www.audioxpress.com

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Audio Electronics
The Correct Use of Rectifier Tubes
By Gerhard Haas

You Can DIY!
Three-Way Loudspeaker System with Active Crossovers
By Ron Tipton

You Can DIY!
The Willow Revisited
By Robert Nance Dee

Speakers
Eighteen Sound
The Younger Face of Italian Speaker Tradition

Questions & Answers
Analoguetube: Recreating the Fairchild Legends

Immersive Audio
Experiments with Dolby Atmos
By Ron Tipton

Standards Review
Audio Network Development: Dante Implementation

Sound Control
Measuring Reverberation Time (RT)
By Richard Honeycutt

$9.00US $10.00CAN
FEBRUARY 2016

February 2016 ISSN 1548-0628

www.audioxpress.com

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

Head Office:
Segment LLC
111 Founders Plaza, Suite 904
East Hartford, CT 06108, US
Phone: 860-289-0800
Fax: 860-461-0450

Subscription Management:
audioxpress
P.O. Box 462256
Escondido, CA 92046, US
Phone: 800-269-6301
E-mail: audioxpress@pcspublink.com
Internet: www.audioxpress.com

Postmaster: send address changes to:
audioxpress
P.O. Box 462256
Escondido, CA 92046, US

US Advertising:
Strategic Media Marketing, Inc.
2 Main Street
Gloucester, MA 01930, US
Phone: 978-281-7708
Fax: 978-281-7706
E-mail: audioxpress@smmarketing.us
Advertising rates and terms available on request.

Editorial Inquiries:
Send editorial correspondence and manuscripts to:
audioxpress, Editorial Department
111 Founders Plaza, Suite 904
East Hartford, CT 06108, US
E-mail: editor@audioxpress.com

Where Is Audio Technology Heading?

Reflecting on many of the developments and announcements in the audio industry from a broader perspective, I thought I should share some thoughts on where things are heading and a few interesting trends for 2016.

In our weekly email newsletter *The Audio Voice*—which we send to all registered members of the *audioXpress* and *Voice Coil* communities—I have been regularly writing about how audio product development is benefiting from the Internet of Things (IoT). As I stated, we should also start discussing the "Internet of Audio Things," considering the extensive evolution in this domain—combining low-power electronics, affordable computing platforms, and connectivity. All are key enablers for an entirely new generation of mobile, portable, or simply connected IoT devices, which might as well be audio devices.

When discussing what's next for the audio industry, I personally believe all those technologies will contribute to a new generation of intelligent devices—call it wearables—combining the best digital processing, personalization, and connectivity for audio streaming and wireless audio. As can already be seen from the strong evolution we are witnessing in the audiology market and with headphone products—improving not only the best music reproduction but also speech intelligibility and managing loudness and dynamics for critical frequencies in noisy environments—future music enjoyment will incorporate environmental sensors (autonomous, connected to smartphones/smartphones, and wearable computers) to provide new quality experiences.

We already have acoustic optimization tools that are able to dramatically enhance the response from loudspeakers in any given environment—from a moving vehicle to any space in the home. But those same tools could also be running on the computing devices we already carry with us. In the near future, we will be able to leverage those "personalization" tools not only to optimize personal devices such as headphones and earphones but also all the devices we stream content to—from TVs to whole-home audio installations. These capabilities are critical to create convincing immersive sound in any environment or, similarly, to provide experiences that effectively approach the creative intent of an original recording and mix.

From another perspective, music streaming is fast replacing downloads and I think there's no going back. It's not just about convenience, as some people think. It's mostly about choice. No matter how large our record library (vinyl, CD, whatever), or how large our audio file library is, nothing can even come close to the 40 million or more titles currently available from every music streaming service. And we should not forget that streaming services also enable us to download files for offline listening, for a limited period when we actually need to be offline. So, the motivation to spend money on downloads is disappearing. My belief is that, as soon as music streaming services start combining higher quality options for streaming audio, including the option to download high-resolution files—that will be the end of discussion.

It's also not just about the cost of media, such as hard drives or even cloud storage, but the time spent managing our libraries and the need to manage the associated metadata with online sources. As many of us have already experienced, no matter how hard we try "managing" a film or music library on a hard drive, we eventually lose control of the process and the entire thing "collapses."

And funny enough, quality might also be the reason why there's a limited future for physical media, even for movies and video content. To achieve acceptable quality levels in forthcoming Ultra-HD Blu-Ray discs, even with UHD-BD 100 GB discs, there is a capacity problem when we add multiple language options, immersive audio formats in high-resolution (for me, the main reason to spend upwards of $30 on a disc), high-frame rate, and higher dynamic range 4K movies. Don't even mention extra content. Funny enough, we now have effective video compression technologies for 4K Ultra-HD (HEVC; Perseus; intoPIX/TICO) but they are being used precisely to power new on-demand ultra-high-definition (UHD) streaming services.

João Martins
Editor-in-Chief

OUR NETWORK

audioxpress | VOICE COIL | circuit cellar | LIS LOUDSPEAKER INDUSTRY SOURCEBOOK

SUPPORTING COMPANIES

NOT A SUPPORTING COMPANY YET?

Contact Peter Wostrel (audioxpress@smmarketing.us, Phone 978-281-7708, Fax 978-281-7706) to reserve your own space for the next edition of our magazine.

COLUMNISTS

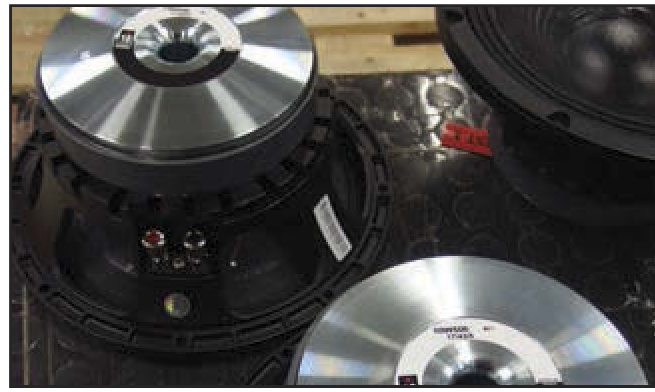
Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—which is now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the 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.

Contents



Features

Volume 47 – No. 2 February 2016

Columns

Departments

Websites

audioxpress.com
voicecoilmagazine.com
cc-webshop.com

@audioxp_editor audioxpresscommunity

Eighteen Sound

The Younger Face of Italian Speaker Tradition

Italian loudspeaker manufacturer Eighteen Sound has ambitious expansion plans with its own product catalog and in reaching out to more pro audio companies on a global level. Recently the company confirmed the acquisition of Ciare, another established Italian speaker brand, broadening the scope of Eighteen Sound's product line to new market segments. *audioXpress* visited Eighteen Sound to find out more about the company.

By
João Martins
(Editor-in-Chief)

Eighteen Sound is a manufacturer of professional loudspeakers, with manufacturing facilities and offices located in Reggio Emilia, Italy, a region with strong industrial traditions. It is also the headquarters for the Landi Renzo Group, to which Eighteen Sound belongs. The company manufactures a full range of high-quality low-frequency transducers and high-frequency drivers in neodymium and ferrite versions, as well as coaxial drivers, line array waveguides, horns and crossovers. As with most OEM speaker manufacturers focused on professional audio and very demanding applications, Eighteen Sound is not a large volume operation in comparison to other companies in the segment, including other rival Italian transducer companies—and neighboring RCF—also based in Reggio Emilia.

The company's production capability reaches 260,000 units per year and 92% of its products are for export. Around 50 people work at Eighteen Sound's headquarters and, as we could see for ourselves, its workforce is young and extremely dedicated.

With the appointment of Pierpaolo Marziali as the new CEO in October 2014, Eighteen Sound started to pursue expansion opportunities and growth strategies, following a five-year business plan established by the Landi Renzo Group board.

Marziali brought to Eighteen Sound extensive experience in corporate Business Development as well as Finance, in his prior role at Landi Renzo S.p.A., the corporate headquarters. Prior to his official appointment, Marziali closely worked for 12 months with Antonia Fiaccadori, the previous CEO and Eighteen Sound's management, on growth strategies and the five-year business plan.

"At Eighteen Sound I found a challenging environment, with very skilled and passionate colleagues. Antonia did a great job preparing all of the elements that a company in this high-end sector needs to innovate, develop, and market professional audio transducers with exceptional performance," he stated. Antonia Fiaccadori remained on the Eighteen Sound Board of Directors as President.

