3a**). Polystyrene (PS) provides excellent performance with a low temperature coefficient (approximately 100 ppm/C). However, it has a low melting point (85°C), which can result in damage due to soldering. Polypropylene is often an attractive alternative with a higher melting point (105°C) but it also has a higher temperature coefficient (up to 250 ppm/C).

Metalized Film vs. Film-Foil Construction Capacitors—The primary difference between polymer file capacitors with regard to ultra-low THD+N designs concerns the construction.

Film-foil capacitors (also called metal foil) are made using two plastic films as the dielectric. Each film is coated with a thin metal foil that acts as the electrodes (see **Photo 3b**).

Metalized film capacitors also use a plastic film as a dielectric, with electrodes formed by vacuum-depositing aluminum on one or both sides of the film. Film-foil capacitors exhibit lower equivalent resistance and support higher current surges. If possible, select only film-foil types for signal path use in audio designs.

Ceramic Capacitors—With one notable exception, ceramic capacitors are not suitable for use in high-performance analog design, as they exhibit very high voltage dependent nonlinearities.

The exception is the "NP0," or "COG" type of ceramic capacitor. These boast low dissipation factors and frequency dependences. They also have low temperature coefficients (30 ppm/C specified, 15 ppm/C typical). They are extremely stable and virtually immune to humidity.

NP0 types are available in values up to 100 nF with voltage ratings up to 500 V. To achieve higher values and the best performance, consider paralleling multiple NP0 capacitors and avoid ones rated at 25 V. The larger voltage rated capacitors (i.e., 50 and 100 V) are not much larger and offer superior linearity.

In addition to common nonlinearities, certain low-grade ceramic capacitors (e.g., Z5U, Y5Y, and Hi-K) can exhibit strong piezoelectric effects. This means that mechanical stresses on the capacitor can induce voltages across itself, which is something to be avoided at all costs.

Mica Capacitors—Long ago, mica capacitors were highly regarded for analog design, but this is no longer true. While mica offers good stability, it is a product of nature for which the better sources are now depleted. With the availability and lower temperature coefficient of NP0 (COG) ceramic capacitors, there is simply no good reason to specify a mica capacitor.

Glass Capacitors—Glass is among the most stable and inert of dielectrics, with virtually no aging and a near-zero voltage coefficient. However, glass exhibits a higher temperature coefficient than NPO ceramics and is difficult to form with great accuracy, resulting in typical tolerances of 5%. Values of 1% to 2% are available, but extremely expensive.

Microphonic Effect in Capacitors—In any capacitor:

$$dQ = d(C \times V)$$

This equation is often simplified to:

$$I = C \times \frac{dV}{dt}$$

However, "C" itself is not necessarily constant. It may vary as a function of voltage and mechanical stress, can be captured as:

$$I = C \times \frac{dV}{dt} + V \times \frac{dC}{dt}$$

The clear insight to draw from this is to minimize the DC potential across all capacitors in series with the signal path.

Details Matter

Ultra-low THD+N does not occur by accident nor does it manifest purely in schematic diagrams. It is the result of careful attention to detail, clever circuit design, and the selection of high-quality components that are appropriate for each specific job.

elysia xfilter 500

Surface-Mount Magic in a Small Format

By
Miguel Marques
(Portugal)

elysia established itself in 2006 with the launch of the alpha compressor, an all discrete hand-built stereo compressor. Made for mastering, it is one of the most expensive outboard compressors ever built. The German-based elysia quickly became known for its "no compromise" designs and build quality. It is also known for creating innovative solutions that fit the modern sound engineer's needs due to their high versatility, creative functions, and overall excellent sound quality.

Since the launch of its flagship models—which also include a creative compressor called mpressor and a program equalizer called museq—elysia surprised the market with more affordable products by using surface-mounted technologies and machine construction instead of hand-built manufacturing. Among them are the xpressor (a versatile true stereo compressor) and the nvelope (a stereo transient design with some clever bonus features). These two products still feature discrete components on the audio path and always run in Class A. This is in fact part of the elysia DNA. Using the company's own words, "In Class-A amplifiers, the transistors are conductive all of the time, so there is no crossover distortion at all. This is the perfect technological basis for an open sound with massive punch and no degradation of your original source." Also, using discrete components makes it "possible to design every single stage in signal processing to do exactly what you want it to do—physics becomes the only limit."

Features

The xfilter 500 is an active true stereo equalizer in a 500-series module format (VPR Alliance certified). Like the xpressor 500 and nvelope 500, it fills two slots on an API lunchbox or 500-series compatible rack. As of publication, the xfilter is also available as a stand-alone 19" rack unit with the same features and functions so this review is valid for both xfilter 500 versions.

The xfilter 500 is a four-band semiparametric equalizer with the spectrum's outer bands working as low- and high-shelf filters. Its two mid-bands are peak/dip filters with different switchable 0.5- and 0.1-Q factors. The equalizer's outer bands can also work individually as high- and low-pass second-order filters. In this mode, the gain knob controls each filter's resonance. Because the xfilter 500 is a true stereo equalizer, it only has one set of knobs to address the processing on both channels. This is beneficial for users but can create manufacturing issues, as it is

The German-made xfilter 500 uses top-quality components, gold-plated PCBs, a rugged aluminum front panel, and solid aluminum knobs.

difficult to find reliable low-tolerance potentiometers for this task.

The xfilter 500 uses dual- and quad-layer potentiometers. The company's production department obtained an Audio Precision device to build custom computer routines so only the best components were used. To ensure the best tolerance for true stereo operation, the xfilter 500 uses FKS film capacitors with a 5% maximum tolerance. The tolerance is half the 10% most MKS capacitors have and is the industry standard for audio equipment. In all, this unit has the typical mismatch of less than 0.4 dB between left and right channels. This is a good value for any true stereo equalizer, especially one that costs just under $975.

While the museq, elysia's flagship equalizer, uses all discrete components, including the actual filters, the xfilter 500 has ICs for the filter sections. elysia also chose the Burr-Brown OPA amp series, which was designed especially for applications in which pristine audio is a requirement. The mixing stage between the filters and the main signal path and the input and output stages uses discrete components running in constant Class-A operation.

The xfilter 500 has proportional Q behavior in its curves (i.e., there is some interaction between each filter's Q factor and the gain applied). This should make the equalizer adjustments more intuitive and immediate. The xfilter 500 also has a built-in passive filter that can be activated by a creatively named function called "Passive Massage," which is a fixed LC filter with a capacitor and a coil per channel. This filter's response results in a small boost/peak around 12 kHz that gently starts to roll off audio at 17 kHz. This passive filter was added to create a different effect for audio processing, as the coil's selected curve and saturation produces a characteristic high-frequency response that should improve the program's detail and texture in most scenarios.

Other notable features include true bypass of all processing, custom-made aluminum knobs (CNCed through a solid aluminum block), a four-layer PCB design with one dedicated ground shield layer close to the filter networks for minimum electric interferences and maximum hum reduction.

Measurements and Technical Specifications

All the measurements I took are on par with the manufacturer's published specifications. The first specification that stands out is the equalizer's bandwidth. I measured the xfilter 500 to be almost linear up to 200 kHz (±1 dB), which is an impressive value in any price range. According to elysia, the xfilter 500's usable bandwidth goes up to 400 kHz

After exploring the unit's layout, this equalizer quickly becomes a comfortable and functional unit.

with a 3-dB loss, though I really couldn't measure that high. The xfilter 500 is not only a high-bandwidth unit; it is also a high-headroom equalizer with a low noise floor and good maximum I/O levels that provide a total of 120 dB of dynamic range. THD+N levels are also low, typically in the 0.001% to 0.005% range, depending on input levels, which is also a good value for professional audio equipment. Stereo tracking in this equalizer works pretty well. Even though elysia didn't publish this specification, I measured the stereo mismatch and found it to

Manufacturer Specifications

Frequency Response: less than 10 Hz–400 kHz (3 dB)
THD+N at 0 dBu (20 Hz–22 kHz): 0.0018%
THD+N at 10 dBu (20 Hz–22 kHz): 0.005%
Noise floor (20 Hz–20 kHz A-weighted): –98 dBu
Dynamic range (20 Hz–22 kHz): 120 dB
Maximum I/O level: 21 dBu
Impedance (I/O): 10 kΩ/68 Ω
Power consumption (total/per slot): 210/105 mA

About the Author

Miguel Marques, a full-time mastering engineer, works in his own mastering studio in the north of Portugal. After earning a bachelor's degree in Music Production and Technologies, Miguel worked in commercial recording studios as a recording engineer.

Miguel has also written technical reviews and articles for pro audio magazines. Miguel recently finished his first book on audio engineering, which is to be released soon in Portuguese-speaking countries.

The xfilter 500 is a 100% Class-A signal path, resulting in an exceptionally open boutique sound with flawless transient projection and solid punch, ideal for processing single signals, creative sound shaping, mix bus duties, or even helping handle demanding mastering scenarios.

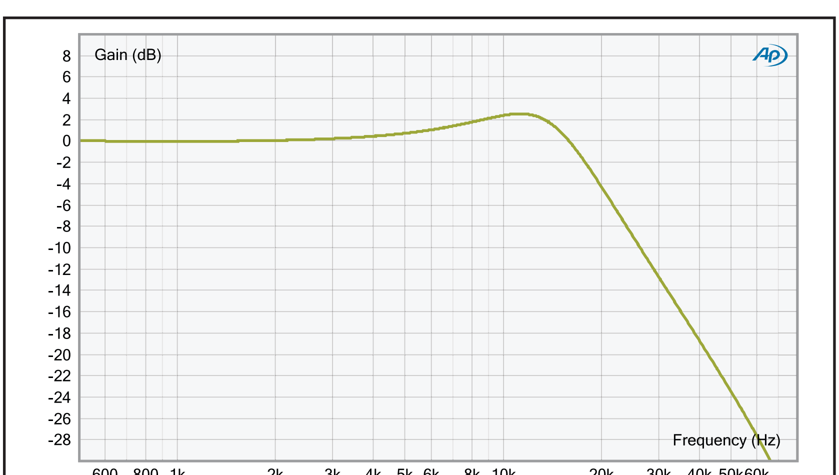


Figure 1: The "Passive Massage" function activates a fixed LC filter with a capacitor and a coil per channel. The response created by this filter results in a small boost/peak around 12 kHz that gently starts to roll off audio at 17 kHz.

be typically under 0.4 dB, which, considering it is a potentiometer-based stereo equalizer, is an impressive value. On paper the xfilter 500 is as good as it gets for most professional audio gear, rivaling most high-end gear's specifications, yet it sells for a relatively affordable price and is a compact unit.

Operation and Sound Quality

The xfilter 500 is a complete equalizer with some knobs used for multiple functions, depending on the processing they're addressing. Because there are so many functions and options (and so few knobs!) it can be difficult at first to operate the unit. Following a few tests with this equalizer, I felt more comfortable with the xfilter 500's layout and operation. It takes some time to get used to it, but after that initial reaction and remembering what each knob does in each mode, the xfilter 500 is actually fairly easy to use.

The only operational issue I found with the xfilter 500 happens when the filter gains need to be set at zero. There isn't a detented zero position for the filter gain, as all 47 steps of each knob are equally detented to help recall. Each filter's zero-gain position can be easily found by looking at the unit and aligning the knob mark to 12:00. It isn't really a big issue, but it requires the user to look directly at the unit. Apart from that, everything feels very solid when operating the xfilter 500. Every knob and switch is clearly labeled.

Even though the xfilter 500 is a stereo equalizer, it can also work on mono sources by feeding one of the unit's inputs. In my tests, I used this unit to process single tracks, buses or groups, and complete mixes. In all the tests, one characteristic prevailed—the open and clean sound—even when doing large amounts of boost and cutting. In almost every situation, the filter range will be more than adequate to accomplish even the most drastic changes to a source.

However, I found that in the majority of cases I wasn't even using half of the ±13-dB range on the mid filters and even less than that on the ±16-dB range of the shelf filters. Because this is a relatively broad equalizer with proportional Q behavior, most adjustments sound immediate, which makes the equalizer a pleasure to use.

If you mostly use digital plug-in equalizers, you'll notice how—with even with the smallest gain, boosts, and cuts—the xfilter 500 will sound instantly effective. Each filter's available frequency ranges provide a decent selection for most duties. At no point in the testing did I feel the need for different or more frequency points.

One curiosity is this equalizer's extended frequency range. For instance, the high-frequency shelf can be set up as high as 28 kHz. Even though 28 kHz is a frequency point already above the hearing spectrum, it creates a gentle curve behind it on the audible range.

The xfilter 500 is broad enough to be used as a program equalizer. It also works on individual instruments with narrower frequency content (e.g., individual drum snares, kicks, or other percussion elements). The xfilter 500 isn't narrow enough to do the most surgical or notch type of filtering because elysia preferred to turn its attention to other functionality types. (The xfilter 500's manual states that most users prefer using digital equalization for these type of corrections.)

One example is the "Passive Massage" function (see **Figure 1**). Even though the function provides a relatively clean saturation and curve change on the program, it creates a noticeable improvement on most sources. Obviously it won't work on everything, but when it does, the "Passive Massage" adds a sheen and sparkle to the source without brightening it excessively. It enhances the top-end response and improves the overall detail and clarity. Finally, the low- and high-pass filters with resonance are a useful addition to any equalizer. The ability to change its resonance makes it more versatile and enables a user to further modify the sound.

From polishing groups and full mixes to changing a single instrument's balance and even to creatively changing a source's balance, the xfilter does it all without ever sounding harsh, uneasy, or over equalized. In terms of function and versatility, the xfilter 500 can only be rivaled by digital plug-in equalizers. I've never seen so much functionality in an analog equalizer of this size.

Surpasses Expectations

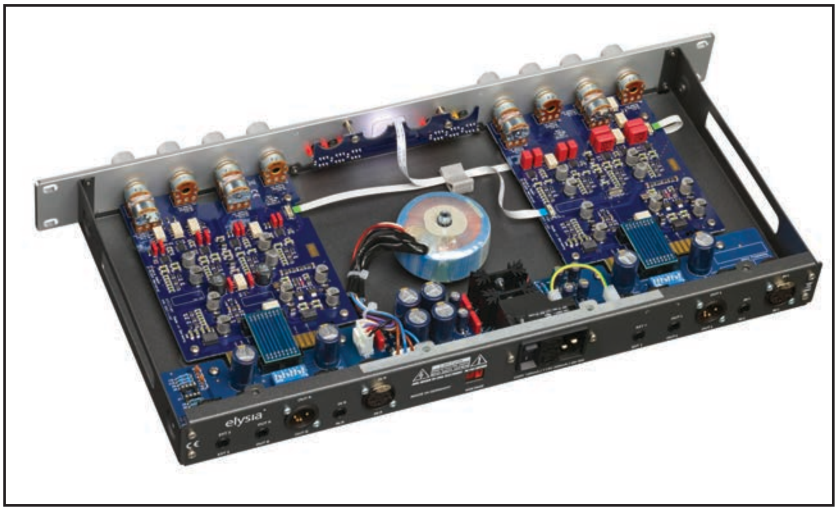
The xfilter 500 excelled with almost every source I tried with it. Apart from notching, the xfilter 500 should be able to accomplish whatever you want to achieve with equalization. And, it can be achieved without damaging the source while producing clean-sounding results. Most of the time, you don't even hear the equalizer itself, which I found to be a great thing! Considering the specifications, functions, versatility, and cost, this is one of the most complete and best-engineered pieces of gear I've seen lately.

While testing the xfilter 500, I felt elysia created one of the finest equalizers I've ever heard. It has everything necessary to become an industry reference. All in all, this is definitely the type of gear that serious professionals will want to have as it will pay for itself, due to its excellent sound quality, adaptability, and build quality. And if you already own an xpressor, pairing the xfilter 500 with it creates one of the most impressive channel strips available.

For more information, visit www.elysia.com. The xfilter 500 can also be seen and heard on YouTube (www.youtube.com/elysiaTV).

The xfilter 500 is shown on the API Lunchbox rack for which it was designed, side by side with elysia's other 500-series products, the xpressor 500 universal compressor and the nvelope 500 impulse shaper.


After the xfilter 500, elysia expanded its rack series with the xfilter true stereo equalizer in a 19" format. Both XLR and balanced 0.25" inputs and outputs are provided.


Showing the two exact circuit boards found on the 500 series unit, the new xfilter rack sports a high-grade linear power supply with shielded toroidal transformer. It is housed in a super-lightweight yet sturdy full-aluminum chassis.

Tech Talk with elysia's Ruben Tilgner and Dominik Klaßen

By Miguel Marques

elysia was founded in Nettetal (Germany) in 2005 by Ruben Tilgner and Dominik Klaßen to make their personal sound philosophy come true. Pictured are Ruben and Dominik with Tom Van Den Heuvel from elysia France.

***audioXpress*: elysia has a history of designing hand-built discrete gear. Was it difficult to create products by using surface-mount designs (SMDs) and still maintain the quality and audio integrity for which the brand is known?**

elysia: Using SMD technology in production has been a new process for us with the introduction of the xpressor 500. As with every new process, you need to learn a couple of things and gain some experience to get things right from the start. All SMD components are placed by a robot, one after the other, in a single production step, so we also needed to organize ourselves better in terms of purchasing components and scheduling production.

In our opinion, audio quality is not so much a question of through-hole technology (THT) vs. SMD. What really makes the difference is if you have a good circuit design or not, and if you have chosen the best components for it or not. In terms of reliability, we can now say we have very good experiences with SMD technology, as our products based on it are even more reliable than the THT ones had already been.

***audioXpress:* What were the biggest technical challenges you encountered when designing the xfilter 500?**

elysia: The biggest challenge in this regard was to match both channels of this equalizer as well as possible. There is a good reason that not many stereo-linked equalizers occupy the market. Some stages in the filter need four resistance layers and this means you need a rotary switch with four layers and enough positions. These are rare and very expensive. The other option is a four-layer potentiometer.

The problem with four-layer potentiometers is that each individual layer comes with its own component tolerance. This means while layer one could be at the minimum tolerance end, layer two could be at the maximum of the tolerance range, or anything in between. It is a nightmare in terms of stereo matching.

For this reason, we built a little machine that uses a microcontroller running our software to read and evaluate every single two- and four-layer potentiometer we buy. As a result, only 65% of the dual-layer potentiometers and only 50% of the quad potentiometers we buy go into production. On a side note, we are also using special low-tolerance film capacitors for even better stereo accuracy.

***audioXpress:* Is designing gear for a 500 series harder than building standard 19" rack products? Apart from the space available—which is obviously a lot less on the 500 series—is it difficult to create high-bandwidth and high-headroom gear like yours for the 500 series?**

elysia: Space is certainly a consideration when building a 500-series product. But we have another advantage using an SMD. When you look at the nvelope 500, it's definitively the most technologically complex analog processor we have ever built. So many stages, so many functions, so many components—and a complete channel still fits on a single 500-series board.

Bandwidth is really not a problem. Take the xfilter 500 for example, which has a frequency response of less than 10 Hz to 400 kHz at –3 dB. Headroom is lower than with our "bigger" units running at ±30 V, but only those who are running their DACs at very hot levels are really affected by this. You might have to be a little bit more careful, but once your levels are adequate, it works well.

***audioXpress:* elysia has a long history designing compressors. In your experience, are equalizers more or less difficult to design and build than compressors?**

elysia: You can't really say that a compressor is more difficult to build than an equalizer, or the other way around, because both types of processors have their individual challenges to master. The audio path needs to be pristine in both, and that's why we have every single one of our amplifier stages operate in constant Class-A mode.

When you design a compressor, the two most important parts are the detector circuit and the gain reduction element. If you look at the many different topologies in the market, you can already tell what a broad and interesting field this is.

The focus in development when it comes to equalizers is the amplifier stages for the individual filter bands and the tuning of the individual filter curves in combination with the overall voicing of the equalizer.

***audioXpress:* How does being a German company affect your builds and gear designs? Can you manage to source parts and components in Germany or do you have to go abroad for most of it?**

elysia: Our products are entirely developed, built, and tested in Germany, and we do a good part of it. Speaking of parts and components, the majority of our metalwork and all the transformers are made in Germany, while most electronic components are obviously produced in the East. We have a good industrial infrastructure here, so finding the right partners to help us during the production process takes some time and patience. In the end, we have a chance to work with some experienced companies. There are also many electronic component distributors, so we only need to buy abroad in a few cases.

The well-known seal of quality, "Made in Germany," actually does mean a lot to us. Our home country has a great history for quality engineering. If you look at our PCBs or the design and accuracy of our housings, you will certainly spot some traces of our heritage here and there.

***audioXpress:* Is the industry and the manufacturing different now compared to 10 years ago when elysia started? What major technological achievements in audio and/or manufacturing have you witnessed during this time?**

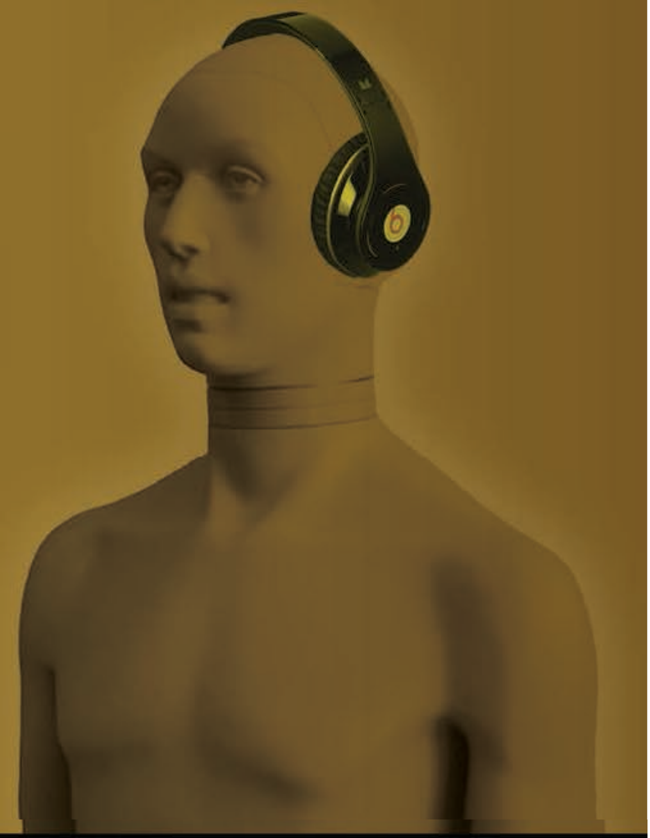
elysia: elysia was founded in 2006, and at this point in time, analog audio technology was already well developed. All the real classics had already established themselves when we hit the market. Today, the pro audio market is certainly in a different situation than in former decades, but does that necessarily mean manufacturing quality has decreased? We don't really think so. Looking at all the great (and sometimes not so great) analog products the past has given us, the true challenge in our book is something else: Trying to come up with something new.

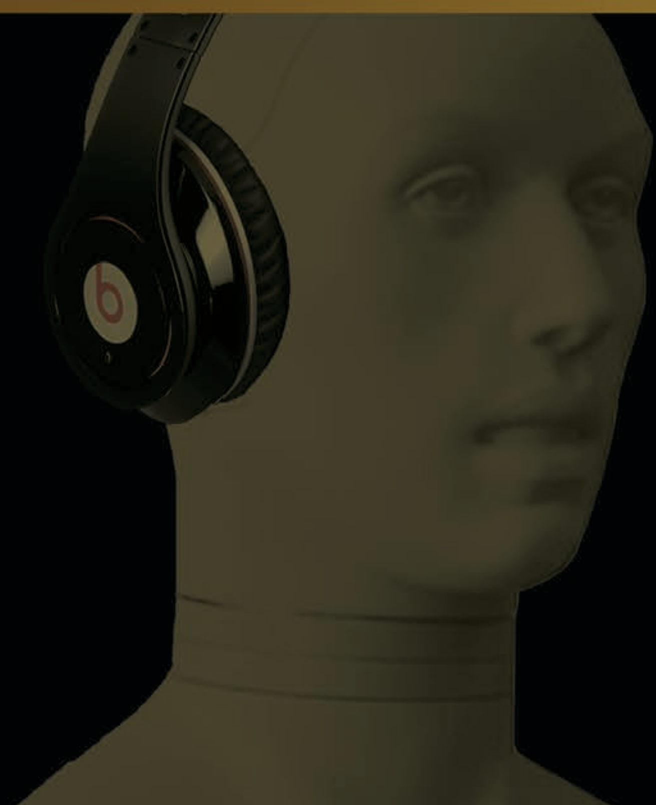
A more dynamic development can be found in the digital segment of our market, but of course, we have to consider that this one is much younger than good old analog, so it seems to have more room for further evolution.

***audioXpress:* Is there any new technology you want to try or are interested in that can be applied to pro audio or general manufacturing? Where do you see the industry going in the next few years?**

elysia: Streaming audio via IP/network is still relatively new, and we are interested in seeing how this technology will develop. This is something so extremely useful and powerful that it is likely to have a great future. As always, there is the question of a possible common standard, and reality often makes it more than one instead… still it looks promising.

As for the future of the industry? (laughter) We really wish we knew… the only thing we can take for granted is that we have some amazing stuff in the pipeline.





The Lowdown on Woofers, Subwoofers, and Bass Shakers (Part 2)

Reinventing Low-Frequency Devices to Fit Compact Sizes

By
Mike Klasco and Steve Tatarunis
(United States)

Last month, we reviewed several bass speaker developments, including their use in movie theaters and in 1930s–1950s broadcast and recording studio monitors. Woofers for audiophiles first appeared in the 1950s, including Klipsch's large horn-loaded designs, Altec Lansing's bass reflex, as well as the JBL and the Jensen designs. Nor can we forget to mention Acoustic Research's innovative acoustic suspension compact deep bass speakers.

Throughout the 1960s and 1970s, woofers for home and autosound increased power handling to accommodate higher-powered transistor amplifiers from McIntosh Laboratory, Crown Audio, Marantz, Mattes Audio, CM Labs, Dynaco, Phase Linear, and so forth. One practical aspect of the higher-power amplifiers was that they enabled woofers to be engineered for good bass in smaller enclosures, but it resulted in lower sensitivity. Many design aspects required additional consideration when woofers and subwoofers were reinvented to accommodate compact size and create deep bass. We will now explore some of those changes.

In the early 1970s–early 1980s, one of the authors (Mike Klasco) owned GLI, a speaker company that manufactured discotheque speaker systems. At first, he sourced some of the woofers from Cerwin-Vega. One of the GLI designs had passive radiators with foam surrounds could not withstand the pressure.

However, the Cerwin-Vega woofer foam surrounds were fine. It seemed that many Cerwin-Vega designs used multiple foam layers laminated together. This provided the strength and structural integrity to withstand the pressures within the enclosure without buckling.

In the mid-1980s, Electro-Voice toughened its surround techniques in the manifold concert-sound subwoofers, and placed four modified EVM woofers into a tiny enclosure not much larger than the woofer chassis themselves. One of this tight packaging's benefits was mobility. The close coupling of these woofers also resulted in the woofer's outputs adding more in phase and in power.

Each time you double the number of woofers you receive approximately 3 dB, but when the woofers are close together you can reach almost 6 dB, at

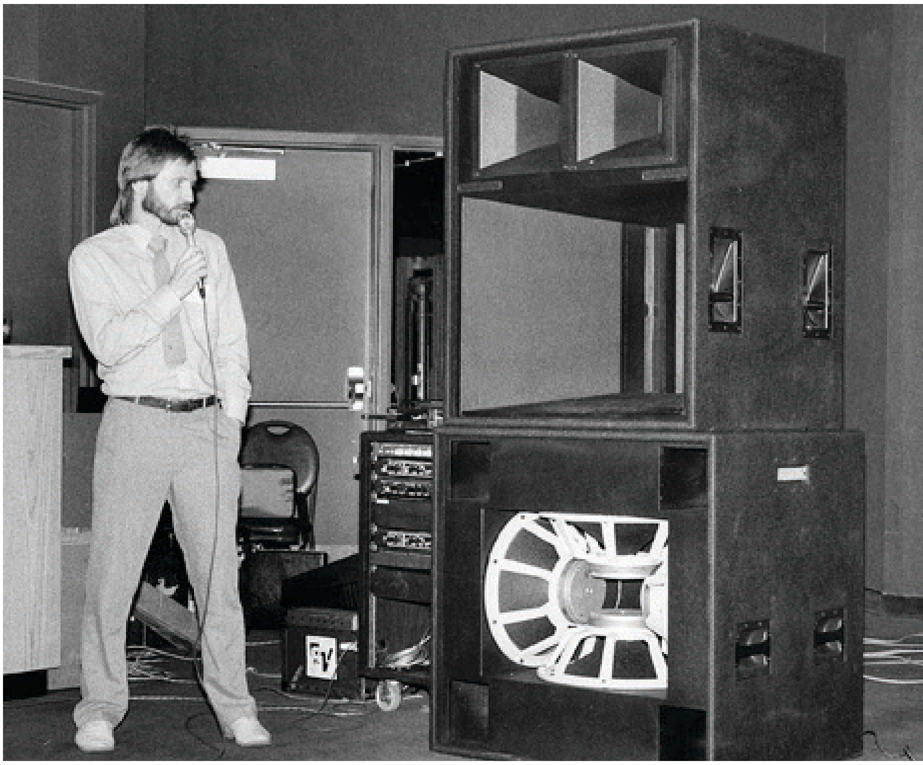


Photo 1: Electro-Voice engineer Dave Carlson introduces the Manifold system at a trade show in the late 1980s. (Photo courtesy of Electro-Voice)

Photo 2: JL Audio offered its big automotive subwoofers with heavy cones and high excursion. (Photo courtesy of JL Audio)

least at the extreme bottom end. The closer the woofer coupling, the higher the bass frequency ideal matching you are able to achieve.

In the late 1980s, Electro-Voice first introduced Doug Button's brilliant EVX-1500 and EVX-1800, which offered 2″ peak-to-peak excursion and even larger clearances behind the voice coil to eliminate bottoming damage during overexcursion (see **Photo 1**). These woofers had many innovations such as fiberglass spiders inside the voice coil, carbon fiber mixed into the cone, and a heat wick self-cooling design.

However, not all these techniques were completely ready for field use. Soon after their introduction, Electro-Voice changed direction and began offering a less radical product series, the EVX-150 and the EVX-180. These transducers are still produced today. They are primarily used in clubs and concert sound.

When surrounds were reformulated, designers needed to address their ability to withstand buckling and flapping from internal box pressures and also distortion reduction. In the 1970s, the British Broadcasting Corp. (BBC) conducted significant research on speaker distortion. (It was on a quest for the ultimate monitoring speaker.) The BBC found that at lower sound levels distortion would rise due to the lack of pressure in the enclosure.

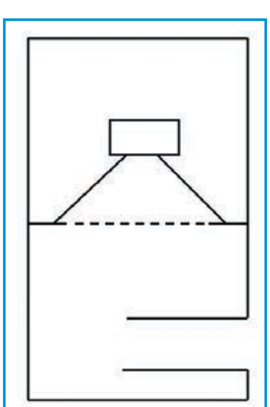


Figure 1: The diagram of a typical bandpass subwoofer enclosure is shown here.

At medium levels, the increased pressure within the enclosure would keep the surround taut enough and reduce harmonic distortion. If enough pressure (and sound level) built up, the distortion would eventually increase. There were a few interesting surrounds designed to counteract buckling forces, such as a Hitachi's V cross-section corrugated surround and Celestion's dimpled rubber design, both produced in the 1970s.

Cerwin-Vega's subwoofers shook their way into the movie market with the introduction of Sensurround in movies such as *Earthquake*. Sensurround produced loud low-frequency sounds through large subwoofers. In the 1980s, the compact disc (CD)-enabled deep bass was no longer limited by the ability of a phonograph record stylus to track a groove (not to mention acoustic feedback issues) and producers could add more low-frequency content to recordings. DVDs with surround sound included a low-frequency effects (LFE) channel for the subwoofer. During the 1990s, subwoofers also became increasingly popular in home stereo systems, custom car audio installations, and PA systems.

Many creative speaker designers focused on the autosound woofer aftermarket in the 1970s and 1980s. JL Audio applied the techniques of Acoustic Research, Cerwin-Vega, and others to the car audio environment. Big woofers with heavy cones and high excursion in tiny spaces in cars put JL Audio on the map (see **Photo 2**). One of JL Audio's innovations was to use a spider spacer that moved the spider further away from the bottom of the speaker frame enabling increased spider diameter to increase the maximum possible excursion of its subwoofers. The cones were reinforced and other innovations were implemented to prevent the cones from crunching due to the strong speaker motors in tight volume enclosures. This resulted in high pressures within the enclosure. Obviously this equation worked well for JL Audio, which has become an established market leader for autosound. JL Audio's audiophile subwoofers are also held in high esteem.

Around the mid-1980s, Earthquake Audio started designing and buying speakers from Sammi Sound Tech Co. in Korea. Earthquake Audio wanted big surrounds and stacked magnets and large-diameter spiders. Sammi Sound tooled and fabricated its own cones, surrounds, frames, and spiders with guidance from Earthquake Audio. For the surround, instead of laminating layers of foam, they just used thicker and higher performance foam. It was not just the thickness that was being scaled, but also the width and height of the foam rolls to enable longer and more linear excursion. Eventually Earthquake Audio applied these techniques to its line of compact high output home theater subwoofers. Soundstream was another autosound subwoofer pioneer that offered some extreme excursion drivers (e.g., the 1993 SS-10R woofer).

In the early 1980s, Velodyne's first effort was the ULD-18, a home theater subwoofer that used servo control to enable a big woofer to have extended bass in an enclosure not much bigger than the woofer itself. Dave Hall, the founder and inventor of the high-gain/ultra-low distortion servo system, based his design on an extreme variant of an acoustic system in which the enclosure was far too small for the woofer. In this case, an 18″ woofer was used in an enclosure about half the optimal internal volume and the servo system corrected for both the bass response and some of the driver distortion.

The audio community first embraced acoustic suspension as a path to more linear reproduction (low distortion) compared to bass reflex. Acoustic suspension does work with one exception: extremely small enclosures with larger woofers operating at very high excursion, making the air nonlinear. In 1985, the Velodyne ULD-18 reached these acoustic output and intense air pressure extremes within the enclosure. Velodyne then integrated a sensor onto the voice coil to monitor the voice coil position. The sensor could detect where the voice coil was actually positioned vs. where it should be positioned and corrected the error on the fly.

Velodyne, a leader in the home theater subwoofer business, helped established the home subwoofer market. In the late 1980s, Velodyne used switching amplifiers to deal with the need for higher power without excessive heat (fan noise is not acceptable in consumer applications). A wave of bandpass enclosures hit the market in the 1990s, although the complexity of construction, less than tight impulse response, and patent issues, have dampened enthusiasm for this approach (see **Figure 1**).

Speaking of patent issues, in the late 1990s Bob Carver's Sunfire introduced a compact home theater subwoofer that output a lot of bass (see **Photo 3**). It quickly became popular. Sunfire's design used the large-surround/super-long voice coil subwoofer that had been evolving in aftermarket subwoofers for cars from JL Audio, Soundstream, Earthquake Sound, Alpine Electronics, Cerwin-Vega, and others. Carver filed several claims with the US Patent and Trademark Office. This was the source of much agitation in both autosound and home theater subwoofer businesses. There are many claims in Carver's patents. Perhaps this is an oversimplification, but it would seem that any further upward scaling of voice coil length, surround size and stiffness, and excursion in general were going to be Carver's property. After more than a year of lawsuits, the Federal Court in Seattle, WA invalidated all Carver's patent claims asserted against Audio Products International (API).

During the two-and-a-half-week trial, a jury considered Carver's allegations that API infringed certain claims of two of his US patents relating to small-box subwoofer technology. After the jury's verdict, the court entered judgment in API's favor on April 15 2003. Carver also failed in an attempt to have the judge overturn the jury's findings and subsequently declined to appeal the decision.

Thanks to all the dedicated audio professionals, today's subwoofer designers have the foundation to achieve good results using tight volume enclosures with high excursion woofers in home theater and autosound applications. Next month, we will tear down a couple of subwoofers and discuss all the components and some of the design decisions needed for a balanced subwoofer design.