18 Years of Eighteen Sound

Eighteen Sound was founded in 1997 and its first product was an 18" woofer, which inspired the name. The original Eighteen Sound R&D and engineering team, led by Andrea Manzini, came from RCF prior to RCF's acquisition by Mackie. They received the support of Vincenzo Baroni and Ivan Paterlini, the owners and founders of AEB Technologies, an alternative fuel company also from the region of Reggio Emilia (Cavriago), Italy.

AEB wanted to diversify its activities and saw the opportunity to support this group of talented loudspeaker engineers, clearly focused on high-end professional audio applications and already with promising projects in hand.

During his time at RCF, Manzini worked with George Krampera, a true industry legend and a pioneer in active sound reinforcement. Born in Czech Republic, Krampera was responsible for several incredibly successful products and developments at RCF. He left to work in transducer development at B&C Speakers, which is also in Italy.

In the 1990s, he relocated to the Czech Republic to create a new company in partnership with Marcelo Vercelli, also from RCF. Their new company was called Fussion, which was eventually bought by Mackie, which subsequently also bought RCF. When Krampera and Vercelli left there to create KV2 Audio, they trusted Manzini and Eighteen Sound to build the speakers they co-designed and to be their preferred OEM driver partner.

It is important to note that, apart from the mentioned facts, there is no relation between AEB Technologies and Arturo Vicari's RCF Group, which also owns another company called AEB Industriale s.r.l. (dBTechnologies). Because of the name "AEB" (which is common in Italy) and the circumstances surrounding both companies' history, many people assume Eighteen Sound was owned by RCF, which was never the case.

AEB was looking to create a new company very different from RCF. Eighteen Sound designed different products, and the clear focus on pro audio transducers clearly worked.

In 2010, the two original founders—Vincenzo Baroni and Ivan Paterlini—sold AEB Technologies to the Landi Renzo group, a powerful industrial conglomerate of companies with activities spanning from energy to automotive and robotics.

In 2013, Eighteen Sound moved to its new Reggio Emilia world-class setting and production facilities, and announced its organizational independence from the corporate structure of AEB Technologies. In the process, they hired industry veteran Jeffrey Cox, former vice-president of pro audio for Loud Technologies (EAW, Mackie, and Martin Audio) to help with worldwide marketing and business development, with particular emphasis on North America.

As Cox says, "These beautiful Italian transducers are the key to some of the most unique and satisfying professional systems in the world. Eighteen Sound resides in many of the loudspeaker systems known as 'flagship' in our industry. The technologies, friendships, and reputation for excellence that are essential in a strong, ongoing manufacturer-to-manufacturer bond are here in Eighteen Sound."

Pierpaolo Marziali, a graduate of the Alma Graduate School (Bologna, Italy) with a Master's Degree in Finance, was appointed Eighteen Sound's CEO in October 2014.

Growth Strategy

When we met Cox at the InfoComm 2015 show, in Orlando, FL, the company announced the appointment of Rat Sound Systems, Inc., as a Pro Provider in

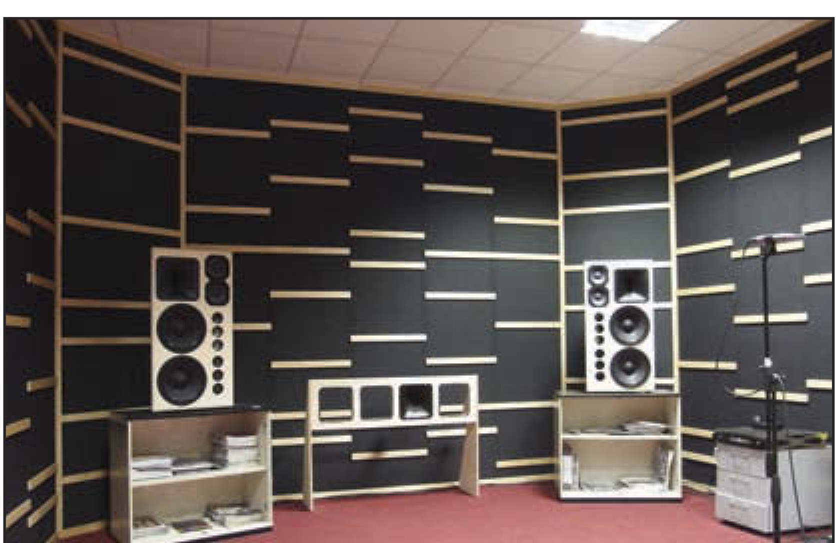
Jeffrey Cox, worldwide marketing and business development, is pictured with Cristina Marconi, marketing operations for Eighteen Sound.

Eighteen Sound usually makes new product announcements at the Prolight+Sound show in Frankfurt (pictured) and InfoComm in the United States.

the United States. Rat Sound is now a supplier of Eighteen Sound products as well as a provider of professional support and service. They also provide re-coning and other services to Eighteen Sound's professional customer base.

But that was just one of the announcements. That same day, Cox and Giacomo Previ (director of global sales), confirmed a new agreement, which allowed Eighteen Sound to control worldwide sales of Ciare's catalog.

Ciare is a well-known speaker manufacturer based in Senigallia, Italy. While Eighteen Sound is a relatively young company compared to other established Italian audio brands, Ciare was founded in 1947. The company manufactures almost all driver parts, including paper pulp cones and silk and titanium domes internally and is respected in the market for its research and development efforts and expanded product line serving both professional and consumer requirements.



This is the listening and evaluation room, when the product is close to completion and everyone gets on the sofa (not pictured) with a bottle of Italian wine and gives a listen.

Unfortunately, few details have been disclosed. Then, we decided we should meet Cox in Reggio Emilia, Italy, so we could visit the factory and better understand the company's plans. Meanwhile, on December 10, 2015, Eighteen Sound confirmed the acquisition of the Ciare brand.

According to Previ, this is "a direct response to our assessment of our product line offerings and will allow us to offer a broader pallet of products. Eighteen Sound will revitalize and strengthen the brand, while maintaining its focus on the professional audio marketplace. The Ciare brand will be maintained autonomously and continue its storied 70-year history, while catering to its loyal client following and maintaining its solid performance standards."

Apart from extending the product portfolio considerably with new home-audio and car-audio market segments as well as an extended range of pro-audio drivers, including midwoofers and high-frequency drivers, Ciare could also considerably extend Eighteen Sound's manufacturing capabilities in components and materials where traditionally the Reggio Emilia company relies on external suppliers.

As Pierpaolo Marziali, stated, "by aligning Eighteen Sound with another loudspeaker manufacturer from Italy, we are expanding our product offerings and sales channels, which will allow us to continue to grow our worldwide presence and capabilities in the OEM marketplace, supplying the finest designers and manufacturers around the world."

"The brand acquisition is strategically aimed at repositioning the Ciare name," he added.

Cox confirms that he was deeply impressed when he visited Ciare's factory. "They have very good products with a different focus than Eighteen Sound and its market is something that we don't speak to. The thing that fascinated me from the very beginning is that Ciare is a two-hour drive from Eighteen Sound and their capabilities and people are really good. And there is a very good product blending with both catalogs."

"Basically they have a steel in, product out, facility. There's not much that enters that building that's not basically metal or raw materials. Everything is built inside, from steel to paper. I've been to Asia and visited many manufacturers and Ciare had equipment in their factory that I had never seen in my life. It's very impressive" says Cox.

Following the announcement, Eighteen Sound also confirmed that new products would leverage

One of the test rooms is used for thermal/power, free-air Thiele-Small parameters, and enclosure measurements. There's a fire extinguisher in case they actually burn units on power test.

the synergy between the two R&D departments. Guess we will know more very soon and further developments are to be expected.

The Factory

When Eighteen Sound moved to its new location in June of 2013, the company suffered a major setback. In July of that same year, a large-scale fire destroyed key areas in the newly acquired production and office facility. The fire caused serious damage and displaced key testing, quality control, and warehousing activities, but the effort to redesign and rebuild was immediate. An "even better" space within the new building was created and equipped with new state-of-the-art equipment, as well as a newly designed anechoic testing chamber.

Simultaneously, an additional 1,000 m² for logistics and warehousing was added allowing for an optimized production flow. All the new facilities were operational in 2015.

When we visited the facility, we could appreciate how Eighteen Sound was able to streamline the flow process throughout the buildings. The new factory includes semi-automated assembly lines, and operations target repeatability of design in every production unit, using proprietary equipment that precisely performs tasks such as applying adhesives in exacting amounts, while skilled assembly technicians handle the essential human interface segments.

Quality control is instituted at every stage of the manufacturing process, whether by automation and software, or close visual and tactile review. In the first stage of manufacturing, all sourced raw materials are individually tested for consistency. Throughout the production process, each stage is equipped with automation and QC workstations. Each product is tested and documented in the company's database, before proceeding to packaging.

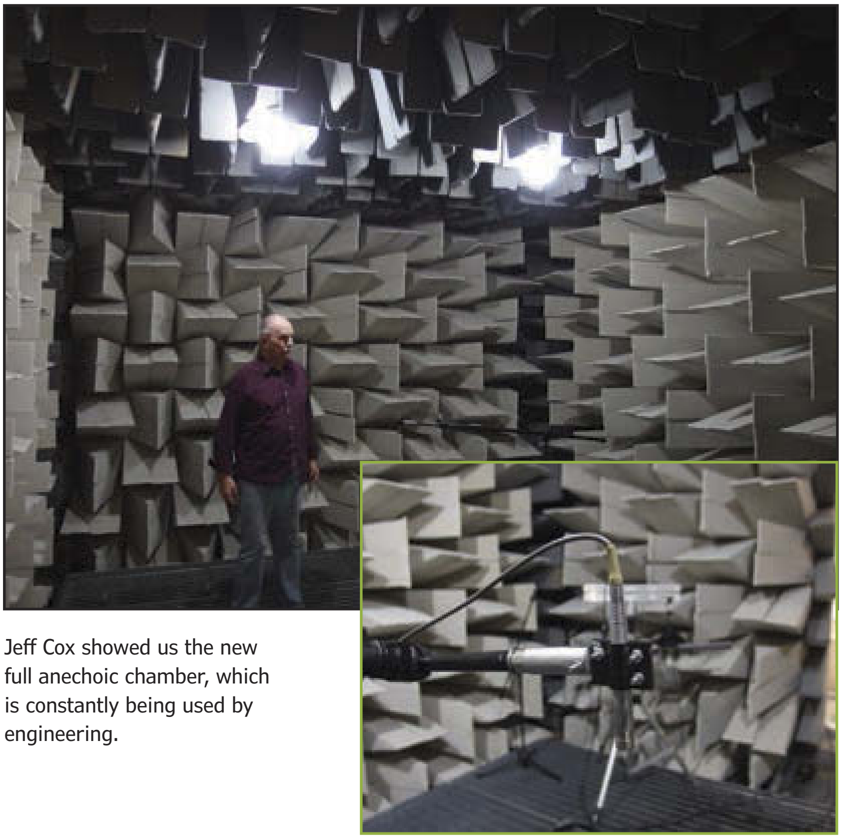
Guided in a full-access visit by Cox and Cristina Marconi, Marketing Operations, we could see how versatile the whole production area can be, with modular production islands allowing simultaneous items, from large sized woofers to compression drivers.

Given his previous experience with Eighteen Sound's products while working with EAW, we asked Cox about his impressions of the company since he joined in 2013. "Kenton (EAW co-founder Kenton Forsythe) was a huge advocate of Eighteen Sound. That led me to go and listen and pay more attention to what was coming out of this company. It was clear that the performance was incredible. For me it was the accuracy of performance, the capabilities of the device itself, and the stability of the product. When I heard an enclosure that was being developed by the company, using different speakers and doing measurements, it was remarkable the capability with Eighteen Sound. Those are the kinds of decisions that designers and engineers like Kenton Forsythe, Jeff Rocha, and Nathan Butler are making. Those are the people that are key to our focus and our growth intention where we want to go. Continue to mark by what the most responsible, the most dynamic, the most adventurous are doing. What do they need?

For acoustic tests, Eighteen Sound developed its own dedicated Advanced Loudspeaker Testing System (ALTS) with a signal generator and programmable test cycles. All test data is stored in a database.

What are they looking for? What is that we have the capability for and maybe they don't even know it…?"

Cox's experience in the North American market also confirmed high acceptance for Eighteen Sound as a brand. "There's a very high perception of the company as a manufacturer. The products are very competitive, incredibly competitive. I've been able to spend some time with designers in companies and I always see the expression in their faces when I talk about pricing and capability. The door of acceptance is there and is just a question of going through the process of evaluation and testing and acceptance, like we would expect of any professional manufacturer."



Jeff Cox showed us the new full anechoic chamber, which is constantly being used by engineering.

R&D and Engineering

Another crucial transition in Eighteen Sound's evolution occurred following Manzini's departure in November 2012. In the interim, the company hired industry-consultant Steve Hutt to reinforce the R&D department and implement new procedures and test protocols as well ensure the development of new high performance transducers, which included new beryllium products.

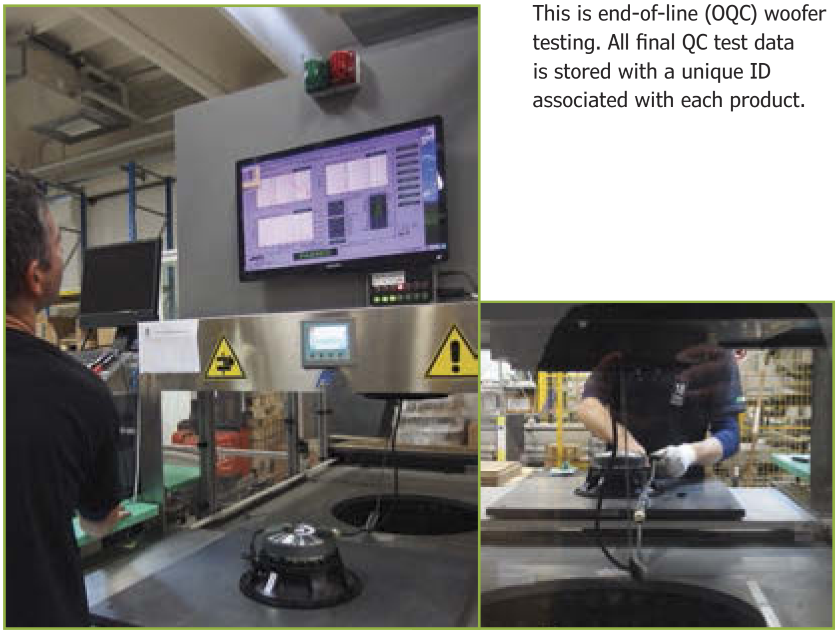
During our visit to Eighteen Sound, we found a solid team of transducer engineers and a restructured R&D operation, fully in control of the company's projects. Eighteen Sound's R&D team is now headed by Marcello Montanari, Project Manager.

We had the opportunity to interview Fabio Blasizzo, the transducer engineer in charge of Quality Assurance and Marco Zanettini, who is also a transducer engineer. As they explained, their R&D team has three people working exclusively on Acoustic Design: Marco Zanettini, Fabio Blasizzo, and Tommaso Nizzoli. Another section—headed by Gianpaolo Lombardi and Francesco Mazzini—handles mechanic engineering, including measuring the samples sent to clients, power tests, product reports, etc. The team includes Filippo Furnari, Supplier Quality Engineer, and Marciliano Campani, who is in charge of samples and prototyping for testing.

All design, evaluation, measurements, and beta versions are handled internally. Zanettini confirmed the company eventually cooperates with other companies for prototyping when they have special needs, but essentially the overall design and prototyping is all done in-house.

Eighteen Sound uses a combination of Klippel tools and Audiomatica's Clio for small signal measurements and they use Audio Precision and other audio measurement tools (sound-card-based) for displacement and acceleration data.

During our visit we found the entire team busy handling multiple projects in the recently built anechoic chamber or in the listening room. Close to the power/thermal test rooms, we also found a rack with a dedicated solution labeled ALTS (Advanced Loudspeaker Testing System). Zanettini explained this is a system developed by Eighteen Sound. "We collect some tone data, the temperature of the voice coil, if there are failures or something abnormal during the test. We store all the data within our database for all the units. The ALTS is a standalone system and only needs an amplifier. It only uses a computer to provide the user interface. It is a really powerful and robust instrument for this kind of test," he explains.



This is end-of-line (OQC) woofer testing. All final QC test data is stored with a unique ID associated with each product.

Assembled 10″ low-frequency ferrite drivers wait for QC. These transducers use a 50 mm diameter copper wire voice coil with Interleaved Sandwich Voice coil (ISV) technology and are recommended for compact bass reflex systems. Note the fine air channels between the chassis back plate and the top plate of the magnet, which draw heated air out from the voice coil gap and dissipate the energy through the chassis casting.

An Eighteen Sound employee supervises training of a new technician.