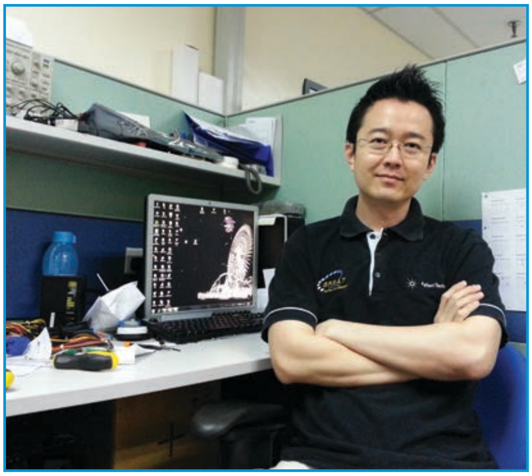
Photo 3: The Sunfire is a compact home theater subwoofer that produces a lot of bass. (Photo courtesy of Sunfire Corp.)

DIY Audio Appeals to Applications Engineer

An Interview with Ken Heng Gin Loo

By
Shannon Becker
(United States)

Ken created the diy-audio-guide.com website because of his interest in DIY audio, in particular tube amplifiers, Class-T amplifiers, NOS DAC, high-efficiency loudspeakers, and high-quality audio reproduction.



SHANNON BECKER: Tell us about your background and where you live.

KEN HENG GIN LOO: I'm a Malaysian Chinese. I spent a wonderful childhood in a small and peaceful town called Taiping, Perak, up North in Peninsular Malaysia.

I'm fortunate that my father could afford to send me overseas to the United Kingdom to pursue a bachelor's degree after locally earning my diploma in Electrical and Electronics. Now, I'm a graduate in Electrical and Electronics Engineering (Honors) from the University of Manchester, UK.

I currently live on a beautiful tropical island named Pulau Pinang (aka Penang) in Malaysia. It is an urbanized and industrialized state that houses several the multinational corporations (e.g., Intel, Agilent, Motorola, Altera, National Instruments, etc.). Yet, it retains its historical heritage and it is one of the UNESCO World Heritage sites. Downtown is filled with historical sites, excellent local delicacies and seafood, and many beachside resorts.

I live in a terrace house with my lovely wife Brenda and 3-month-old baby girl Ember. In the past, I worked as an engineer for several multinational corporations. Now I work at Intel (for more than eight years) as an applications engineer. I focus on customer-enabling team program management, working with high-speed electronics applied in tablets, notebooks, and PCs.

SHANNON: How did you become interested in audio electronics?

KEN: I was "trained" in hi-fi very early, way back in my primary school years. I'm fortunate to have a hi-fi enthusiast as a father. He plays vinyl, and previously used cassettes before moving to CDs in the 1990s. I was brought up (or spoiled?) with high-quality stereos since I was a youngster.

I started to like tube equipment when I heard the combination of a Unison Research Simply 2 and a B&W 601S2 speaker in a shop in Hsinchu, Taiwan. I seriously started to create DIY audio projects when I was working in Taiwan in early 2000. A friend, who was also a tube dealer, introduced me to his DIY 300B tube amplifier that was driving a pair of vintage Tannoy 15" dual-concentric speakers. It was the Tannoy GFR, if memory serves me right. The sound they produced was made in heaven and no setup that I'd encountered at that time was close to producing what I heard that day.

I was hooked on the glowing tubes. I started reading about vacuum tubes online, in books, and in magazines (e.g., *Sound Practices, Audio Amateur, audioXpress,* etc.). I wanted to learn more about vacuum tubes so that I could build a sound system of my own. I also started "wasting" money collecting NOS tubes for my future projects. Now, I have more tubes than I will ever need in my entire lifetime. Maybe if my daughter inherits my interest in audio technology, she will find a use for them. Who knows?

SHANNON: Describe your first personal project. Why did you build it? Is it still in use?

KEN: Tube amplifiers were still uncommon in those days and purchasing one was costly. It was definitely out of my budget when I was a young engineer. Back then, an ordinary tube amplifier alone was two or three times my monthly pay. Since I have knowledge in electrical engineering and electronics (well, sort of, since I studied all solid-state electronics), I thought I would attempt to build a tube amplifier. I thought making a tube amplifier from scratch certainly would not be difficult for me. So, I started building tube amplifiers because the DIY methods made them accessible to me.

A Taiwanese friend assisted me with my first build. He did most of the design work and I did all the soldering since I was still very new to DIY tube amplifier construction. It was a two-stage all triode amplifier with a 300-B direct heat triode driven by a 5842 miniature triode. The sound was sweet and warm even though it was (un?)matched with a small B&W 601S2 bookshelf speaker. (I still have this pair of speakers today!) They are definitely 10 times (exaggerated!) better than all the entry solid states I owned or bought when I was in the UK.

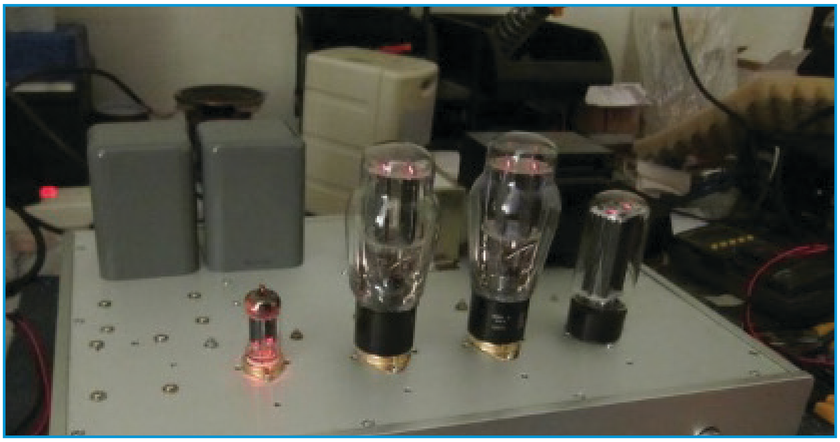
My true first 100% built-by-myself project was an all Russian Reflektor 6C45PI with a 6C33CB triode mono-block tube amplifier. I read an article about a 15-W SE 6C33C-B amplifier that Erno Borbely published in *Glass Audio* magazine. His article sparked the idea for my own project. I used his design to build my own 6C33C-B amplifier circuit.

"I'm a mischievous youngster! I'm an engineer! I can do much better than him!" So I thought. Well, it didn't go that well.

SHANNON: What kinds of audio projects do you build? Can you share some of the challenges involved with the designs?

KEN: I build speakers, pre- and power amplifiers, DACs, subwoofers, mains filters, and almost any kind of audio-related gadget. Some of the projects I have built over the years include: Class-D amplifiers with Philips and Tripath integrated circuits, Gain Clone clones (LM3886 and LM1875), Fostex FE167E bass reflex bookshelf speakers out of real wood, a Fostex FE167E in TQWT enclosure, an Altec 640D in Altec 620D cabinet, several 300B SE amplifier variations, 45 SE amplifiers, 6B4G amplifiers, an 1H4G preamplifier, a 6SN7 preamplifier, a 5687 preamplifier, a 6C45PI SPUD, a 5842 SPUD, an EL34 single-ended amplifier, a Tannoy HPD385A active crossover, and many more that I can no longer remember.

I still have some of the completed projects. I've also posted a few of them on my website (www.diy-audio-guide.com). DIY audio is challenging in many aspects, especially if you want your designs to sound really good and be reliable. To get something to work is easy. To master it is rather difficult.



Among his many projects, Ken built a 2A3 amplifier. However, instead of a 2A3 tube, he used the double plate NOS 6B4G because it has the same electrical characteristics as a 2A3 tube. The main difference is that the filament uses 6.3 instead of 2.5 V.

Some of the most challenging areas include:

Aesthetics—I admit this is one of the challenges I always face. I do see that there are a lot of DIY designs online that look fantastic, almost as good as commercial designs. But for the projects to look good, you need to spend significant time, effort, and money from the design's start until the end. However, my bias would lean more toward sound than looks. For my projects, I definitely spend more time on the design and the components rather than the finished look.

Test and Measurement—This is something I find really challenging on financial and knowledge terms. My daily job includes testing and measuring computer motherboards for power, signal integrity, compliance, eye diagram, and so forth. I use a lot of test and measurement equipment (e.g., multimeters, LCR meters, oscilloscopes, spectrum analyzers, and various other meters). This equipment is expensive and not something an ordinary DIY guy can afford to purchase for personal use. An oscilloscope can cost more than a house in Penang.

Measurement Methodology—This is a topic on its own. A different methodology or setup yields different

Ken used this chassis for a hybrid amplifier design. The front plate and heatsinks are made of aluminum. All the holes are pre-drilled and the chassis accessories are supplied as a package.

Siegfried (Sigi) Maiwald of Wuppertal, Germany, shared his multi-cellular horn project on Ken's DIY website.

results and that is sometimes misused because end users occasionally deviate from the figures/specifications that matters the most. Audio design is an art. Or perhaps it is better to say audio design is a black art? When I was a young graduate, I thought that if I got the circuit right, good sound would follow. It is not that easy! Everything matters, from component selection to the layout.

I'm often amazed that some people think tidy wiring equals good sound. This will not guarantee good sound, but it does help future troubleshooting.

Separating Truth from Fiction—One thing I personally do not like is trying to differentiate among the hype or claims with hidden/personal agendas. I was, and sometimes I still am, tricked into buying something (DIY parts/components) that does not perform as claimed. There are many out there. So, beware!

Last but not least, the biggest challenge for all DIY audio hobbyists is that DIY audio projects often involve carpentry and electrical/electronic, which can sometimes be dangerous. You must work with sharp objects, live electrical connections, and tools. Take precautions and be safe.

SHANNON: What has been the most creative project you've received on your website?

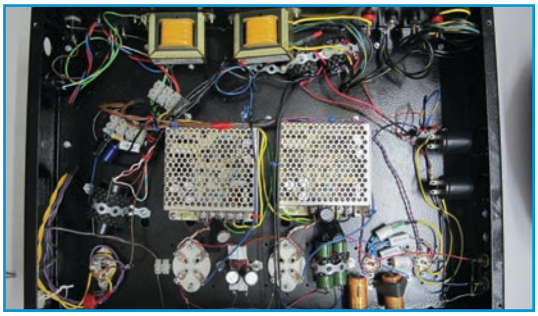
KEN: I received these two amazing DIY speaker and DIY tube amplifier projects from Siegfried (Sigi) Maiwald of Wuppertal, Germany. One day, I met Sigi via the website's feedback form and we became good friends due to our DIY audio hobby.

Sigi's approach to DIY audio, his attention to detail and workmanship, not to mention his energy (and strength to manage such a humongous speaker), and spirit are simply outstanding.

In terms of creativity, he used a broom as the support/stand for a multi-cellular horn! In addition, he used the "УO186" as an inexpensive substitute for the ultra-rare and ultra-expensive RE604, which is definitely cool. I would not have known you could do that. Too bad the project suffered at the end due to a mishap.

You can check out his projects (write-up and photos) at www.diy-audio-guide.com/sigi-audio-setup.html and at www.diy-audio-guide.com/RE604-tube-amplifier.html.

SHANNON: With all the products that are available, why do you think audiophiles continue to experiment and build their own equipment?



The DIY 300B amplifier circuit design and components use a Tung-Sol 5687WA military tube as the pre-amplifier and driver stage. The power tube is JJ 300B and the power supply uses a RCA 5U4GB full wave rectifier tube.

KEN: DIY audio is one of those continuing trends. Fanned by the increasing price of audio equipment, it remains popular among the DIY-audiophiles. Everyone wants a piece of high-end equipment but the disproportionate price vs. performance and the return of investment places many high-end products out of reach for the general community.

Cost aside (DIY is not inexpensive either!), I'm sure DIY audiophiles will continue to design and build because of the satisfaction and enjoyment they receive when listening to their own creations and masterpieces! I am proud to say that I made most of my home audio gear!

The Oldest Tool in the Modern Acoustician's Toolbox

By
Richard Honeycutt
(United States)

When most people hear the words "acoustical treatment," they immediately think of the fuzzy stuff on the walls that deadens a room's sounds. The fuzzy stuff acts as acoustical absorption. While it's certainly not the only tool in the modern acoustician's tool box, absorption was the first tool to be scientifically quantified.

In 1895, Harvard University's Physics Department was asked to improve the school's Fogg Lecture Hall, which had impossibly bad acoustics. The senior faculty considered the assignment as impossible as the acoustics and passed it off to a young professor named Wallace Clement Sabine.

After carefully listening to the difference between Fogg Lecture Hall and rooms considered more acoustically acceptable, Sabine concluded that the critical factor was the time required for a sound spoken at a normal conversational level to decay to inaudibility. He called this the room's reverberation time (RT). Fogg Lecture Hall was found to have a 5.5-s RT, which was long enough for 12 to 15 words to be spoken before the first word faded away. By comparison, good lecture halls had RT's below 1 s.

Working with a team of his students, Sabine measured the effects of seat cushions, rugs, and people on Fogg Lecture Hall's RT. He understood that sound pressure level (SPL) is closely related to air molecules' kinetic energy and by subjecting the air molecules to friction—such as that provided by upholstery padding, fabric, or rugs—the kinetic energy would be absorbed more rapidly and the sound would decay faster. By applying acoustical absorption to Fogg Lecture Hall in the correct amounts and locations, Sabine succeeded in markedly improving the lecture hall's sound. As a result of his efforts, he was selected as the acoustical consultant for Boston's Symphony Hall, which is now considered one of the world's finest concert halls.

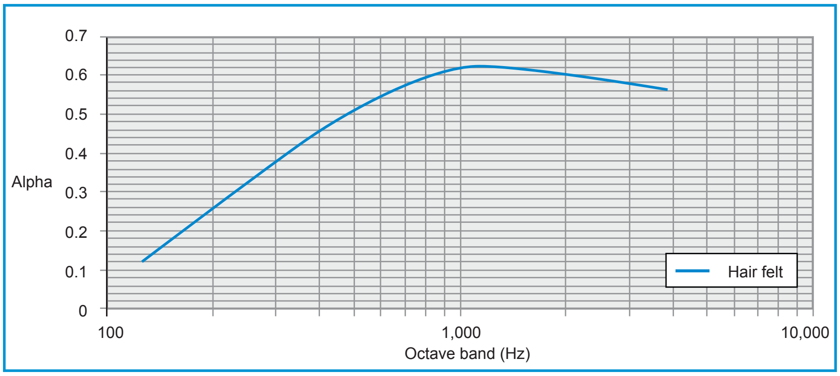


Figure 1: The acoustical absorption of hair felt varies with frequency.

Acoustical Absorption

When a sound wave strikes a material, some of the sound energy will pass through the material, some will be absorbed, and some will reflect back. The acoustical performance is quantified by the acoustical absorption coefficient, symbolized by α (alpha). Alpha can have a value anywhere from zero (completely reflective) to one (completely absorptive). It is listed as a decimal fraction. For example, the alpha of a 0.375″ pile carpet laid on concrete is 0.21 (21% absorption) in the 500-Hz octave band. Most materials' alpha changes greatly with frequency (see **Figure 1**). For example, the alpha of 0.375" carpet on concrete varies from 0.09 in the 125-Hz octave band to 0.37 in the 5,000-Hz band.

For a quick idea of a material's absorption in the most important voice frequencies, materials are rated with a noise reduction coefficient (NRC). The NRC is the average, rounded to the nearest multiple of 0.05, of the material's alphas in the 250-, 500-, 1,000-, and 2,000-Hz octave bands.

Sabine's analysis of the effects of acoustical absorption led to the development of his famous equation that enables the prediction of a room's RT from knowledge of its physical dimensions and its acoustical absorption:

$$RT = \frac{0.161\ V}{\sum S\alpha}$$

where V = the room's volume in cubic meters and the denominator of the fraction signifies the summation of the products of the area (S) and absorption (α) of each material. The product Sα is the acoustical absorption, measured in sabins.

Even though many acoustical absorbers resemble thermal insulation, and in some cases may act as thermal insulation, acoustical absorption is not the same as acoustical insulation. Failure to understand this distinction can cause miscommunication. Just as thermal insulation prevents the movement of thermal (heat) energy, acoustical insulation prevents the movement of acoustical energy. Thus, heavy concrete walls are good acoustical insulators and compressed fiberglass is a good thermal insulator, but not vice-versa.

Absorption Material

During Sabine's time, few materials were available for use as acoustical absorption in a room. Most of them (e.g., hair felt, carpet fabrics, and raw wool) depended on the material's internal friction to convert the air molecules' kinetic energy to heat. Many acoustically absorptive materials currently available use the same principle.

Probably the most common acoustically absorptive material now used in buildings is acoustical ceiling tile (ACT). There are several ACT variations. Contrary to lay opinion, not all suspended ceiling tiles provide significant acoustical absorption. In fact, each ACT manufacturer provides a few products that have almost no absorption.

The most common materials used to make ceiling tiles are mineral fiber and fiberglass covered with perforated plastic film. **Figure 2** compares the absorption of mineral fiber and fiberglass ACT. An absorptive ACT panel's surface usually depends on porosity to enable air molecules to enter the material and convert their kinetic energy to heat. Thus, much of the acoustical performance is lost if the tiles are painted after installation.

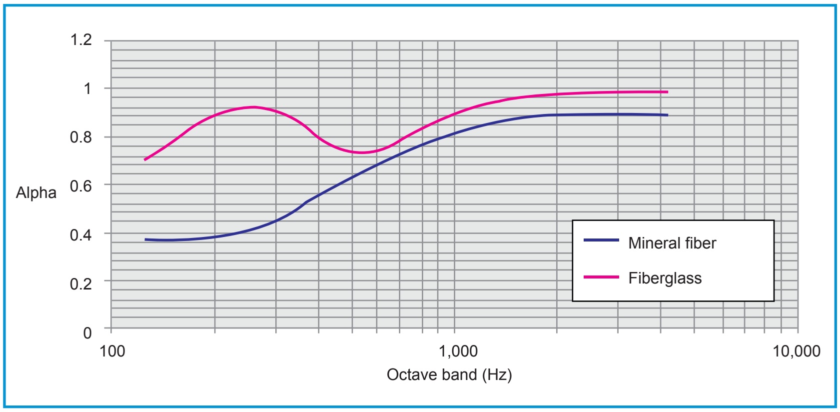


Figure 2: Fiberglass ACT provides more absorption than mineral fiber ACT.

Perforated hard materials such as pegboard are often used with a fiberglass backing to provide acoustical absorption. Contrary to common opinion, it is the fiberglass backing that provides the absorption, not the pegboard.

Using a perforated face material such as wood or MDF with a fiberglass or acoustical foam backing provides good acoustical absorption while offering an interesting visual effect. These products offer an option that pleases the architect's eye and the acoustician's ear (see **Photo 1**).

Any building material provides some acoustical absorption. In some cases, a common material's performance has desirable characteristics that enable it to be used as an acoustical treatment. The most common materials for this purpose are draperies and carpets. While they both can provide useful absorption, in some cases these materials' acoustical value is often overestimated (see **Figure 3**).

Many common building materials have little

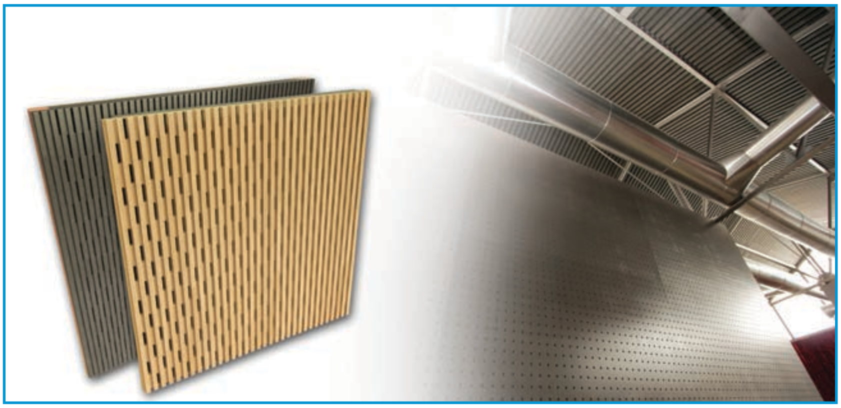


Photo 1: Jocavi's Addsorb uses a perforated MDF face backed with acoustical foam, combining an attractive appearance with acoustical absorption. (Photo courtesy of the Jocavi Group)

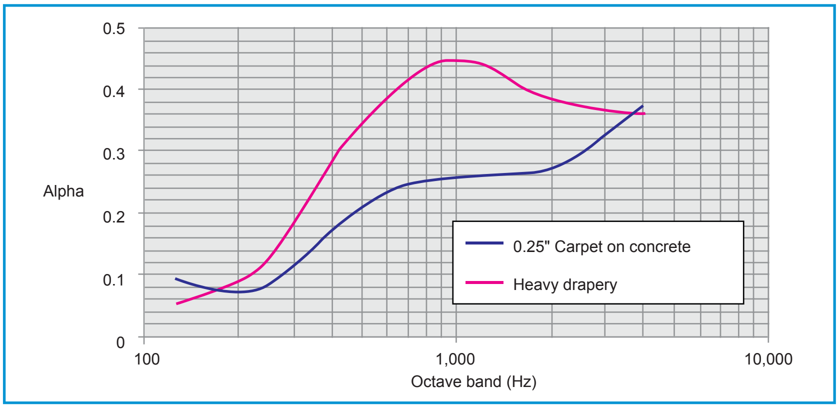


Figure 3: Draperies and carpets provide useful absorption at higher frequencies.

Figure 4: Wall materials can provide significant absorption at some frequencies.

acoustical absorption at low frequencies. This can require the use of thick wall panels and/or expensive ceiling treatments, since thin panels (e.g., 1″ fiberglass or foam) generally provide useful absorption only in and above the 250-Hz octave band. An exception is thin wood paneling and gypsum wall board (GWB). These materials provide diaphragmatic absorption, in which the sound causes the whole panel to vibrate. As a result, sound energy dissipates as heat due to the material's internal friction. **Figure 4** shows the alpha of several such materials. The curves show a wall with 0.5″ gypsum on one side, 3.625″ air space without absorptive batts, and 0.0625″ gypsum on the other side; a wall with a 0.25″ wood panel over a 2.5″ air space; and a wall with 0.625″ gypsum on both sides and a 3.5″ air space with batts.

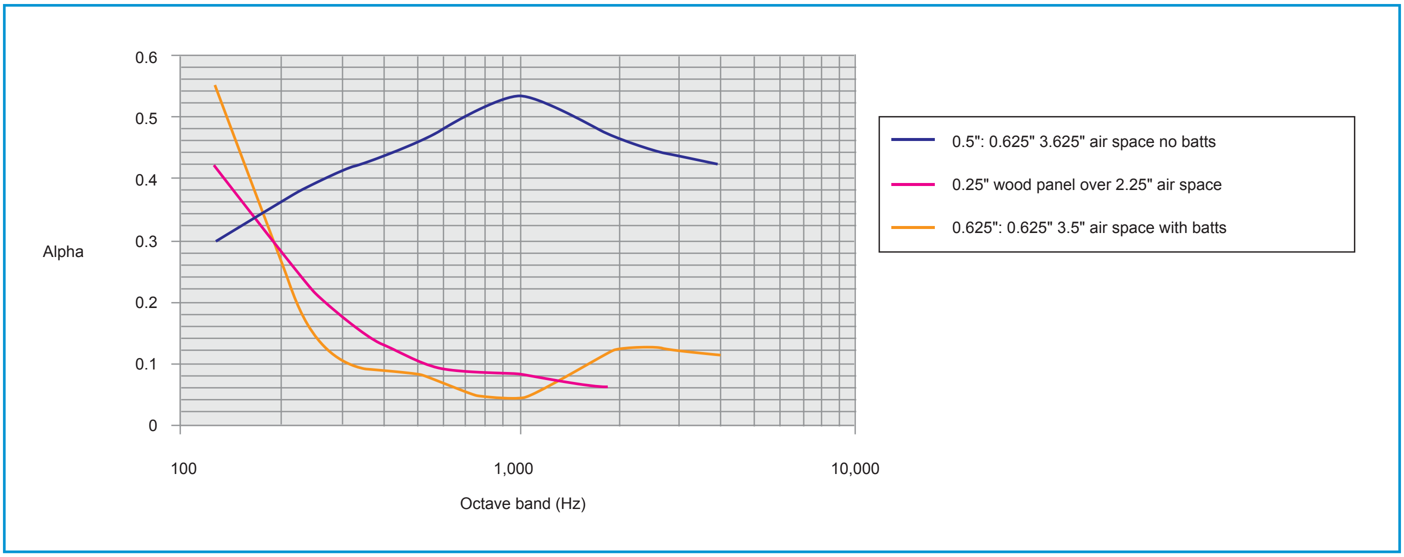
Important variables controlling the wall materials' acoustical behavior are the framing's on-center spacing, the material's thickness, the framing's thickness (controlling the airspace between the wall panels), and the presence or absence of absorptive materials (usually fiberglass batts) in the air space. Since the RT depends on the product of **S** and **α**, a small change in a wall's α can have a large effect on RT, due to the wall's large area. For example, the effect of a 7-m × 3-m wall having a 0.1 **α** is the same as that of a 2.97-m^2 (4′ × 8′) panel with a 0.707 **α**.

Other specialized acoustical materials have been developed that replace common materials. For example, acoustical plasters that absorb sound were introduced in the 1920s. These plasters incorporate fibers or aggregate materials into their formulation. In some cases, these plasters provide alphas in excess of 0.5 at some frequencies, which can have major effects on RT when used for whole walls and/or ceilings. More recently, composite materials—such as Baswaphon, which is made of compressed fiberglass skinned with acoustical plaster—have been introduced. These materials can provide almost 100% absorption over much of the audible frequency range.

Adding Acoustical Absorption

In treating an existing room, one seldom has the option of replacing whole walls. Any added acoustical absorption must be provided by the use of ceiling and/or wall panels. The most common acoustically absorbing wall panels are made of tangled wood fibers, melamine or polyurethane foam, or fiberglass.

Acoustically absorbent wall panels made of tangled wood fibers with a specific mineral treatment are manufactured under the trade name Tectum. Tectum provides a cost-effective, durable option for gymnasia and similar spaces. **Photo 2** shows



a Tectum panel. Tectum's low-frequency acoustical absorption depends greatly on the mounting method. If furred more than 40 to 80 mm of air or fiberglass batts, Tectum is quite an effective absorber down to the 125-Hz octave band. Mounted directly to a hard wall, the 125-Hz alpha is less than 0.1.

Fiberglass panels come in four major varieties: plain, cloth-covered, vinyl-wrapped, and hardened-surface. Plain fiberglass panels are the least commonly used for wall treatment. When plain fiberglass is wrapped with acoustically transparent cloth, it becomes aesthetically attractive with almost no effect to the absorptive properties.

Vinyl wrapping increases low-frequency absorption through diaphragmatic absorption, at the expense of some high-frequency absorption. Hardened-surface fiberglass provides a durable surface of about 2.5 mm of hard fiberglass adhered to a compressed fiberglass substrate. It is impact-resistant and also provides a diaphragmatic effect that enhances low-frequency absorption. **Photo 3** shows the construction of each of these types of fiberglass panels. **Figure 5** shows the absorption characteristics for plain, vinyl-wrapped, and hardened-surface 2" thick fiberglass panels.

Many manufacturers use melamine or polyurethane foams to make acoustical absorption. The foam can be obtained as flat panels or with various sculpted surfaces. Foam must be used in greater thicknesses than fiberglass to achieve the same low-frequency absorption, but it is available in many shapes and colors. It is also lightweight (see **Photo 4**).

Photo 2: This Tectum material is made of tangled aspen-wood fibers treated with specific minerals. (Photo courtesy of Tectum)

In recent years, many manufacturers have introduced microperforated wood acoustical absorbers, in which tiny perforations permit good acoustical absorption, especially in the 500- and 1,000-Hz octave bands, while presenting the appearance of solid wood (see **Photo 5**). Other products are also available as planks, lay-in ceiling tiles, V-groove wall panels, and in other decorative forms.

Tuned Bass Traps

In small rooms such as recording/broadcast control rooms, resonant modes can cause problems by emphasizing specific narrow frequency bands of sound, especially in the frequency range below 150 Hz. Since the room's purpose heavily depends on the engineer's ability to accurately hear what

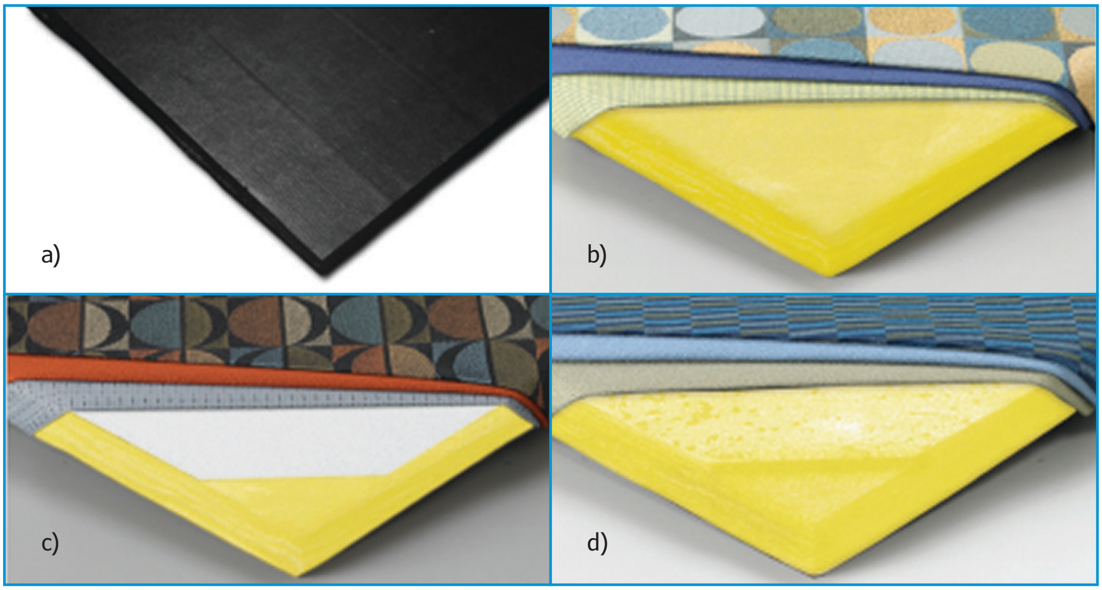


Photo 3: Fiberglass treatment is available in several varieties: plain fiberglass—black "theater board" (a), cloth-wrapped fiberglass (b), vinyl-wrapped fiberglass (c), and hardened-surface fiberglass (d). (Photos courtesy of G&S Acoustics)

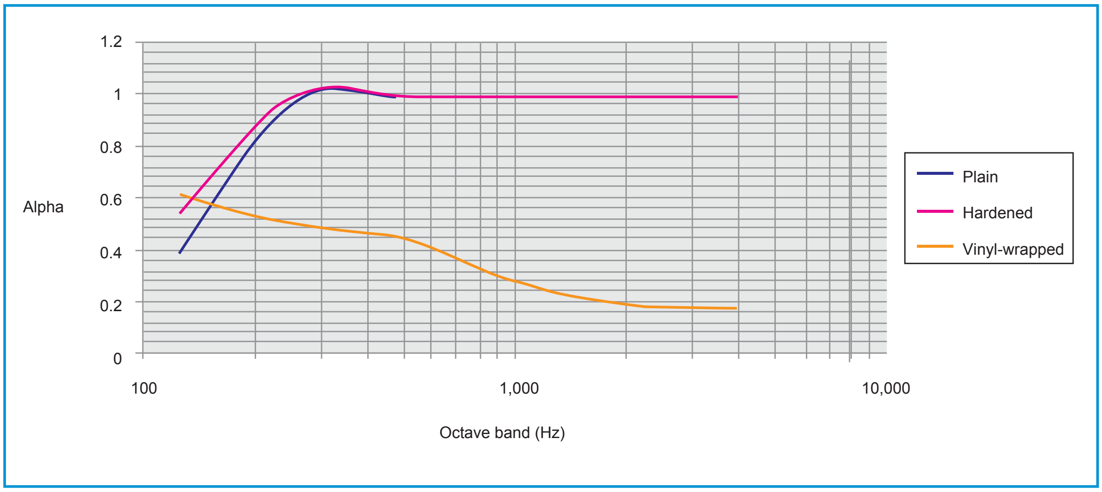


Figure 5: Different varieties of fiberglass panels provide different absorption profiles.

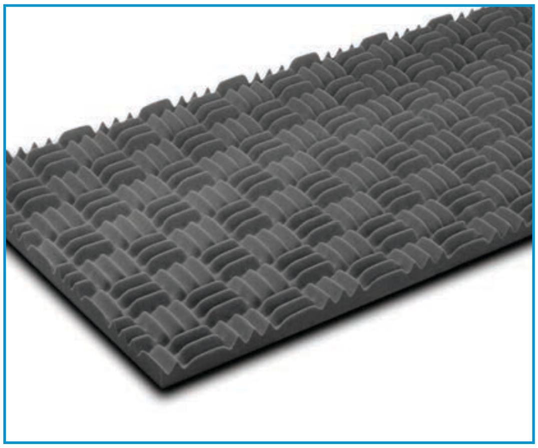


Photo 4: Acoustical foam is available in many colors and shapes. (Photo courtesy of Sonex Acoustics)

Photo 5: This Topperfo Micro perforated panel has a solid wood appearance, coupled with good acoustical absorption. (Photo courtesy of RPG Diffusor Systems)

is recorded or broadcast, having certain notes unnaturally boosted is highly detrimental. Not only is broadband absorption expensive if low-frequency absorption is desired (due to the necessary thickness of 10 cm or more), but broadband absorption does not really address the problem of narrow-band boosting of certain frequency ranges.

The solution most appropriate for this issue is tuned bass traps. Many of these devices work on the principle of a Helmholtz resonator. A Helmholtz resonator consists of an enclosed volume of air that opens to the outside through a neck. A jug played in some forms of folk music is a good example. The Helmholtz resonator is a mechanical mass-spring oscillator. If a mass is placed atop a spring and then displaced, the mass will bounce at a frequency given by:

$$f = \frac{1}{2\pi}\sqrt{\frac{k}{m}}$$

where f = the frequency of oscillation, k = the spring constant in newtons of force per meter of displacement of the spring, and m = the mass.

In a Helmholtz resonator, the spring function is provided by the air's compressibility in the enclosure and the mass is the air's mass in the neck. This leads to the following equation for a Helmholtz resonator's resonant frequency:

$$f = \frac{c}{2\pi}\sqrt{\frac{A}{V_0 L}}$$

where c = the speed of sound in air (345 m/s), A = the cross-sectional area of the neck in m^2, V_0 = the volume of the enclosed air in m^3, and L = the length of the neck in m.

If absorptive material such as fiberglass is placed inside the enclosure, the resonator's sound energy will be converted to heat. Thus, sound energy at and near the resonant frequency can be removed from the room. A Helmholtz resonator used in this way is one form of resonant bass trap.

This form of bass trap is a common DIY project for home studios. In such projects, a shallow box is topped with a perforated material such as pegboard and the box is partially filled with fiberglass. The pegboard's holes and thickness serve as the necks of an array of coupled Helmholtz resonators and the portion of the enclosed volume behind each hole acts as the air volume.

Another type of bass trap uses the diaphragmatic absorption provided by a plywood sheet affixed at the edges to form the top of a shallow box that is loosely filled with fiberglass. If the box is 3″ deep, a 4′ × 8′ piece of 0.25″ plywood will resonate about 110 Hz; 0.125″ plywood, 150 Hz; and 0.375″ plywood, 87 Hz. Using a piece of uninsulated 14-gauge wire as a spacer between the plywood and the box at the attachment points enhances the resonant action. A variant on the panel absorber is the membrane absorber, which also uses diaphragmatic action. Many manufacturers produce bass traps using the Helmholtz or diaphragmatic principles (see **Photo 6**).

Bass traps are not magical: They cannot suck bass out of anywhere in the room. For optimum operation, place a bass trap where the air pressure is greatest. For the lowest-frequency modes (often the most problematic), this location is a corner of wall/wall/ceiling or wall/wall/floor. Wall/wall corners are also effective. The third option—on a short wall—primarily helps with longitudinal modes, which represent sound bouncing back and forth along the room's length. Most manufacturers offer guides as to optimum placement of their bass traps.

Photo 6: This commercial bass trap is designed for use in a room's corner. (Photo courtesy of the Jocavi Group)

The Cathedrals

By
Ken Bird
(United States)

My Castle speakers ("Castle Tower Omni Speakers," *audioXpress*, September 2011) were so much fun to build I thought I would design a pair of speakers with a gothic architectural theme. I call them the Cathedrals, because I was attempting to duplicate the equilateral arch used in European medieval cathedrals. However, the "ruined Abbey" might be a better description.

Photo 1: The completed Cathedral speakers are shown with the speaker grilles in place.