This is the semi-automatic woofer cone sub-assembly production line.

Otherwise, Eighteen Sound relies on NTi Audio solutions for quality control stages and the production line. "It is a very robust machine and we are happy with it," Zanettini adds.

Technologies and the Future

Blasizzo helped us to categorize Eighteen Sound's catalog and future directions for the company. According to him, Eighteen Sound's hallmark is a combination of technology innovations and the high quality of the final product. Many of the innovations listed in the brand's catalog are not necessarily proprietary to Eighteen Sound but the result of many specific projects and key developments, which the company tries to make available to all manufacturers.

Among those key developments, Blasizzo highlights the Tetracoil Double Voice Coil (TTC) technology as something that is proprietary to Eighteen Sound and evolved there. TTC technology is based on an innovative magnetic structure where two different inside-outside voice coils are wound on the same former and suspended evenly in the two magnetic gaps. Key advantages include ideal motor symmetry over large displacement and excellent thermal dissipation and reduction of thermal distortion. The Double and Triple Silicon Spider technologies were also developed by Eighteen Sound.

Other key design features of Eighteen Sound's products include different ways of extracting heat from the transducer motor to minimize power compression and increase power handling, as well as the exclusive titanium nitride coating process and the use of pure beryllium membranes that dramatically improve stiffness with great benefits in transient and intermodulation distortion response.

According to Blasizzo, Eighteen Sound was able to achieve excellent results with beryllium for demanding applications. "We have a very good beryllium dome (ND4015BE), which is something only a few companies in the world have. The sound of that product is very good." He also highlights the company's neodymium driver with nitrogen-coated diaphragm (NS4015N) and treated titanium drivers as fine examples of their development efforts. "Our customers truly appreciate the nitrogen treatment, for very demanding applications, since the results are similar to beryllium with much lower costs. Maybe other companies have something similar but it's not the same treatment. For instance, B&W in their systems uses diamond treatment on the surface and the quality of the sound is very similar, but we have a different physical and chemical treatment," he notes.

Of course, many of the developments this team has been working on for OEM clients cannot be disclosed and are different from standard catalog items.

Ten years ago, Eighteen Sound started to invest in new product families and recently increased the number of drivers. "In general our products are very robust, the 3" is able to manage much more power, than stated in the catalog," says Blasizzo.

In its catalog, Eighteen Sound also features loudspeakers designed to couple with iPAL Differential Pressure Control technology from Powersoft. The iPAL power amplification module features a zero latency pressure-sensor feedback applying real-time correction that maximizes the control of these specific high efficiency transducers for unparalleled output at low frequencies.

As Cox explains, the 18iD and 21iD woofers were developed in-house. "This is a market that Powersoft was defining and it was something that we felt we could provide with a better motor than was currently available and it was something that we went into on our own. Knowing the IPAL technology, Eighteen Sound developed those two products, both in neodymium."

Trying to work with increased power amplification, increased professional demands, and dealing with power compression and heat dissipation has become very important for the development of products.

As Blasizzo confirms "when we test and make choices for speakers, we consider the maximum peak power, the current that flows in the voice coil and the mechanical stress that we have on the speaker. The RMS power is pretty much the same as it was 10 years ago. It's the peak power that it's much more now. We have better soldering points on the terminals and we need bigger lead wire and the wire section is higher and the voice coil has much more mass than in the past."

"On another perspective we have some speakers that allow managing as much power as we want, which even the most powerful amplifier in the world could take to the maximum level—like the 18TLW3000 woofer and other products that are able to manage very high power," he adds.

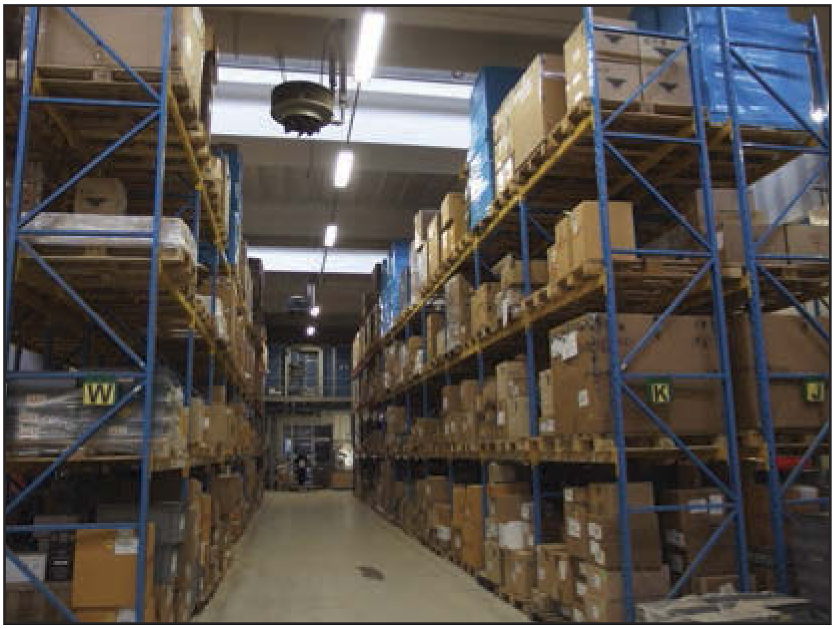
Cox anticipates we should see Eighteen Sound leading the way as a serious player in the professional audio market.

"It's clear that performance balanced by weight is a very important design element. But we also need the capabilities to have additional performance that measures. Development of products is not inexpensive at all. This is a pricey industry to be involved in. The materials to achieve that lightweight product is an expensive endeavor," Cox explains.

"We are also looking at capability and size performance. Can we get the performance of a 10" out of an 18" size? What can we do with an 11" speaker? I would say that we are looking at those kinds of things. Performance is always key. If we have learned something in this decade, is that the technologies are morphing quickly. So why can't we think further out of the box?"

Visit www.eighteensound.com for more information about its products.

During semi-automated production, magnet assemblies are attached to frames, cone assemblies are glued to frames, and spiders are glued to frames.



The new factory building is complemented by another 1,700 m^2 of warehousing, storage, shipping, and receiving, directly adjacent to the production area.

Immersive Sound (Part 2)

Experiments with Dolby Atmos

Introduced in 2012 as a commercial motion picture theater system with the soundtrack for Pixar's *Brave*, Dolby Atmos from Dolby Laboratories, is another object-based immersive audio system. But what exactly is an object? This article answers that questions and explores the functionality of Dolby Atmos.

By
Ron Tipton
(United States)

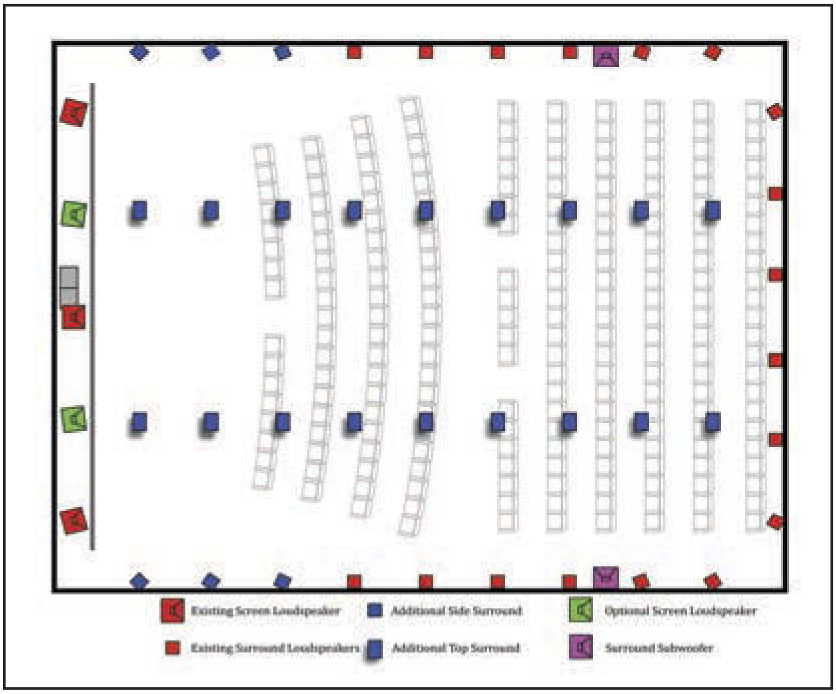


Figure 1: An overhead view of a typical movie theater's loudspeaker arrangement, before and after the addition of Dolby Atmos. (Image courtesy of Dolby)

An object can be any sound data and the accompanying metadata, up to 112 of them. The sound can be a person screaming, a running vehicle, or anything else. The metadata describes the sound, which includes the loudness, where the sound originates (which speaker), and how it moves from speaker to speaker, (if that is applicable) and other things.