I started the project because I had four very nice 8″ Peerless speakers in my collection that needed an enclosure (see **Photo 1**). At 12 Ω each, they were ideal for a 6-Ω parallel connection, which is close enough to the proposed 8-Ω tweeter and midrange. To construct the enclosure, I opted to use 1″ × 12″ dimensional lumber for all the sides. This produced a 2.1-ft^2 sealed enclosure (see **Figure 1**).

The Drivers

The Cathedrals contain a three-way system with two 8″ Peerless woofers, a 4″ GSR-sealed-back midrange, and a 1″ MCM dome tweeter crossing over at 1,000 and 5,000 Hz. I used 20-gauge magnet wire with capacitors to make the crossover coils. The woofers specifications include a resonance frequency (F_S) of 43, the combine electric and mechanical damping (Q_{TS}) is 0.56, and an equivalent compliance volume (V_{AS}) of 2.7. I chose a GRS 4″ midrange with a 600-to-10,000-Hz range. My tweeter is a 1″ dome unit covering 2,000 to 17,000 Hz. Almost any 8″ woofer with similar specifications would also work well in this type of enclosure (see **Photo 2**). The crossover schematic notes the potential use with 8-Ω woofers.

The Enclosure

The enclosure is constructed from 1″ × 12″ pine boards with the front and rear panels configured into an equilateral arch. The rear panel has a decorative trefoil opening while the front panel's arch houses the dome tweeter. The front and rear panels have 0.75″ dado groves that the side panels fit into making very solid joints once glued and nailed. Two 2″ × 2″ braces are used to reduce cabinet resonance. I was concerned that the "flying arches" would become self radiating, but I have not detected any obvious audio artifacts from them.

Construction

I used a 48″ long front panel to construct the enclosures. I laid out an equilateral arch by drawing a line across the panel that was 12″ down from the top. I measured 0.75″ in from the end of this line on each side. Then, I used a large compass and set the fulcrum on one end of the 0.75″ mark and the pencil on the other 0.75″ mark. I swung the compass to the

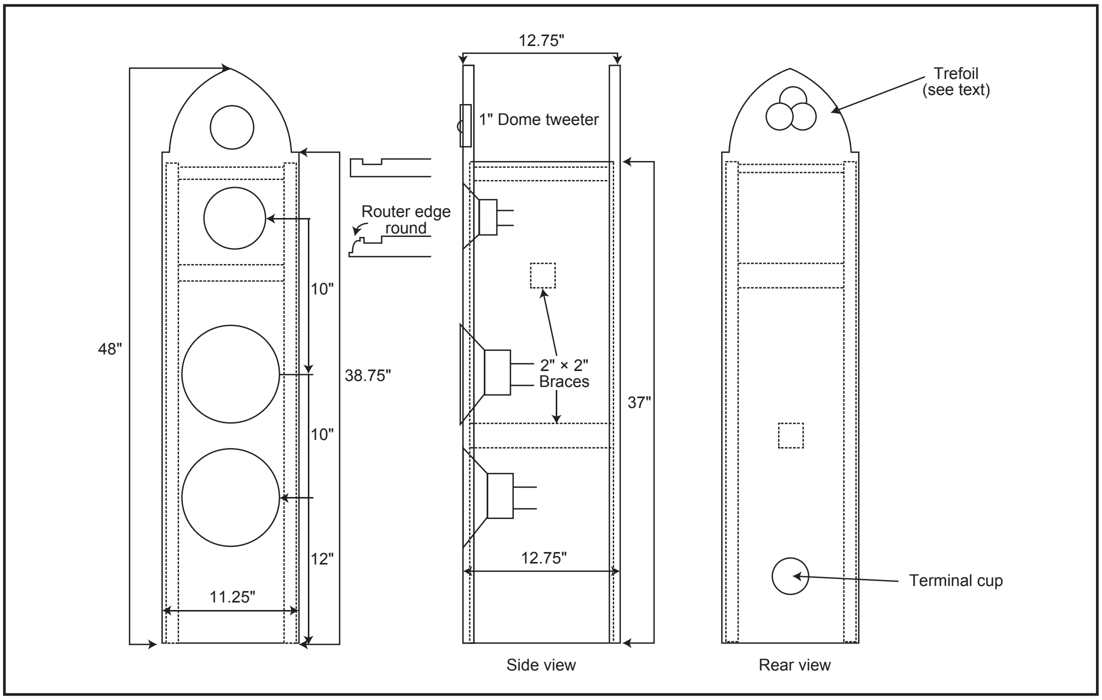


Figure 1: I created a drawing to determine where to fit the Peerless speakers. To construct the enclosures, I used five 1″ × 12″ × 8′ pine (or hardwood) boards and made a 0.75″ Dado groove that was 0.25″ deep and 0.5″ from the edge on the front panel and on the front of the back panel.

top of the panel. I then repeated this procedure by placing the compass's fulcrum at the other 0.75″ mark and drawing another curving line. Once completed, the arch was properly marked.

I performed the same procedure on the back panel. Then, I used a jig saw to carefully cut along the drawn arch lines on each panel. I drew the trefoil first making an equilateral triangle with 2″ long sides on stiff paper. You could also use card stock. I used this for a pattern and centered it on the front panel 3″ above the base line of the equilateral arch (see **Figure 2**).

Next, I used a compass set to 2.″ I placed the compass's fulcrum on one point of the triangle and drew a circle. I repeated this process on each triangle point and created a trefoil pattern. Once the pattern was completed, I placed a thin saw blade in the jig saw and cut out the trefoil. I used a router and a circle jig saw to make the speaker cutouts (see **Photo 3**). I inset the woofers 0.25″ and surface mounted the midrange and tweeter. The Peerless woofers required a 7.5″ cutout. The GSR midrange needed a 4.125″ hole, and the MCM tweeter hole was 3″. If you use other drivers, consult their literature for the proper holes sizes.

I held the side panels in place in a 0.75″ deep dado groove that I made with a 0.75″straight router bit using a guide attached to the router. I adjusted the guide so the groove was 0.5″ from the panel's edge. The groove should end where at the bottom of the top piece (see **Photo 4**). If you have a table saw, use a dado blade to cut the grooves.

Next, I used a rounding bit to route the edges of the back and front panels and the tops of the side panels. I did the same to the trefoil's edges. This resulted in the decorative enclosure that evoked my cathedral theme.

Before assembling the enclosure, I sanded the edges of the side panels so they would fit better into the dado grooves (see **Photo 5**). I used a small brush to apply wood glue to the dado grooves and the side panel edges. After placing the back panel on a hard surface, I inserted the side panels. Keep a damp rag handy to wipe off any glue seepage. You may need a rubber mallet or a block of wood and

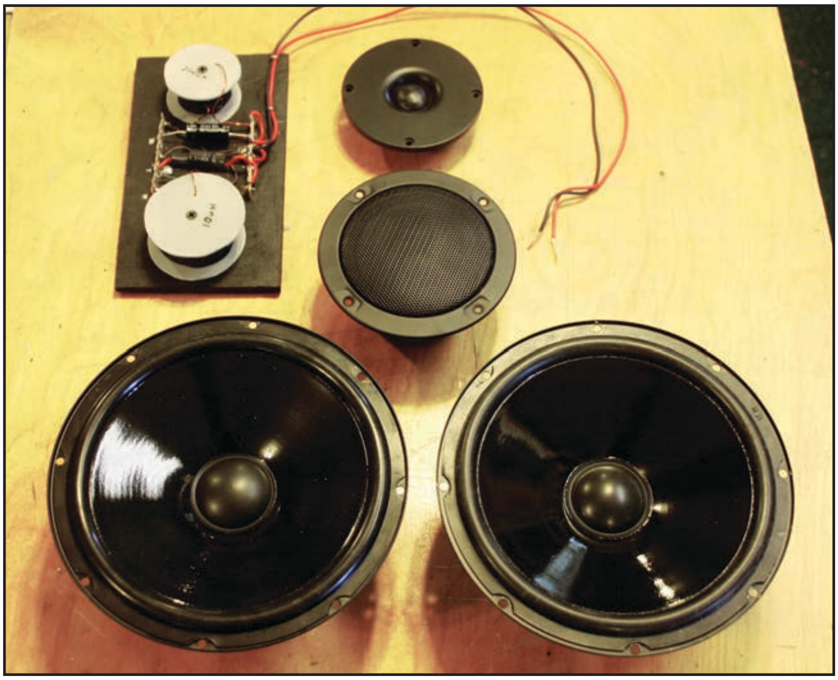


Photo 2: I used four 8″ woofers, two 5″ midranges, and two dome tweeters. The drivers and crossover used in the Cathedrals' project.

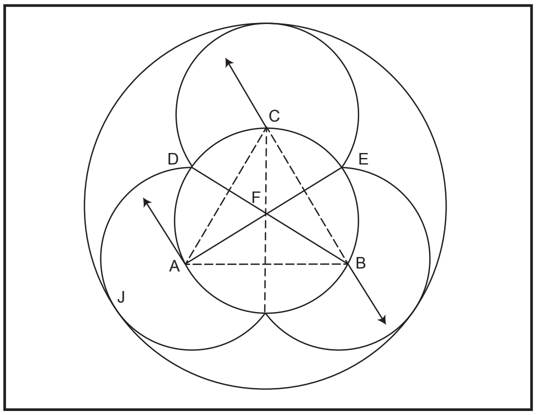


Figure 2: I used a trefoil pattern to position the speakers by placing the compass fulcrum at points A, B, and C on the triangle.

regular hammer to ensure the panels are seated. With the side panels inserted, I caulked the joints and repeated the same procedure for the front panel. I kept the enclosure in the same position and nailed the front panel in place. To prevent the wood from splitting, I used a 0.0625″ bit to predrill the nails holes to a 0.5″ depth. I used 1.5″ finishing nails spaced 5″ apart then I nail punched them 0.125″ deep (see **Photo 6**).

I added two 2″ × 2″ braces using liberal amounts of wood glue and a center-placed 1.625″ wood screw. I mounted the bottom piece flush with the panels and the top or "roof" 0.25″ below the side panels top edge. Before mounting the top, I drilled a 0.25″ hole at the center edge below the tweeter mounting hole. Caulk all the joints thoroughly. I had to hand caulk the top and bottom and I used a small hand mirror to ensure I covered all the seams.

Photo 3: The assembled enclosures are shown in their raw form.

Surface Finishing

In the process of building the enclosures, an audiophile friend, Jimmy Schnakenberg, brought me his 1983 JVC RS 33 receiver and two Altec Series 9 speakers for refurbishing. He immediately liked the looks of the Cathedrals and talked me out of them. Schnakenberg took the raw enclosures to a friend who had worked in the local gunstock industry and had the friend finish them to match his antique armoire equipment cabinet. The Cathedral enclosures were sanded, sealed with a wood conditioner, and stained. A clear top coat was sprayed on them to complete the finish. Obviously, you can choose any finish and technique you desire (see **Photo 7**).

Crossover and Speaker Mounting

I used 20-gauge magnet wire secured from a local motor rewinding shop to build the crossover

Material List for Two Enclosures

The dimensions specify pine or hardwood boards.

Building Materials

- Front and rear panels: four each 11.25″ W × 48″ H
- Side panels: four each 11.25″ × 37″
- Top panels: two each 10.75″ × 9.25″
- Bottom Panels: two each 10.75″ × 9.25″
- Cross braces: two each 2″ × 2″ × 9.25″ long
- Cross braces: two each 2″ × 2″× 10.75″ long
- Wood glue and 1.5″ finishing nails

Speakers

- Woofers: four 8″ woofers (Parts Express, 299-068 or equivalent)
- Midranges: two 5″ GRS-sealed (Parts Express, 292-432)
- Dome tweeters: two (MCM Electronics, MCM 53-1080)

Crossover

Capacitors are all NP at 100 V minimum

- C1 = 4 µF (Parts Express, 027-330)
- C2 = 22 µF (Parts Express, 0127-348)
- L1 = 0.214 mH (Parts Express, 266-808)
- L2 = 0.9 mH (Parts Express, 266-824)
- L2 = 0.64 mH for use with 8-Ω woofers (Parts Express, 266-818 see text)
- Speaker terminal cup (Parts Express, 260-311)

Photo 4: The front panel is shown in detail. Note the dado groves I made for the side panels.

Photo 5: I used a router to make dado grooves in the enclosure.

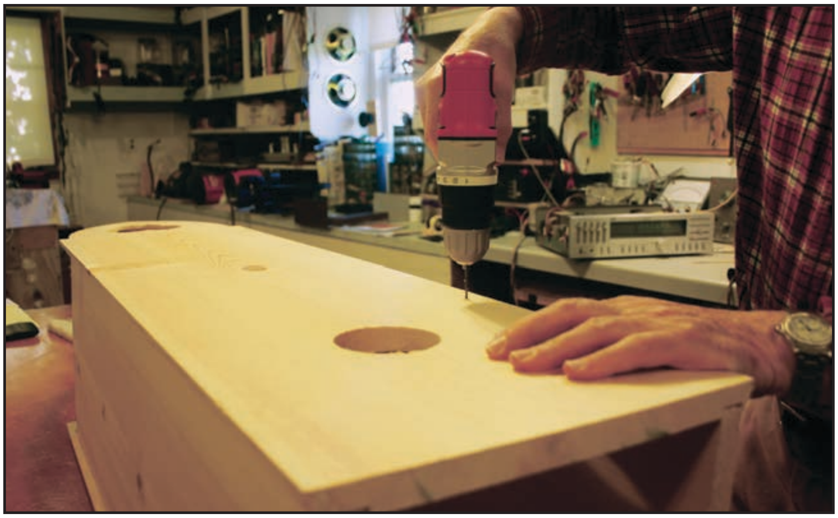


Photo 6: A key component for a polished finish is to drill the starter holes for the finishing nails.

About the Author

Ken Bird has had a 40-year engineering and product management career focusing on consumer, industrial, and telecommunications electronics. Ken retired to his native Missouri and now pursues his interest in building speakers and tube amplifiers, along with model railroading, amateur radio (WOKLB), and digital photography.

coils. Most shops will give you the end runs, spools and all, at no cost. You can also purchase magnet wire from Parts Express or All Electronics.

The coils were wound on a spool made from 0.75″ dowel rod sections 1″ long with the end caps cut from heavy cardboard. Instead of counting the wire turns, I wound a quantity of wire on the spool until it was about a 0.5″ thick. Then using a B&K Precision 878B LCR meter, I measured the coils and added wire or subtracted it until I reached the required value. The latest version of the Dayton WT3 Woofer Tester has a built-in LCR meter that can be used to measure the coils. I secured the finished coils by wrapping electrical tape around them. Then, I mounted them along with the capacitors on a scrap piece of 0.5″ plywood. The capacitors were all 100-V-rated electrolytics and were secured to the base using a hot glue gun. If you don't want to make your own coils, you can purchase them.

I prewired the crossovers using 16-gauge red and black automotive wire soldered to the respective speaker connections and mounted the crossovers on the lateral brace near the woofers using drywall screws. **Figure 3** shows the wiring schematic. The arch-mounted tweeters' rear magnets were painted flat black making them unnoticeable from most viewing positions. The round terminal cup with five-way binding posts used for the system's external connections are not shown, but it is listed in the parts section.

Each enclosure was stuffed with 10 oz of polyester pillow fill. Tease out the fill as you use it to prevent dense clumps of the material from forming.

I used silicone caulking and #4 × 0.5" pan head sheet metal screws to install the woofers and midrange speakers. The silicone caulk ensures a good air seal and if required, makes speaker removal easier. The tweeters did not require any seal in their location.

The speaker grilles were constructed using 0.125" plywood circle frames over which fiberglass window screen was stapled. This material, known as "pet screen," provides good protection to the speakers with no measurable effect on the sound quality. I used 0.5" brad nails to attach the grilles to the enclosures.

Photo 7: The completed enclosures were finished to match a large antique armoire used as an audio equipment cabinet.

Performance

The Cathedral speakers achieve a 0.7 Q_{TC} with an 58-Hz F_C, and a 54-Hz F3, with a pleasing bass roll off. The midrange is well defined and the treble is bright without any harshness. My bride said I should describe the sound as "Heavenly," but the more secular critics described the sound as natural and well balanced for all types of music (from organ to country music). Their unusual design evoked a medieval theme many liked.

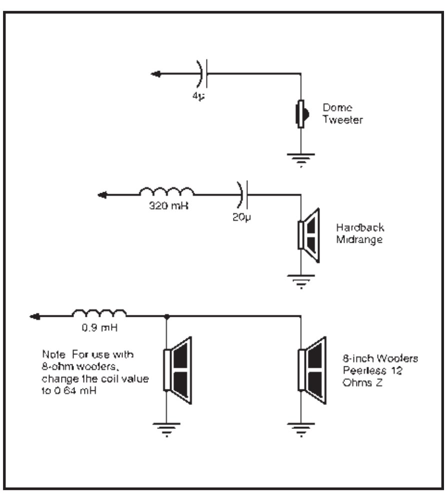


Figure 3: The crossover schematic is shown. I used a terminal cup with binding posts to complete the external connections.

Resources

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K. Bird, "Castle Tower Omni Speakers," *audioXpress*, September 2011, www.cc-webshop.com.

Parts Express, www.parts-express.com.

Sources

Series 9 speakers
Altec Lansing, LLC | www.alteclansing.com

878B Dual-display hand-held LCR meter
B&K Precision Corp. | www.bkprecision.com

WT3 Woofer tester
Dayton Audio | www.daytonaudio.com

R-S33 Receiver
JVC | www.jvc.com
(available from online retailers such as eBay)

Tips to Resurrect a Classic Speaker or Design a New System (Part 2)

A Viable Solution to Speaker Sensitivity Problems

By
Thomas Perazella
(United States)

This is the second installment of a three-part series about resurrecting an old speaker that used some interesting technology. This article will describe the speaker's final construction. I will also offer a solution to a sensitivity issue that impacts not only this design, but possibly the speakers you are currently using.

Photo 1: The completed speaker is shown from an oblique angle with the grille in place.

Photo 1 shows the revamped speaker, which closely resembles the original design. In the original speaker, the midrange and tweeter were mounted on an open vertical baffle that tapered from bottom to top and had rounded edges. That was done to minimize diffraction. At first I thought this section could be used without changes, but that was not the case.

The Top End

The Heil tweeters were mounted to a shelf on the baffle using the original bolts that came with the drivers. When doing some preliminary sine wave sweep testing, I noticed the housing on the tweeters would resonate at approximately 160 Hz. Vibrations generated by the woofers traveled from the enclosure through the baffle even though a rubber pad separated the enclosure from the baffle and the locating pins were isolated by pieces of latex rubber tubing.