The number of objects is not the same as the number of loudspeaker channels. The playback processor uses the metadata to direct sound to individual speaker channels. The processor programming can be changed to accommodate the number of speakers and their locations for different theater arrangements. A motion picture may have many objects, while a symphony recording may have only a few, but the metadata is still used to control how the sound is presented.

Height loudspeaker channels are also an important part of Dolby Atmos, as they are in the Auro-3D system described in my previous article, "Immersive Sound (Part 1): A Look at Auro-3D. **Figure 1** shows a typical theater loudspeaker arrangement.

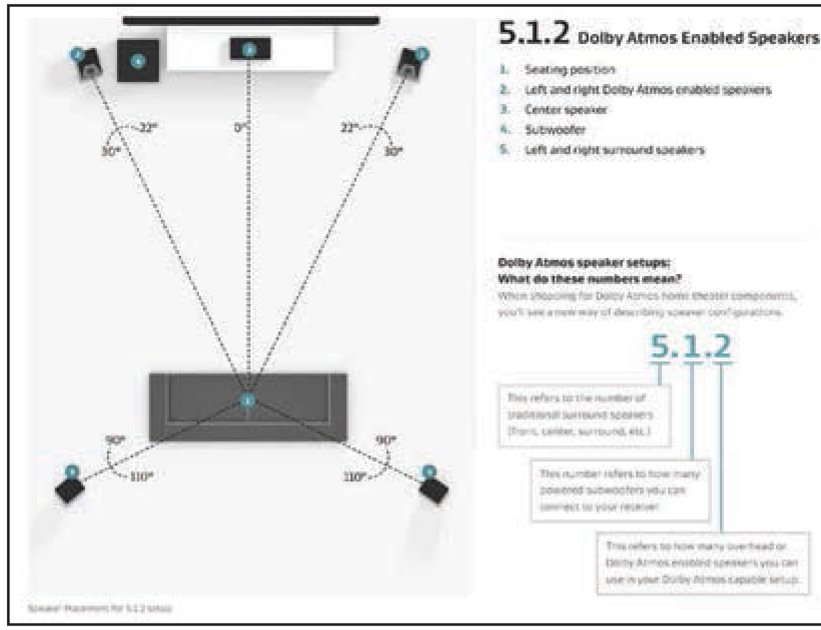


Figure 2: This diagram shows the suggested loudspeaker placement for 5.1.2 Dolby Atmos. This entry-level system can be built by adding two height loudspeakers to a traditional 5.1 surround system. (Image courtesy of Dolby)

Home Atmos

Theater Atmos is interesting and you can find a lot of information on the Internet, but I'll concentrate on Home Atmos. The height channels are still essential, but they are limited to just 10, with four being a more usual maximum.

Dolby Labs recommends a variety of loudspeaker positions from its so-called "entry level" (5.1.2) to its 11.2.4, a fairly advanced system, although not its maximum, which is 22.2.10. The final number in the label is the number of overhead or height loudspeakers. Thus, a traditional 5.1 system becomes 5.1.2 with the addition of two height loudspeakers. **Figure 2** shows this recommended speaker placement. This diagram appears on page 17 of the *Dolby Atmos Home Theater Installation Guide* (see the Supplementary Material).

It's probably apparent by now that Home Atmos is aimed toward reproducing motion picture soundtracks rather than just music. Several hundred movies have already been re-mastered to Blu-ray or DVDs, but not all of them have Atmos soundtracks. According to Big Picture Sound (see Resources) just 29 have been released or are scheduled for release by the end of 2015. I looked at four of my music-oriented DVD movies that have been re-mastered to Blu-ray and I found the following for the English language soundtracks:

- *Abba the Movie* (Polar Music Production B0005474-09). DVD: 2.0 PCM and 5.1 PCM. Blu-ray: 2.0 PCM, 5.1 PCM and Dolby Digital 2.0.
- *Walk the Line* (20th Century Fox). DVD: DTS 5.1. Blu-ray: DTS-HD 5.1.
- *Beattle's A Hard Day's Night* (The Criterion Collection 711). DVD: PCM 5.1. Blu-ray: DTS-HD 5.1. (This boxed set includes the Blu-ray disc.)
- *All That Jazz* (20th Century Fox). DVD: DTS 5.1, Blu-ray: DTS-HD 5.1.

Although none of these have Atmos soundtracks, the sound is very good to excellent and, in my opinion, amazing that the Beatles (1964) and Abba (1977) films have six track recordings.

My point is that as of August 2015, music lovers are out of luck because nothing seems available in Atmos or any of the new surround formats. However, the situation may have changed by the time you read this. The realism of a train roaring around a curve or a jet plane taking off is rather hard to judge. Impressive yes, but realistic, I don't know.

Atmos Playback

I have been using a compact Ambery 7.1 decoder (see **Photo 1**) with an HDMI input and eight analog audio outputs in my surround system, which cost less than $150 on the Internet. It supports Dolby TrueHD, Dolby Digital Plus, and DTS-HD Master Audio as well as plain-old pulse code modulation (PCM) 5.1. Its input is an HDMI cable from the Blu-ray player, and it has a pass-through HDMI connector for output to the TV set. Of course, it requires eight power amplifiers to drive the loudspeakers but this has some advantages, which I'll get to later.

It seems, after careful searching, that no such home decoder yet exists for Atmos. The alternative is to use an Audio Video Receiver (AVR) with a

Photo 1: The rear view of the Ambery HDMI to 7.1 decoder shows the four stereo jacks that supply the eight analog outputs. The input HDMI connector and the jack for the 5 VDC wall power supply are on the other end of the box.

Photo 2: The Model 465 volume controller's front panel has the Master Volume knob and the six individual loudspeaker volume controls. The small hole near the lower left corner is the IR port, which also controls the Master Volume. In and out connectors are on the rear panel.

minimum price of about $500. They have built-in power amplifiers for all the loudspeakers with the sometimes exception of the subwoofer. The built-in power amplifiers are not necessarily an advantage, which I will explain later.

I purchased an Onkyo TX-SR444 7.1 AVR to get an Atmos decoder but it supports many formats including mono, PCM 2.0, Dolby TrueHD, DTS, DTS-HD Master, DSD, and others.

This Onkyo AVR has a detailed series of setup screens to define your loudspeaker system. You can enter the type of system from 2.0 stereo to 7.1 surround, each speaker's size, location, and distance from the listener. It also has the AccuEQ Room Calibration feature and I'm sure other AVRs have something similar. However, I did not find this useful because the AVR has a master volume control but no control over the loudness of each individual loudspeaker. This is the disadvantage problem I mentioned earlier. The loudspeaker setup and automatic room calibration would probably work reasonably well if all DVD and Blu-ray movies had equally loud soundtracks, but they don't.

Atmos includes parameters in the metadata that, if set correctly, normalizes loudness and dynamic range for each movie release. I haven't been able to verify this because I don't yet have enough Blu-ray films with Atmos. However, I want to be able to play any available surround format, so I had a problem.

The center and front tracks are usually similar from one film to another but the surround tracks are not. I have ripped soundtracks from several newly released DVD and Blu-ray movies (not Atmos) and, using my dynamics program (see "Signal Dynamic and Loudness," *audioXpress*, May 2015), found surround tracks with dynamic range differences of more than 20 dB. A rocket ship taking off can literally shake my house even though the dialog is at a normal level.

I prefer a system with individual speaker volume controls so I built a 12-channel volume controller using a pair of TDL model 465 6-channel controllers (see **Photo 2**). I modified one of the 465s to accept

Photo 3: The two Dayton B652-AIR loudspeakers are about 6' apart on the rear wall. They provide the height channel for my 5.1.2 surround system.

About the Author

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at the 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.

Project Files

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

Resources

C. Boyland, "Which Blu-ray Discs Have Dolby Atmos?," Big Picture Sound, December 2014, www.bigpicturebigsound.com/List-of-Blu-ray-Discs-with-Dolby-Atmos.shtml.

R. Tipton,"Immersive Sound (Part 1): A Look at Auro-3D, *audioXpress*, December 2015.

——, "Signal Dynamic and Loudness," *audioXpress*, May 2015.

Sources

7.1 HDMI decoder
Ambery.com |
www.ambery.com/2hddodtsdihd.html

DVDFab Blu-ray Creator
DVDFab | www.dvdfab.cn

Onkyo TX-SR444 7.1 AV Channel Receiver
Onkyo | www.onkyousa.com

Dayton B652-AIR
Parts Express | www.parts-express.com

TDL Model 465
TDL Technology, Inc. |
www.tdl-tech.com/data465.htm

the control voltage from the other unit so the Master Volume knob controls all 12 channels. This is a low-level controller so I also modified my AVR to bypass the power amplifiers and put the low-level analog signals on the AVR rear panel connectors. I'm still using the eight power amplifiers I had in my system with the eight-channel HDMI decoder (see **Photo 1**). Of course, I voided my AVR warranty by making the modifications, but the system works well and I'm pleased with the listening experience.

Sale of the Model 465 was discontinued because of lack of interest. However, I have added a file, buildl465.zip, to the Supplementary Material. This file contains complete information for constructing the controller including the Gerber files for the PC boards and the microprocessor control program. I also have a supply of the stepper motors in case someone is interested in building one.

Height Loudspeakers

The height loudspeakers can be located in the ceiling, on the ceiling, or near the ceiling on either the front or rear wall. If four are used it may be convenient to locate two on both the front and rear

Photo 4: The Dolby Atmos-enabled floor loudspeaker has the sound from the upward-facing loudspeaker reflected from the ceiling to provide the height channel. In this photo, we can see the Klipsch Reference Premiere RP-280FA Dolby Atmos-Enabled Floorstanding Speaker. (Photo courtesy of Klipsch)

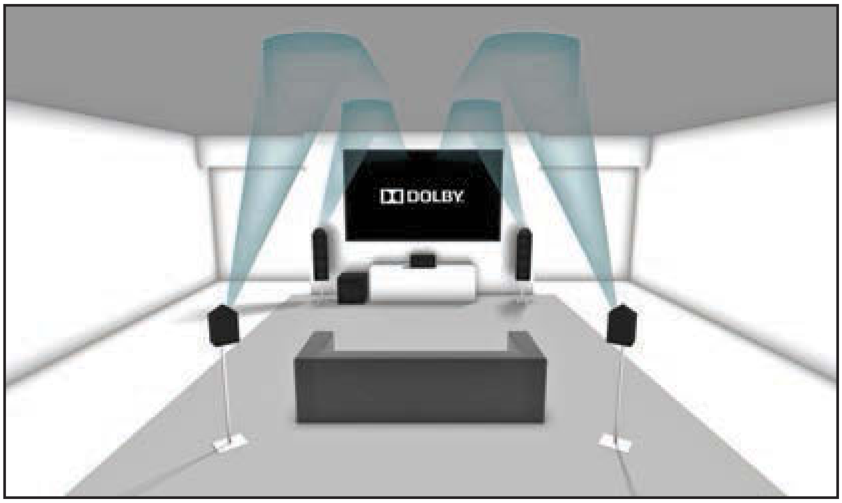


Figure 3: A room diagram shows the use of floor-standing Dolby Atmos-enabled loudspeakers to provide the height channels by ceiling reflection. (Image courtesy of Dolby)

walls. **Photo 3** shows a pair of Dayton B652-AIR full-range on my listening room rear wall near the ceiling. These perform very well but the location you choose depends on several factors. In-ceiling and on-ceiling are best on a flat ceiling so their distance to the listener is the same for both. Ceiling height should be at least 8′ for in-ceiling and at least 9′ for on-ceiling, otherwise the sound becomes directional rather than immersive. As you can see from **Photo 3**, my ceiling is not flat, sloping from 8′ on the left side to 10′ on the right. Wall placement is better in this case and even though the distances are not quite equal, the sound is definitely immersive.

There is yet another way to accomplish the height effect—the Atmos-enabled floor loudspeaker (see **Photo 4**). This is an upward facing full-range, often slightly tilted so the reflected sound falls on the listener as shown in **Figure 3**. According to the Dolby literature, many audio experts think the reflected sound is as good as or better than using "real" height loudspeakers. Certainly, the mounting can be easier.

A frequently asked question is why are there predefined speaker positions in Atmos Home if it is object-based and not a channel-based system?

Dolby Labs answers this in its publication: Frequently-Asked-Questions-About-Dolby-Atmos-for-the-Home.pdf (included in the Supplementary Material) with this excerpt:

"Because Dolby Atmos is new to home theater, we defined a few 'reference' speaker configurations to ensure that early customers could have a great experience while having the option to keep most of the equipment they already have. Among those reference setups are the 5.1.2 configuration, which involves adding two ceiling or Dolby Atmos enabled speakers to a traditional 5.1 system, and the 7.1.4 configuration, which starts with a traditional 7.1 system …"

Atmos Demos

There are free downloadable demos that really "show off" Atmos sound—of course, they were designed to do just that. In particular, four 7.1 files can be downloaded from www.demo-world.eu/2d-demo-trailers-hd. The files of most interest are:

- dolby_amaze_lossless-DWEU.m2ts, 160 MB
- dolby_horizon_lossless-DWEU.m2ts, 334 MB
- dolby_leaf_lossless-DWEU.m2ts, 103 MB
- dolby_silent_lossless-DWEU.m2ts, 355 MB

These can be burned to a Blu-ray disc with a program (e.g., DVDFab Blu-ray Creator) and then played with a Blu-ray player. The catch is, you have to have a Dolby Atmos system to hear the sound. If you don't, the video is fine but you hear a sound track that tells you to set up your AVR for Dolby Atmos. Reasonable, but still somewhat irritating!


18
EIGHTEEN
SOUND
PROFESSIONAL
LOUDSPEAKERS
NOW SHIPPING
HD2000
from our hands
to yours

Audio Network Development (Part 3)

Developing Products Based on Dante

In the first two articles in this series, we touched upon the evolution of audio network connectivity, available technologies and platforms, and provided an update on existing audio networking technologies. In this article, we address how to implement products over Dante—a technology directly developed and supported by Audinate.

By
João Martins
(Editor-in-Chief)

Among the available technologies and platforms for audio networking product development, Audinate's Dante is the most widely available solution. There's a reason why Dante is so popular and that is because Audinate licenses and fully supports all development efforts directly. We also cannot dispute the amazing success Dante has had in the professional audio industry.

A white paper published February 2015 by British firm RH Consulting that examines the current state of the audio networking market states, "Audinate's Dante audio networking solution has had significant growth in licensees over the last 24 months, nearly four times the next largest protocol. (...) Over 700 networked audio products are currently available, with the number of Dante-enabled products introduced in the last 12 month significantly outpacing all other networking protocols. The number of Dante-enabled products is forecasted to grow by 75% in 2015, and 130% by 2016."

"Audio networking is following the same pattern as most new technology," says Roland Hemming principal audio consultant at RH Consulting. "The success of Dante is consistent with easy-to-use, end-to-end solutions driving the market when technology is in the early growth phase." He continues, "Over time networking has become less about specifying a protocol and more about specifying products that work together."

As Audinate states in its published paper, "The Three Pillars of Audio Networking," successfully adopted technologies stand upon three key attributes:

- They deliver complete toolsets—not simply parts—that solve real problems for users
- They are developed and supported by trusted organizations
- They are widely distributed with a goal of fostering ecosystem growth

Audinate based Dante on these foundations, leveraging modern audio networking directly from computer networking—specifically, switched TCP/IP over wired Ethernet and its associated standards.

Focused on Support

Another reason Dante captured the attention of the audio industry in its early stages was because it focused on solving the challenges of delivering tightly-synchronized audio using standard IP networks, achieving ultra-low network latency and simplifying network set-up, creating an intuitive and easy-to-configure user interface.

As Lee Ellison, CEO of Audinate, explains, "Audinate's core expertise is in IP networking that began development more than a decade ago. With digital networking, the physical connecting point is irrelevant: media signals can be made available anywhere and everywhere, eliminating the many bulky cables needed to provide point-to-point wiring for analog AV installations.