To solve the vibration problem, I resorted to an isolation technique I had successfully used with shop machinery. Sorbothane rubber is one of the best vibration absorbing materials. To achieve effective vibration isolation remove as much of the vibrational energy as possible from the system. Don't try to just re-transmit at a different frequency or amplitude combination. Sorbothane is effective at converting vibrational energy to heat. Depending on the durometer the first time you use it, the consistency may seem like a gum rubber eraser or even a squishy rubber version of well-chewed Turkish taffy. For this application, I chose a 0.25" sheet of 30 durometer, which definitely falls into the Turkish taffy arena.

My original question was how to achieve the best isolation from the lower enclosure. Instead of using the Sorbothane at the baffle's base, I decided to isolate the Heil from the baffle. This made the isolation effective against potential woofer and midrange induced vibrations. I decided to use Sorbothane sheets to sandwich the air motion transformers (AMTs) on the top and bottom. I held them in place by making plates with attached threaded rods to clamp them to the platform on the baffle (see **Photo 2**). I placed the rods from the clamping plates through holes drilled in the platform. I isolated them with the same rubber tubing I used with the baffle locating pins from the baffle base (see **Photo 3**). I also isolated the nuts and washers used to fasten the plates using small wooden backers with Sorbothane pieces to separate them from the platform (see **Photo 4**). The entire tweeter/mounting plate assembly was effectively isolated at every contact point. This completed the project's construction phase. The finished speaker with and without grilles is shown in **Photo 1** and **Photo 5**, respectively.

Crossover Issues

When designing a speaker, as with almost everything else in life, the key to success is making informed compromises. If audio is your hobby, your efforts should make you happy, or why bother? Don't let anyone else tell you the "correct" way to do things. As former-President Ronald Regan once said, "Trust but verify." When it comes to audio, you need to know what you expect from your music reproduction.

That being said, before voicing or designing a speaker, you should define the reproduced sound's most important characteristics. Then try to make the fewest number of compromises in the areas most important to you. For me, I define them in order of preference—flat frequency response, high dynamic range, broad frequency response, controlled radiation pattern, good transient response, and reasonably low distortion.

These criteria should not be viewed as standalone parameters as they definitely can and do interact with each other. For example, if your speaker does not have enough linear volume displacement when you try to achieve the output levels that produce high dynamic range, you will receive high distortion levels.

Regardless of your preferences, in a multi-driver speaker you should first decide what kind of crossover you want to use—passive or active. Next, you should define the crossover points and slopes because they will affect other considerations. Then, determine each driver's drive level. You should also address any driver performance anomalies.

In my case, the first decision was a slam dunk. I have not used passive crossovers for years. Designing a comprehensive crossover is more difficult using passive components and inevitably results in too many compromises. Also, when working with passive components, any errors that creep into your initial assumptions require a lot of time and

Photo 2: To secure the AMT tweeters to the baffle assemblies, I made brackets from a steel plate with a 0.25″ × 20″ steel-threaded rod brazed into holes.

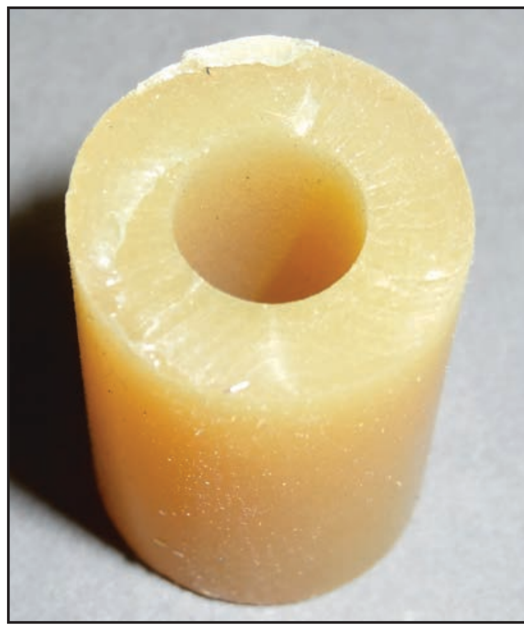

Photo 3: Surgical rubber tubing cut into appropriate length pieces is an effective method to isolate vibration between the baffle's support shelf and the mounting bolts.

Photo 4: Mount the tweeter on the baffle shelf with layers of Sorbothane between the tweeter and rubber tubing around the bolts. Also use Sorbothane under the bolt mounting plates.

Photo 5: The completed speaker is shown without the grille.

money to change parameters. Many options for electronic crossovers provide several choices of crossover types, frequencies, and slopes at a low cost. Some of them enable you to adjust for driver offsets through time delays, set up base parametric EQs, and implement frequency sensitive dynamic EQs that can help mitigate sensitivity-driven amplifier distortions.

Some even enable you to link a PC and make changes, saving virtually unlimited numbers of combinations, while listening to the music. Just the ability to hear the effects as you make changes is reason enough to use electronic crossovers. If you have not designed a speaker with electronic crossovers, you owe it to yourself to give it a try. I predict you will never go back to passive crossovers.

I used a Behringer ULTRA-DRIVE PRO DCX2496 digital loudspeaker management system for the crossovers. I have used this device for years in my main system. It provides a degree of flexibility and control that is hard to believe. There are an incredible number of parameter choices when you design and use a speaker. In my main system, I used Linkwitz-Riley configurations with 48-dB/octave slopes. I used this configuration as the starting point for this project. My original choices for crossover points were 151 Hz and 1 kHz. Later testing required changes, but that took only a few seconds with this crossover.

Sensitivity Issues

Many speaker designers do not address the effects of sensitivity on dynamic range because they cannot determine what amplification will be used with a particular design. They are also faced with Hoffman's Iron Law. Named after J. A. Hoffman, one of the founders of KLH, the law states that small bass enclosure size, low reproduced frequency, and high sensitivities form three branches of a triumvirate. With any design, you can choose any two branches, but you cannot achieve all three. Small enclosure size has become an important consideration in modern designs as well as the demand for low-frequency reproduction. The inevitable result is a reduction in sensitivity, which is common with most speakers today.

Because my design is a tri-amped speaker, I chose three amplifiers I already had available. I used a Crown Audio Studio Reference amplifier rated at 1,160 W per channel into 4 Ω for the bass, a Hafler DH500 power amplifier rated at 255 W per channel into 8 Ω for the midrange, and a Parasound HCA 800-II amplifier rated at 150 W per channel into 4 Ω for the high frequencies.

Considering the three drivers' sensitivities, the amplifier choices were clear. The Dayton Audio UM12-22 Ultimax DVC subwoofer sensitivity was 84 dB with 4-Ω impedance. The Eton midrange driver sensitivity was 89 dB with 8-Ω impedance. The AMT tweeter driver sensitivity was 92 dB with 4-Ω impedance.

Beware of another misleading trend that is occurring: Sensitivity is given as the sound pressure level (SPL) at a 1-m distance created by providing the driver with 1 W of power. For a driver with 8-Ω impedance, that represents a 2.83-VRMS voltage. Many specifications are now given with a 2.83-V drive level regardless of the driver impedance. If you do the math, that voltage level represents 2 W with a 4-Ω impedance and 8 W with a 2-Ω impedance. That makes low-impedance drivers look much more sensitive than their higher-impedance counterparts. However, if they are driven with the same voltage, they are actually consuming more power. The reasoning for this logic is the ability of some voltage-limited amplifiers to provide more power into lower impedance loads. However, they still require more electrical power to achieve their stated acoustical output.

If you take into account the drivers' sensitivity, **Table 1** shows the driver/amplifier combinations are good matches as far as maximum SPLs are concerned. If all the peak levels at every frequency in the music are the same, the amplifiers shouldn't limit dynamic range. As it turns out, that is a bad assumption because of the high levels in the bass range with some well-recorded material. On drum recordings, I actually clipped the Studio Reference amplifier. It was interesting to note that this powerful amplifier clipped before the Ultimax driver complained, which is a true testimonial to this driver's exceptional performance. I will discuss a rather elegant solution to this problem later in the article.

Another problem occurred when I set the amplifier gains. The DH500 amplifier had no level controls. Although the DCX2496 crossover has gain and attenuation available in the digital domain on each channel, I added analog level controls to the DH500. I also placed two independent potentiometers on

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Acoustic Power vs. Drive Power					
Dayton UM12-22 Woofer		Eton 7″ Hexacone Midrange		Heil Air Motion Transformer (AMT)	
Acoustic Output (dB)	Power Watts (W)	Acoustic Output (dB)	Power Watts (W)	Acoustic Output (dB)	Power Watts (W)
84 dB	1 W	89 dB	1 W	92 dB	1 W
87 dB	2 W	92 dB	2 W	95 dB	2 W
90 dB	4 W	95 dB	4 W	98 dB	4 W
93 dB	8 W	98 dB	8 W	101 dB	8 W
96 dB	16 W	101 dB	16 W	104 dB	16 W
99 dB	32 W	104 dB	32 W	107 dB	32 W
102 dB	64 W	107 dB	64 W	110 dB	64 W
105 dB	128 W	110 dB	128 W	113 dB	128 W
108 dB	256 W	113 dB	256 W		
111 dB	512 W				
114 dB	1,024 W				

Table 1: Each driver generates a sound pressure level (SPL) for a given input drive power.

the rear of the chassis. I connected them to the input jacks and added a dial plate with reference numbers for repeatability in setting the levels. I then used pink noise and a sound level meter as a starting point to set each amplifier's gain levels. I adjusted the levels during testing.

Taming the Clipping Problem

Previously, I mentioned that on certain well-recorded cuts I was actually clipping my powerful Studio Reference amplifier during drum strikes. I have no idea how much additional power the UM12-22 Ultimax DVC subwoofer would have taken. But I am almost certain most audiophiles do not have more than 1,200 W per channel to drive their speakers, which makes that issue a moot point. In addition, the clipping was less of a problem in this tri-amped system because only the woofer received the distorted signal. The more critical midrange and high-frequency drivers were unaffected. If you use a single amplifier to drive the speaker full range, the clipping problem would be greatly compounded.

Until recently, other than getting a more powerful amplifier, there were several ways to prevent clipping. You could listen at lower levels, which compromised the music's dynamic range. You could use a compressor, which would provide better audibility of low-level signals but squish the music, including the nonoffending parts and makes it relatively lifeless. Or, you could use an equalizer to reduce the offending frequencies, which leads to poor frequency balance that affects the music at all levels.

Wouldn't it be great if you could identify the most common offenders that produced clipping and reduce the drive level only for those frequencies and only when they were high enough in amplitude to cause a problem? The good news is that such a system exists—the Behringer DEQ2496 high-precision digital 24-bit/96-kHz EQ/RTA mastering processor, which is what I use.

The DEQ2496's functions enable you to determine several key signal modifiers and set specific levels where action is taken to modify the gain applied to the signal. The control parameters include the adjusted signal's modified gain level, the threshold where the modification begins, the ratio or rate that the change occurs, the time delay after the trigger level occurs for action to be taken (Attack), the time delay after the signal falls back below the threshold for the modification to cease (Release), and the type of action, including 6- and 12-dB/octave high- and low-pass filters or a bandpass filter with adjustable Qs. All filters have adjustable frequencies.

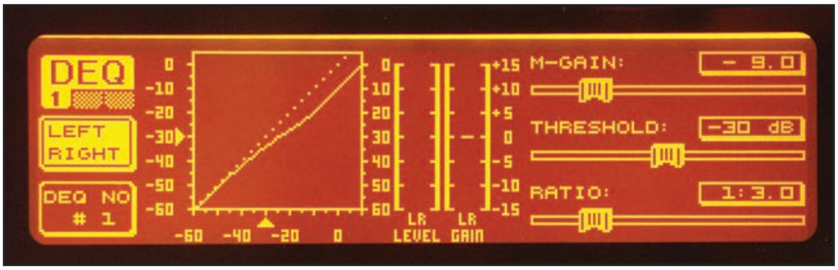

Photo 6: The DEQ function's first page on limiting amplifier clipping shows the parameters numerically and graphically.

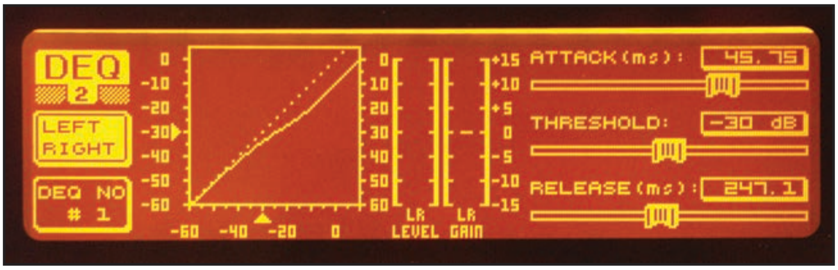

Photo 7: The DEQ function's second page shows the attack and release times.

At first it is hard to realize this function's power. Essentially, you can now tailor the drive level of the music you play to conform to restrictions caused by your combination of driver sensitivity and available power. You can also tailor the drive level at different signal levels to prevent distortion caused by insufficient linear volume displacement in woofers, which affects virtually every speaker. This is a sophisticated control and requires some understanding to utilize it to your advantage. In the DCX2496 crossover, there is a single iteration of this function. In the DEQ2496 equalizer, three individual iterations are possible at the same time.

In my case, the problem with the high-level drum notes was centered at frequencies around 100 Hz. Since they were very transient in nature, it was not a problem of average power. The problem was instantaneous demands that drove the amp into clipping. The goal was to reduce the drive for a range of frequencies centered on 100 Hz with a bandwidth wide enough to reduce the problem yet have minimum impact on other frequencies. I also wanted to make those changes with a ramp up and down timeframe to minimize the chance of them being audible. Since this function works to reduce gain above a certain drive level, the goal was also to shape that gain reduction in a way that it was not abrupt.

To set the parameters, I used an extremely well-recorded drum set from a sampler provided by Legacy Audio. It has a wide dynamic range from light taps on the cymbals to monster drum whacks. With the volume level set to hear the low-level sounds, the drum strikes drove the amp into clipping. This is the perfect scenario for DEQ2496 use.

The DEQ2496 function is controlled on three menu pages. The first enables you to set the amount of modification you make to the gain and is indicated in tenths of a decibel. I set the value to –9 dB. The second is the signal level where the correction begins to take place. I set the value to –30 dB. The third is a ratio setting that effectively determines how fast the gain changes from normal to the earlier modified gain set. The lower the ratio, the slower the DEQ2496 will make the change. I selected a 1:3 value, which provided a relatively slow rate. The settings you choose are shown graphically on the page as they are entered (see **Photo 6**).

The second page of the menu enables you to set the attack and release times and change the trigger point. I set the attack or onset time to 45.75 ms. I set the release to 247.1 ms. **Photo 7** shows the times set in that menu page.

The third page enables you to choose the correction mode and parameters. In this case, I chose a bandpass function with a 100-Hz center frequency. I chose a 0.75-octave bandwidth, which was a good compromise between high effectiveness and low audibility. **Photo 8** shows the results.

After I chose the settings, I repeated the test at the same volume level. The DEQ worked as expected, enabling levels high enough to clearly hear all the quiet passages with the proper frequency balance while preventing clipping on the drum strikes. The difference was impressive. If you watch the first page of the DEQ2496 menu while the music is playing, you can see vertical bar graphs of the signal level and gain modification side by side. As the signal level goes up, you can watch the gain being reduced (see **Photo 9**). You should notice the signal level approaching –25 dB and the gain decreasing to around –5 dB.

In the final part of this article series, I will continue to problem solve issues related not only to this speaker but most speakers on the market. I will also address speaker voicing and compensating curves for various recording shortcomings.

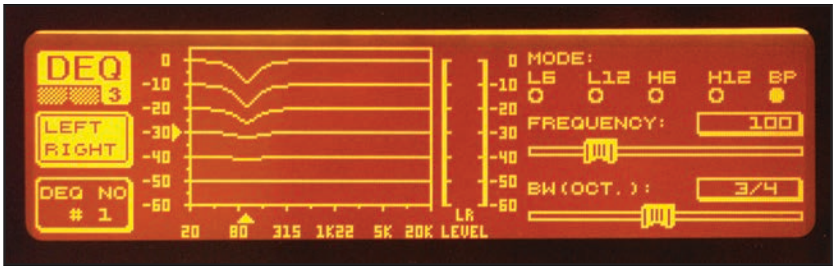

Photo 8: The DEQ function's third page shows the mode employed, the frequency used, and the bandwidth numerically and graphically.

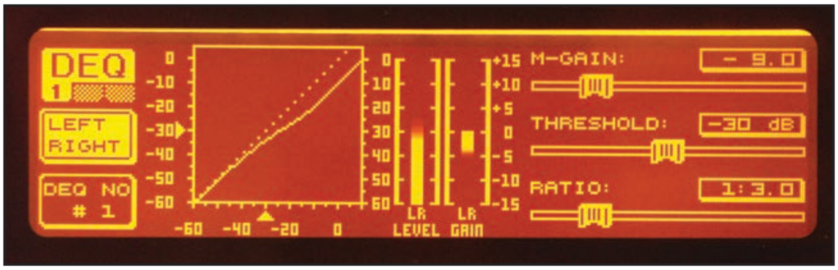

Photo 9: The DEQ menu screen shows the DEQ's action as signal levels change with the music.

About the Author

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

Resources

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Legacy Audio, www.legacyaudio.com.

Legacy Audio, Music Sampler Volume 1, Track 12, "Dynamic Drums."

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Studio Reference amplifier
Crown Audio, Inc. | www.crownaudio.com

UM12-22 Ultimax DVC subwoofer
Dayton Audio | www.daytonaudio.com

DH500 Power amplifier
Hafler | www.hafler.com

HCA 800-II Amplifier
Parasound Products, Inc. | www.parasound.com

Build a Sound Level Meter and Spectrum Analyzer

By
Ron Tipton
(United States)

I designed my sound level meter and spectrum analyzer, the Model 527, around a Velleman Audio Analyzer kit (K8098). It is a bare-bones kit with no enclosure or power supply, but for the price, it is very versatile. I found it easy to build and thought it would be useful in a project. It comes with a 15-page assembly manual and a user guide, which can be downloaded from the Velleman website.

Photo 1: The completed Sound Level Meter includes a shaped aluminum microphone holder.