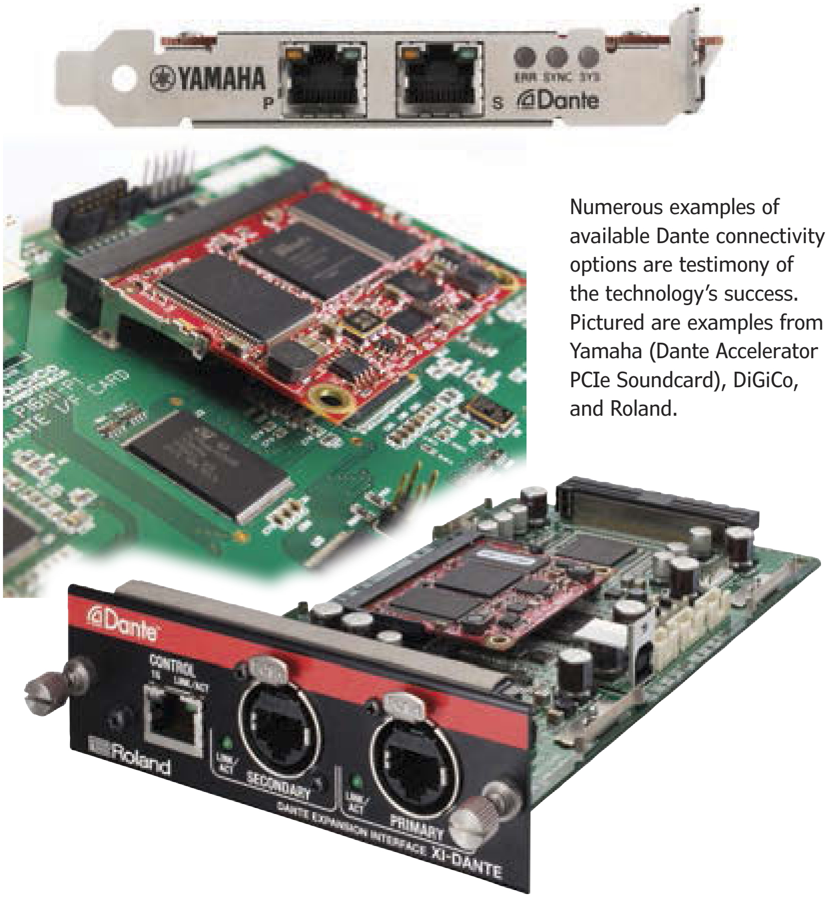
"Audinate simplified network configuration compared to other audio networking approaches that required complex programming of ID numbers and bundles. The Dante networking solution is a true plug-and-play solution, because devices and channels are automatically discovered and easy to program. Patching and routing are logical functions configured in software, not via physical wired links. Audio networking brings enormous benefits enabling audio equipment to become intelligent devices," he adds.

Even though early success stories for the technology originated from live sound applications—in 2008 the successful Dolby Lake Processor became the first Dante-equipped professional audio device—the technology was quickly adopted for installed systems and gradually evolved to professional recording, corporate systems and broadcast.

And Audinate did face some competition. The company adopted a flexible attitude toward existing or emerging networking protocols, promising interoperability when clearly required by the market. Early on, it dismissed competing efforts from the Audio Video Bridging (AVB-IEEE 802.1) proponents, by committing to support an upgrade path when so required. Since the AES67 standard was established and Audinate announced it would also introduce an update to support AES67, it has maintained the commitment to support Institute of Electrical and Electronics Engineers (IEEE 1733) Layer-3 standard, including IEEE 802.1AS for synchronization and real-time transport protocols (RTPs).

Since existing Dante hardware devices can be firmware upgraded as Dante evolves, providing a migration path from existing equipment proved to be an effective way to gain market support, as it was recently demonstrated with the implemented AES67 support, incorporated as an option within Dante. The existing ATP and AES67 (RTP) mechanisms coexist in Dante solutions and the updated firmware "speaks" both.

The way support for AES67 was rolled out, started with the company's highest volume shipping modules—the Dante Brooklyn II card—released for



Numerous examples of available Dante connectivity options are testimony of the technology's success. Pictured are examples from Yamaha (Dante Accelerator PCIe Soundcard), DiGiCo, and Roland.

Sennheiser's Digital 9000 wireless system is one of the examples of Dante networking conquering the pro audio industry.

"With over 500 Dante enabled OEM products available, and hundreds more in the development pipeline, Audinate has become the "de facto" solution for audio networks" — Lee Ellison, Audinate CEO

Focusrite RedNet modular Ethernet-networked audio interfaces harness the power of Audinate's Dante digital audio networking system designed with multiple audio applications in mind, from Live Sound to Multi-room Recording Studios.

integration and testing into products, helping to reassure OEM partners. The upgrade option was offered to OEMs so they could determine if and when AES67 would be incorporated into their products.

As Audinate explained "Dante is a complete media networking solution designed for high-quality AV streaming. The AV industry has embraced Dante because it is easy to set up, delivers a rich and robust feature set and is the most interoperable networking solution available. From the beginning, Audinate has incorporated standards to create the Dante product suite, and AES67 provides another standards-based transport choice within Dante for Layer-3/IP-based audio networks."

Audinate also states, "Neither AES67 nor AVB are competitive equivalents to Dante. Dante is a commercially supported solution, and more than just a standard. (…) Our OEMs recognize the benefit we provide to enable them to develop their products quickly and benefit from our expertise."

No doubt, Audinate will continue to introduce new extensions to its technology, as its adoption to new application fields and commercial requirements progress.

Dante Product Development

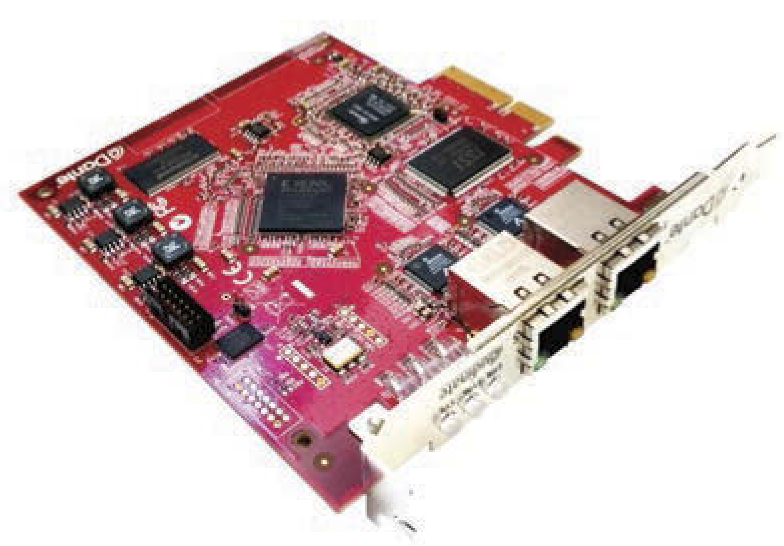
When we visit any audio, AV, or broadcast trade shows, the number of "Dante Spoken Here" signs displayed everywhere serves as testimony to the technology's momentum. Visiting Audinate's Dante online Product Catalog provides the same impression (see Resources).

As Ellison states, "Customer research tell us that the single biggest factor when selecting an audio networking solution is the number of available products on the market. With hundreds of products launched in the last 12 months, Dante has reached the tipping point and has become the industry standard for Audio over IP networking."

"Dante is more than a protocol and is actually a complete networking solution. We recognized early on that there was a huge gap in terms of simplifying the networking technologies, not only with respect to how easy it was to configure and deploy a networked-based audio system, but in terms of providing a completed solution to OEM companies integrating the technology into their products. Audinate provides OEMs with a complete toolkit of networked implementations. This enables OEMs to implement Dante into their products quickly and cost-effectively based on the product requirements. Effective, reliable, and standardized connectivity is a key issue for hardware and software vendors for widespread adoption," Ellison says.

So far, approximately 260 OEMs have licensed Dante to integrate into their products and, as Ellison explains, development is a simple straightforward process. "The Dante PDK (Product Development Kit) contains all you need to quickly become familiar and confident with Dante technology and products. Audinate has a series of PDKs for it various platform implementations. The PDK is a fully functioning system with digital and analog I/Os, word clock, and software components."

Audinate also delivers ready to implement solutions that range from low channel count



For basic testing and high channel count professional audio applications Audinate supplies the Dante PCIe-R soundcard, which also benefits of Thunderbolt external chassis support and is firmware upgradeable.