First, I will review the specifications so you will know what the audio analyzer kit can do. It has a front-panel push-button Mode switch with six selections: Peak power, RMS power, mean (or average) voltage in decibels (dB), peak voltage in dB, linear audio spectrum (20 Hz to 20 kHz) and one-third-octave log audio spectrum (band centers from 31 Hz to 16 kHz). Its sensitivity is –34 dBu which is equal to 15.5 mV RMS. More importantly, it has a 40 dB dynamic range, which means the voltmeter modes have a 15.5-mV-to-1.55-V (–34 to 6 dBu) RMS and peak range. (The voltage and spectrum analyzer modes have the same dynamic range.) The LCD shows the input as a bar graph and as numeric values. It also has a Setup Menu (a "long" press on the Mode switch) for language selection, auto, or manual range selection, display contrast, and so forth. The only specification that I may dispute is the "easy panel mounting,"but I will discuss that in more detail later.

A SOUND LEVEL METER

To convert the kit into a useful test instrument, I designed and built a sound level meter (see **Photo 1**). I opted to call my design the Model 527, because I wanted to keep track of all the drawings and photos. (It may or may not be a TDL Technology product.) The cast aluminum enclosure measures 6.73″ × 4.76″ × 2.17.″ An addendum with full-size drawings, more photos, a parts list, and wire lengths is available at www.tdl-tech.com/build527.htm. A 24-VDC wall power supply—or an external battery pack for operation away from power mains—powers the sound level meter. The Velleman kit uses enough power (75 mA) that I think internal batteries are not practical.

It has a built-in electret condenser microphone (ECM) and an external input to a female Bayonet Neill–Concelman (BNC) connector as selected by a left-side mounted toggle switch. The external input impedance is 47 kΩ and it is DC coupled.

The sound pressure level (SPL) front-panel rotary switch is numbered 50 to 130 dB in 10 dB steps. This is the measured SPL when the graphic display MODE is set to "Peak dBu-Meter" and the reading is 0 dBu. But at each switch position the display has a 40-dB range (–34 to 6 dBu) so the overall SPL range is 16 dB (50 – 34 on the lowest range) to 136 dB (130 + 6 on the highest range). The useable range is about 30 to 136 dB because of the inherent noise in the amplifier circuit. Although the display label is "Peak," the reading is the mean or average SPL. The spectrum analyzer amplitude value is the same as the displayed SPL.

There are three frequency weighting responses as selected by a front-panel rotary switch: FLAT, "C," and ITU-R ARM. The flat response is ±1 dB from 20 Hz to 20 kHz and is primarily set by the microphone cartridge rather than the amplifier. The "C" response attenuates the low and high frequencies (see **Figure 1**). Its use is still favored for subjective sound measurements especially at high SPLs. The old "A" and "B" responses were found to be generally unsuitable in practice and the ITU-R ARM is suggested as a replacement for both. **Figure 1** also shows its frequency response.

The approximate voltage gain from the external input is shown on the SPL range switch. It increases 10 dB for each switch step position as the SPL range decreases. This gain value is approximate because the microphone sensitivity varies slightly from unit to unit and depends on the Calibration control's setting: a 20-turn variable resistor accessible through a small hole in the enclosure's right-hand side.

The signal to the Velleman analyzer also goes to

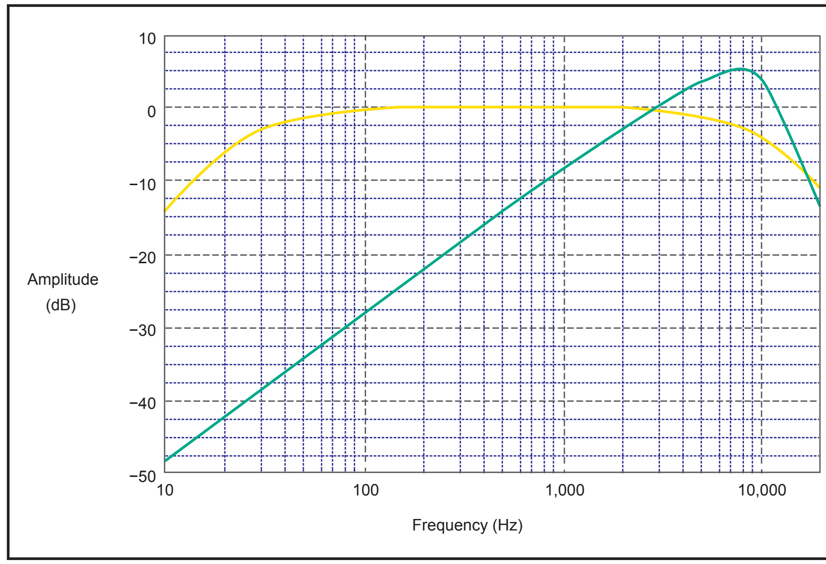


Figure 1: The Model 527's weighting responses are shown. The upper (yellow) curve is the "C" response. The lower (green) curve is the ITU-R ARM.

a female BNC connector on the enclosure's right-hand side. It could be useful for driving an oscilloscope or another instrument (e.g., a computer sound card). Tripod mounting is provided by a 0.25"-20 threaded insert near the center of the enclosure's bottom side.

THE CIRCUIT

The input amplifiers, microphone bias voltage, frequency weighting networks, and power regulators are

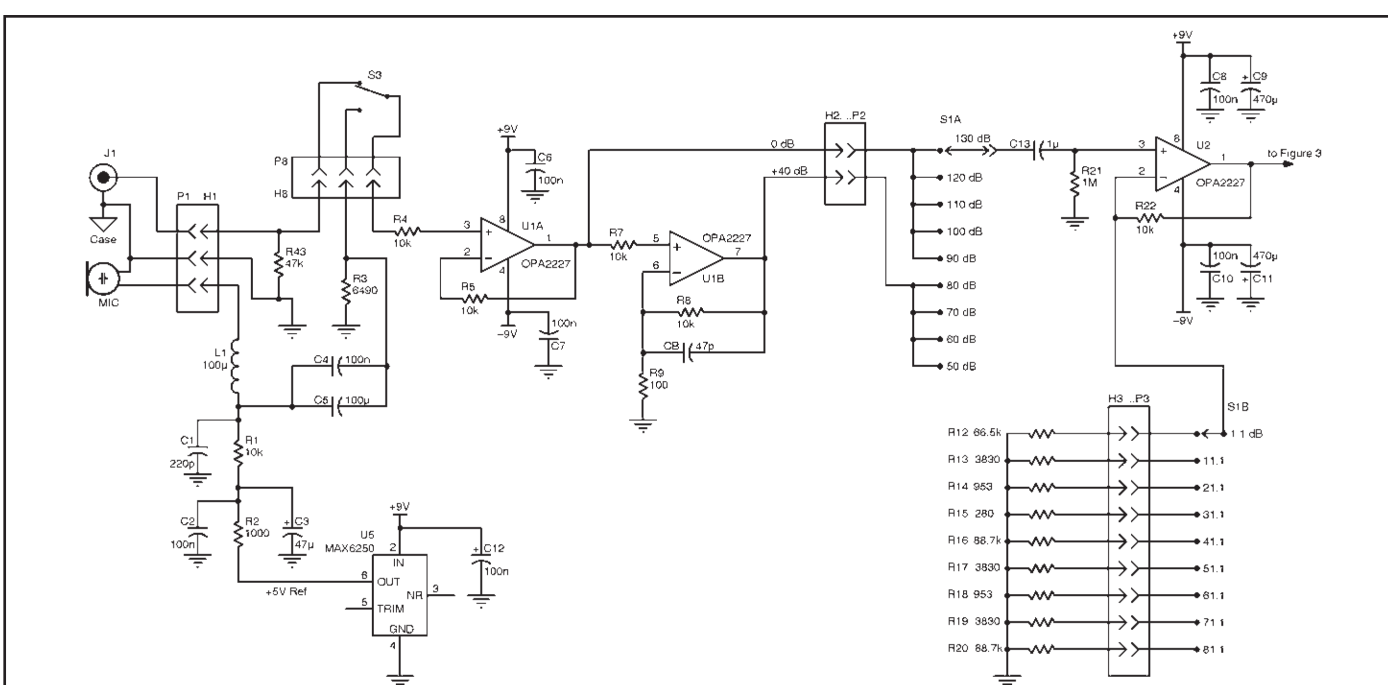


Figure 2: The Model 527's input amplifiers and the microphone bias circuit are shown. The OPA-series op-amps have very low noise.

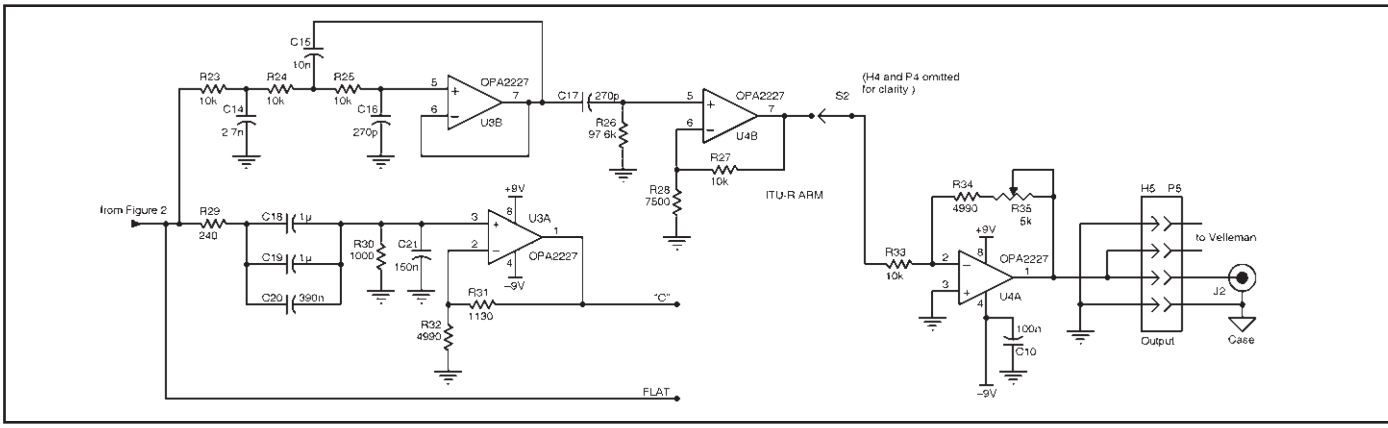


Figure 3: The Model 527's schematic shows the weighting networks and the calibration control, R35.

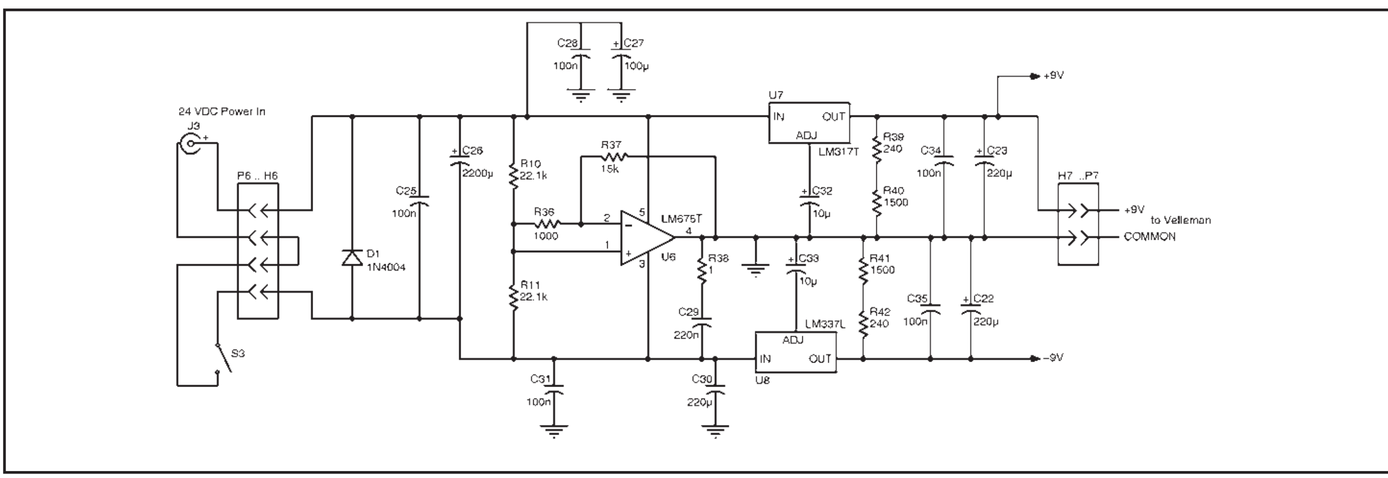


Figure 4: This portion of the Model 527's circuit diagram shows the power supply voltage regulators.

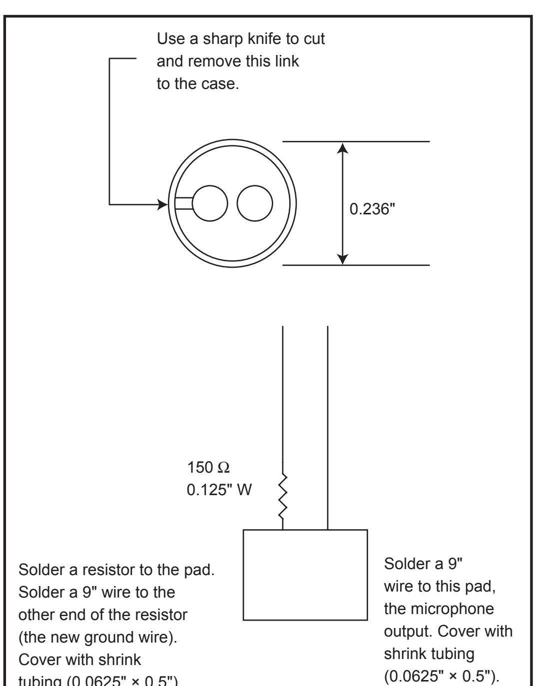


Figure 5: Modify the Panasonic WM-61A microphone cartridge for better linearity at higher sound pressure levels (SPLs).

on a single circuit board measuring 4.5″ × 2.7.″ This is the largest board that will fit in the enclosure's lower half so the electronics are rather compact.

Figures 2–4 show the circuit diagrams. These figures and the parts list only apply to the added amplifier and filter circuits on the Model 527's main board. The Velleman analyzer's circuit diagram and parts list are included in its User Guide.

The microphone cartridge is a modified Panasonic type WM-61A for better linearity at higher SPLs (see **Figure 5**). Working on this small cartridge is easier if you drill an 0.125″ deep, 0.242″ diameter (a "C" drill) hole in a hardwood block to serve as a holder. Also, roughen the cartridge's outside surface with a small file because it must be glued into the microphone holder with conductive epoxy or Wire Glue.

A good electrical contact with the holder is essential for mechanical stability and low-noise operation. The Velleman User Guide calls for operation from a 12 VDC power supply. It is regulated down to 3 to 3.6 VDC on the circuit board, so my ±9 VDC power supply is fine.

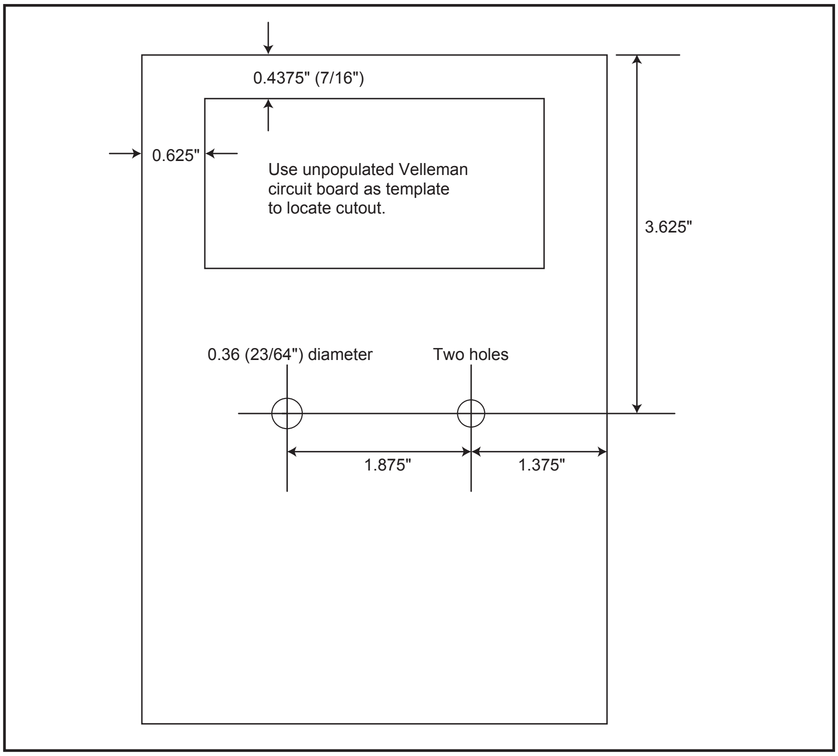


Figure 6: Locate the Model 527's front cover analyzer kit cutout and the holes.

CONSTRUCTION

Mounting the completed Velleman kit is not as easy as its assembly manual claims. The cutout shape is a rectangle with the corners cut off. Use the circuit board as a template before starting the assembly (see **Figure 6**). Most of the cutout can be removed with an 1.5" diameter hole punch or hole saw. Punch or saw two overlapping holes in the cutout area and then remove the rest of the material with a file. The enclosure is made of a cast-aluminum alloy and it files rather easily. I made this cutout in three top plates in a little more than an hour total. One caution if you use a hole saw to get started: Stop sawing frequently and let the saw cool or it will melt enough of the alloy to clog the saw teeth.

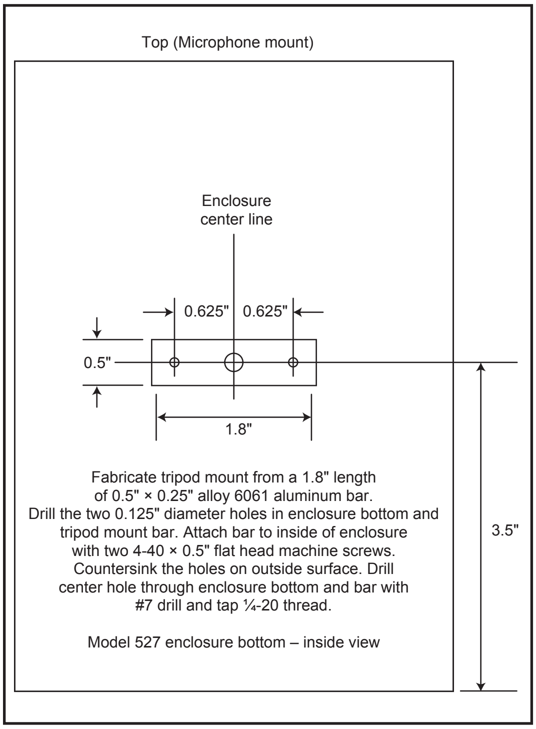


Figure 7: The tripod mount detail is located on the inside of the enclosure's bottom. It is better to use an unpopulated circuit board as a template to locate its mounting holes in the enclosure's lower half.

Figure 7 shows the tripod mount detail but it's better to use the Model 527's circuit board as a template to mark its mounting holes. Place the board, component side up, in the lower half of the enclosure and mark through the holes. (Do not mark through the two 0.14" holes inside the white rectangle. These are used to attach the heatsink bar to the board.) There are three 0.125" diameter holes for the 4-40 × 0.375" machine screws. Put the screws in the holes and secure them finger tight with 4-40 nuts. Do not fully tighten them yet. Put the circuit board in position and then finish tightening the nuts. Before you mount the completed board, place a #4 nylon washer on each of the three screws between the nut and the board. To complete this process, secure the board with three more 4-40 nuts.

Figure 8 and **Figure 9** show the holes to be drilled in the enclosure. In addition to the published drawings, refer to the full-size drawings, circuit diagrams, parts list, construction photos, and connector and switch wire lengths posted online. See the Resources and Sources sections at the end of the article.

I built two versions of the microphone holder: the shaped-aluminum type shown in **Photo 1** and the "tube and bar" design shown in **Figure 9**. Performance is essentially the same and the **Figure 9** design is much easier to build. The cast enclosure has slightly outward sloping walls for good mold release but this means the microphone holder surface is not perpendicular to the enclosure top and bottom.

This is easily fixed with a belt sander. Sand off enough of the enclosure's microphone side to make the top half perpendicular to the enclosure bottom. Sanding just the top half is sufficient, and it prevents you from making that wall too thin.

The new ground wire from the microphone connects to a solder lug under the left-side of the microphone holder mounting screw as shown in **Photo 2**. Another wire continues from this lug to the common connection at H1 on the circuit board.

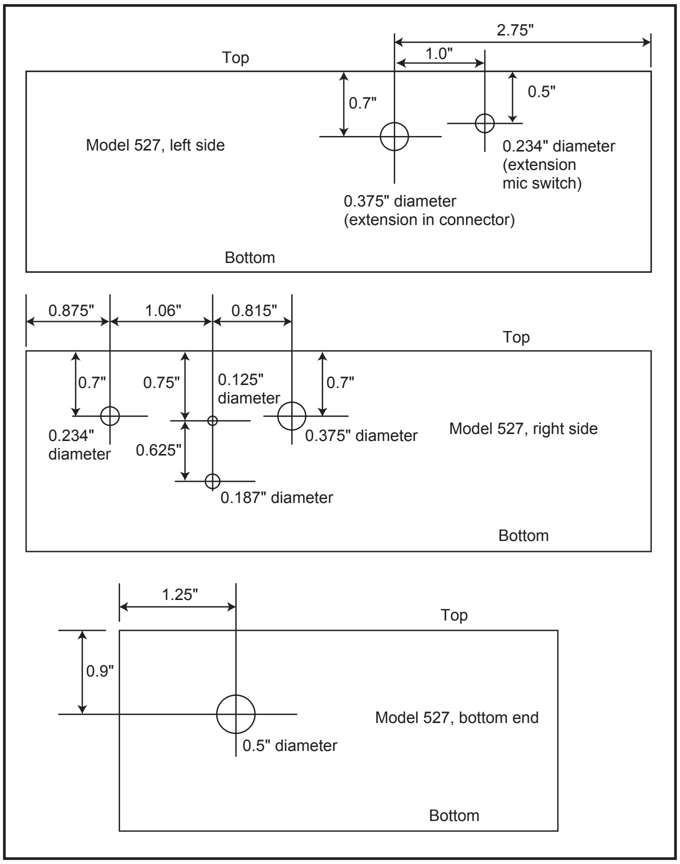


Figure 8: Locate where to drill the holes in the left, right, and bottom (power supply input) sides of the meter enclosure.

CALIBRATION

To calibrate my Model 527, I used a Tenma Sound Level Calibrator (model 72-7260) from MCM Electronics. This is a fairly inexpensive unit that generates a 1,000-Hz tone at an SPL of 94 dB (1 Pa). It is built for a 0.5″ diameter microphone, which is one of the industry standards (along with 0.25″ diameter) and its size dictated my choice of a 0.5″ diameter holder.

Another option is to calibrate by comparison using another sound level meter.

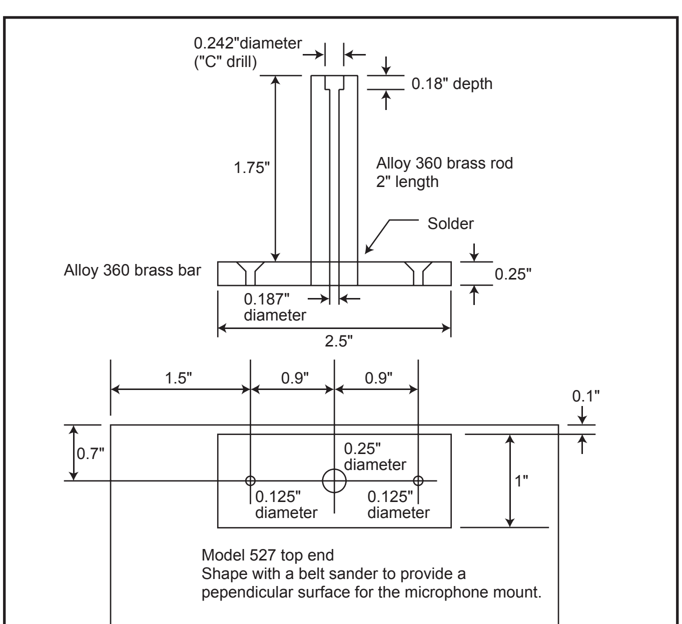


Figure 9: Locate the rod and bar microphone holder on the meter enclosure's top side.

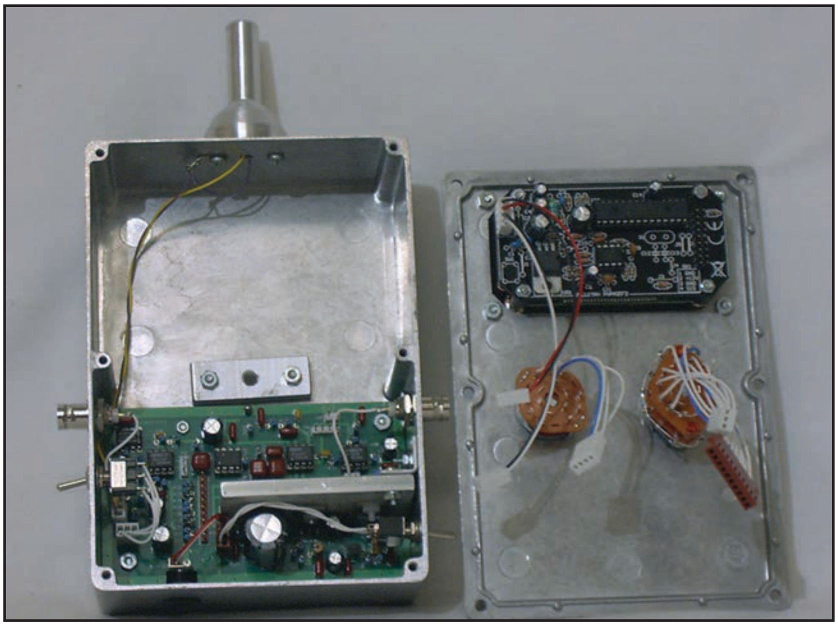


Photo 2: The Model 527 is shown with the top cover removed. All wire lengths are included in the online article addendum (see the Resources section).

Just closely place the two meters side-by-side at least 3′ in front of a loudspeaker and play a 1,000-Hz low-distortion tone at an SPL between 90 and 100 dB. Adjust the Model 527's calibration control to match the reading on the other meter.

If you have a measurement microphone (e.g., the Dayton EMM-6) you will have a calibration graph that shows the sensitivity at 1,000 Hz. It will be around 10-mV RMS at 1 Pa or 94 dB SPL. With a microphone preamp having 40-dB gain, you will have a 1-V RMS output at 94 dB SPL. Using a test oscillator and loudspeaker, you can calibrate the Model 527 by comparison to the microphone's output.

PERFORMANCE

The amplifier noise level referred to the input is 10- to 12-μV RMS depending on the weighting switch setting. This means the meter's useful range is about 30 to 136 dB SPL. To put this range into perspective, a quiet office has an ambient noise level of about 40 dB SPL and a jet engine at 100′ has an approximate 140-dB SPL. An Internet search will provide various SPL tables for different environments.

Overall, the Model 527 seems to perform well for the effort it takes to build it. The parts cost approximately $110, plus the Velleman kit. It may cost less if you already have some of the parts on hand.

Resources

All Electronics, www.allelectronics.com.

Digi-Key Corp., www.digikey.com.

Jameco Electronics, www.jameco.com.

Marlin P Jones Associates, www.mpja.com.

Mouser Electronics, Inc., www.mouser.com.

Parts Express, www.partsexpress.com.

TDL Technology, Inc., Addendum with full-size drawings, more photos, a parts list, and wire lengths, www.tdl-tech.com/build527.htm.

Velleman, "K8098 Audio Analyzer Assembly Manual" 2012, www.vellemanusa.com/downloads/0/illustrated/illustrated_assembly_manual_k8098.pdf.

Sources

Tenma 72-7260 sound level calibrator
MCM Electronics | www.MCMelectronics.com

Microphone holder brass and aluminum
McMaster-Carr | www.mcmaster.com

Dayton EMM-6 measurement microphone
Parts Express | www.parts-express.com

527 Circuit board and microphone holder
TDL Technology, Inc. | www.tdl-tech.com/parts527.htm

K8098 Audio analyzer kit
Velleman, Inc. | www.vellemanusa.com

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.

Member Profile

Member Name: Doug Pomeroy

Location: Brooklyn, NY

Education: Doug has a BA in Psychology from the University of California, Los Angeles (UCLA).

Occupation: Doug is an audio engineer. Among his career highlights, he was a Columbia Records staff recording engineer from 1969-1976.

Member Status: Doug said he has been reading *audioXpress* magazine since its first issue.

Affiliations:
He is very active in the professional audio industry. Doug is a member of the Audio Engineering Society (AES), the Boston Audio Society, and the Association for Recorded Sound Collections (ARSC).

Audio Interests:
Doug said he enjoys audio restoration and audiophile recording.

Most Recent Purchase:
His most recent purchase is WaveLab7, which is used for mastering, audio editing, and restoration. Doug said he tried it and does not like it.

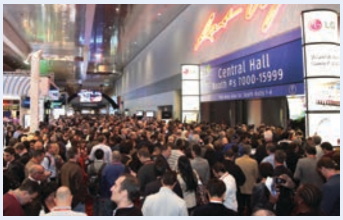
Current Audio Projects:
Doug is working on a project that involves the transfer and restoration of Count Basie "acetates."

William James "Count" Basie (August 21, 1904–April 26, 1984) was an American jazz pianist, organist, bandleader, and composer.

Dream System:
In response to this question, Doug said "I have no more dreams, but I might like some good ribbon mics."

Industry Calendar

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



January 7–10, 2014 **CES 2014**

Las Vegas Convention Center (LVCC), Las Vegas, NV, USA
www.cesweb.org

CES, the largest show in the US, is also the best place to find the latest audio consumer technologies in many different categories including: audio high-performance, portable audio players, and accessories. Unfortunately, this means lots of walking between the LVCC—where all the main large consumer electronics (CE) brands are to be found—and the suites at the Venetian Resort Hotel Casino, where the high-end brands will show their latest solutions in controlled environments.

January 23–26, 2014 **NAMM Show 2014**

Anaheim Convention Center, Anaheim, CA, USA
www.namm.org

The NAMM Show is one of our favorite shows in the world, because you can find an amazing mix of passion for music, audio production, and sound all in a vibrant and happy environment. Look for *audioXpress* in those magazine bins. Our team will be checking the latest software developments for studio production and looking for those smaller booths that show the latest stomp boxes, series 500 modules, and the return of the analog synthesizers.

February 4–6, 2014 **Integrated Systems Europe (ISE) 2014**

Amsterdam RAI, The Netherlands
www.iseurope.org

This is a very successful European show, promoted jointly by InfoComm and Custom Electronic Design and Installation Association (CEDIA). The show has already surpassed all growth expectations and is becoming larger than any of the original North American events. Integrated Systems Europe (ISE) is also becoming an interesting convergence point for new Internet Protocol (IP)-based technology solutions, from home networks and automation to building management. Audio over IP is becoming the hot topic and *audioXpress* will be there, checking for the latest innovations.

February 24–27, 2014 **Prolight+Sound Guangzhou**

China (Guangzhou) International Professional Light and Sound Exhibition
Area A, China Import and Export Fair Complex, Guangzhou, China
www.messefrankfurt.com.hk
www.prolightsound-guangzhou.com

Benefiting from the strategic alliance between Messe Frankfurt and Guangdong International Science and Technology Exhibition Company (STE), Prolight+Sound Guangzhou is now China's largest professional audio, lighting, and stage products show. Exhibition space at the 2014 show will expand from nine to 11 halls (110,000 m^2) and the event is expected to attract an estimated 50,000 professional visitors. Messe Frankfurt Shanghai will debut three new halls, that feature several pro audio brands including AKG, Beyerdynamic, Clair Bros, Crown, dbx, JBL, Lexicon, L-Acoustics, Martin Audio, Mediamatrix, Peavey, Sennheiser, Shure, Soundcraft, and Studer.

Located in the Guangdong province, Enping City has long been China's major import and export base for microphones and entertainment equipment. For many years, the Enping Audio Industry Association has organized a pavilion at Prolight+Sound Guangzhou to strengthen trade between the city and international markets. For the 2014 show, the Enping pavilion will expand from one to one-and-a-half halls to showcase more than 1,000 different types of audio visual systems, conference systems, consoles, and microphones.

March 12–15, 2014 **Musikmesse e Prolight+Sound Frankfurt**

Exhibition Centre, Frankfurt, Germany
www.musikmesse.com
www.prolight-sound.com
www.messefrankfurt.com

This is by far the world's largest professional show dedicated to music instruments, music technology, and professional audio. *audioXpress* will attend in full force covering all the audio novelties in the Prolight+Sound halls and the technology-oriented halls of the Musikmesse, where "the return of the analog synths" is apparent.

This is also the place to look for acoustical solutions, recording studio equipment, and lots of sound reinforcement solutions. In fact, we believe no other show in the world unites as many audio companies from different segments as the annual Frankfurt event. Look for *audioXpress* in the International Press distribution areas.

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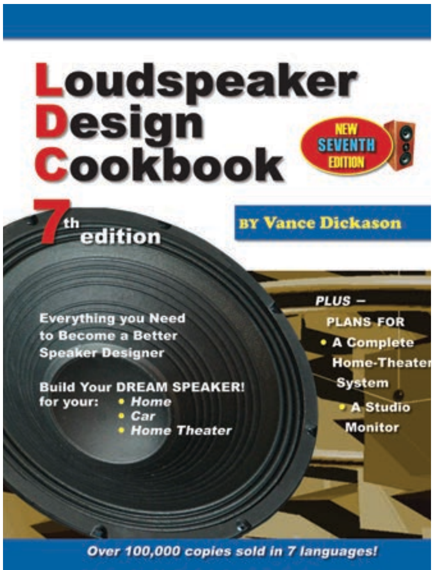

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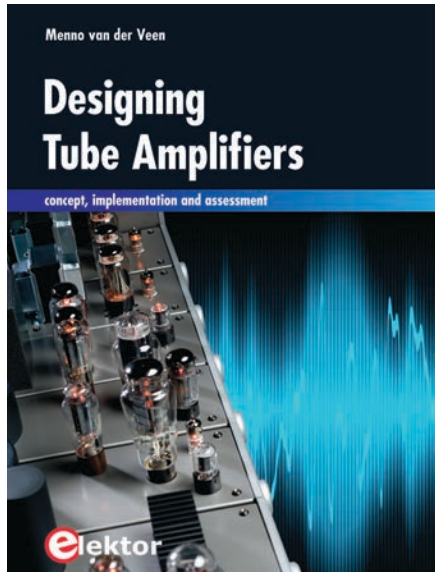

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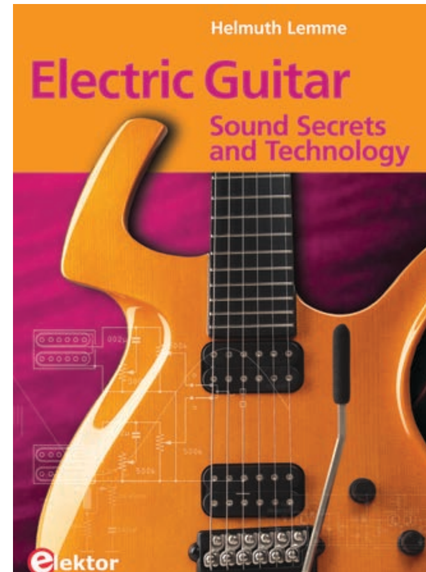

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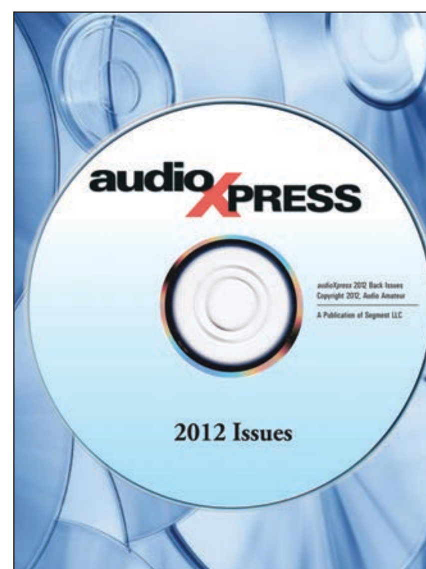

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