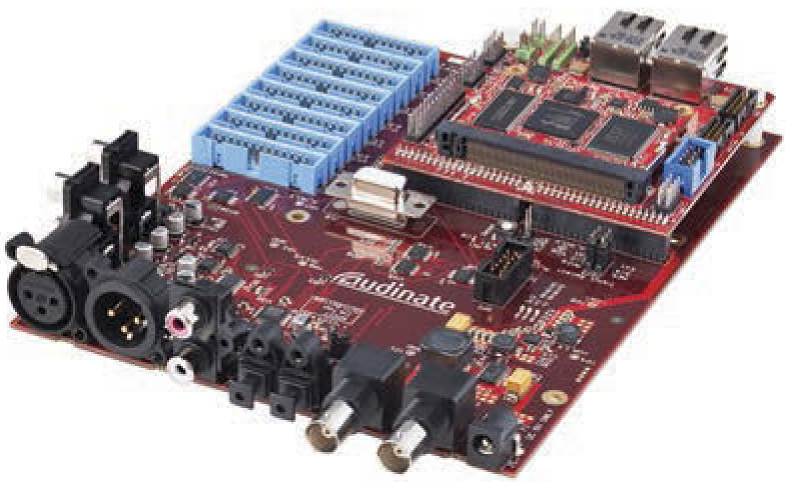
microcontroller chips, to a high channel count FPGA-based solution supporting up to 512 × 512 channels. This way, manufacturers are able to tailor their Dante implementation to the specific needs of their application/market. Dante has been embraced by OEMs across professional live audio, commercial installation, broadcasters, recording and production, transportation and evacuation public address, and music instrument (MI) markets. And even though no Dante products have been introduced for consumer-oriented applications and no specific marketing efforts have been planned, Audinate confirms that might soon be changing. High-end home theater companies are already implementing Dante and the company confirms that its roadmap will also help open up the consumer space.

The Dante Brooklyn II module features a MicroBlaze CPU, which includes its own Linux environment, enabling the development of custom embedded applications for device and network control and monitoring.

Interoperability

As a solution, the Dante API toolkit facilitates rapid development for custom user interfaces and applications to interact with Dante devices. Control data can be shared over the same data network using SPI or UART ports. Access to Dante's powerful control and monitoring capabilities, via the Dante API, enables deep and seamless Dante integration with manufacturer's custom features for the development of highly sophisticated systems and solutions.

And contrary to the long and complicated certification process embraced by the AVB member companies within the AVnu Alliance, projects built using the Audinate-provided platforms and development kits have guaranteed compatibility following simple test procedures.



The Dante Brooklyn II PDK (Product Development Kit) contains all manufacturers need to design, develop, and test fully integrated Dante-enabled products. The PDK is a fully functioning system complete with digital and analog I/Os, word clock, and software components (including the Dante API toolkit), as well as an Audinate technical support package.

As Ellison details, “One of the enormous benefits Audinate brings is guaranteed interoperability. The core Dante software across all of the implementations, from Dante Ultimo (up to 4 × 4 channels), Dante Brooklyn II (Up to 64 × 64 redundant channels) Dante HC (up to 512 × 512 channels), and Dante PCIe (128 × 128 channels) cards are effectively the same, and are as a result automatically interoperable. This benefit and peace-of-mind interoperability should not be underestimated, taking this burden off the manufacturer. Interoperability is not a one-time phenomenon. Audinate is continually making improvements and new features so the end customer does not need to be concerned that changes are going to impact them and customer’s interoperability. In our development and test center, we have racks of equipment from partners for regression testing of new updates and releases. With Dante, manufacturers do not have to spend tens of thousands of dollars to ensure interoperability as they do with other technologies. With other networking protocols, or with manufacturers who try to spin their own networking implementation, they really have no control over the changes made by other third party’s code development.”

“In addition, Dante Controller, a feature reach control management software provided by Audinate, is an enormous benefit to both OEMs and end-users. Having a robust feature rich system configuration and management toolkit is core to the interoperability and industry adoption of Dante. The lack of a sophisticated management controller has been a stumbling block to real world use of other protocols. Audinate’s Dante Controller incorporates label based device and channel names, one click signal routing plus the added support tools like health status monitoring,” Ellison adds.

For manufacturers who do not have the resource bandwidth to develop new networked products, Audinate also works closely with authorized implementers who can provide engineering support. Attero Tech and Auvitran are two authorized Dante implementers providing additional turnkey development to companies who have limited available development resources.

The Future of Dante

According to Ellison, this technology will continue to evolve supporting market requirements and industry initiatives such as AES67. “Audinate introduced AES67 for the Dante Brooklyn II module to connect transport streams to other AES67 compliant implementations. AES67 will allow streams from one manufacturer’s implementation to be set up to another vendor’s implementation. This is a positive step, but it should be stressed that someone has to develop and support the implementation to make a usable networking solution, and Audinate has one of the largest organizations in the world dedicated to develop and support of networking as a solution. Dante is built on standards but over a hundred of engineering development years went into making Dante the full solution suite available today. Audinate presently has 45 people in its global organization, 70% being engineering, with a goal to provide networked solutions that meet the customer and end customer needs. Audinate works closely with OEMs and installation contractors to provide input to its roadmap that is planned for the coming year.”

The Dante Ultimo chip is a cost-effective but feature-rich Dante solution for low channel-count applications. Ultimo delivers up to 4×4 channels at 44.1 and 48 kHz, or 2×2 channels at 88.2 and 96 kHz.

One example of its roadmap was the planned development of software solutions such as Dante Via, an innovative network solution for Macs and PCs, which opens up a considerable number of new applications for the technology. Dante Via software connects any audio application or device from a computer to a Dante network and enables USB, FireWire, or Thunderbolt devices—including microphones, legacy mixing consoles, or I/O boxes—to join any Dante audio network. The software also enables a Dante network to be created without the need for dedicated Dante hardware, providing straightforward approach to routing audio using only computers. Sporting an intuitive, easy-to-use "drag-and-drop"-style interface, Dante Via enables rapid discovery and simple connection of devices and applications.

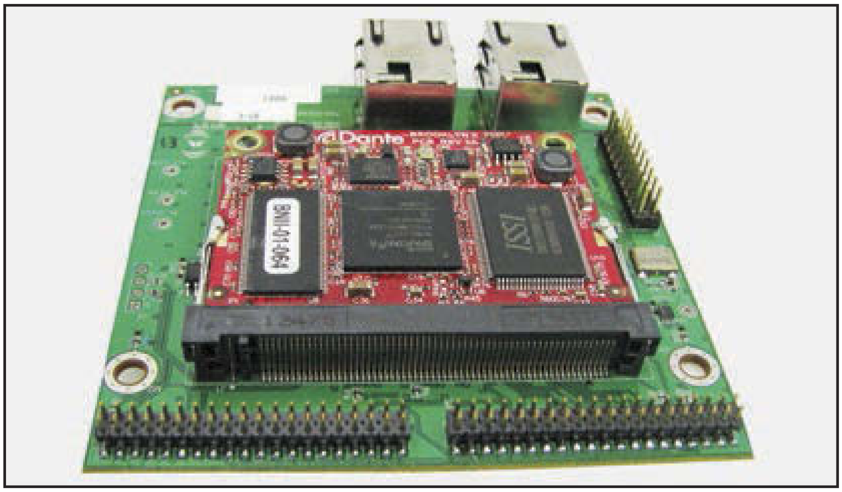
How to Start

Since this a licensed solution, directly supported by Audinate, implementing Dante on a new audio product should naturally start with contacting the company. Licensing policies are not in the scope for this article series but Audinate does make it easy for any company who wants to explore Dante prior to licensing. Audinate's executives attend all major industry trade shows, and they promote educational and training events on a regular basis. At those events (e.g., the Dante AV Networking World conferences promoted throughout the US, Europe, Middle East, and Asia), there's ample opportunity to speak with the company, Dante development partners, authorized implementers, and many of the existing OEMs. Apart from demonstrations

The Ultimo PDK (Product Development Kit) includes hardware, software, documentation, and a technical support package for the design and testing of Ultimo integration projects.

The Dante HC reference design is the industry's highest-capacity networked audio ready-to-use solution for professional AV systems. Dante HC supports up to 512×512 redundant bi-directional uncompressed audio channels on a single Xilinx FPGA with automatic device discovery, one-click signal routing, and user-editable device and channel labels.

The new Dante Via software makes it even easier to connect any audio application or device from any computer to a Dante network. Dante Via enables USB, FireWire or Thunderbolt devices—including microphones, legacy mixing consoles, and I/O boxes—to join any Dante audio network.



Dante implementers like Attero Tech and Auvitran also supply turnkey development to companies that have limited available development resources. Pictured is a module from Attero Tech, featuring a Brooklin II Dante board.

of products in action, presentations, training workshops, and interactive panel discussions, the events provide ample opportunities to discuss the products with other audio manufacturers and meet with installation and systems integration companies. But more importantly, Audinate supplies a full range of development tools, including product development kits and software applications.

For basic technology testing and audio networking information, Audinate supplies the Dante PCIe-R soundcard with supports for up to 256 uncompressed audio channels (128 × 128 redundant channels at up to 96 kHz or 64 × 64 at 176.4 or 192 kHz sample rates) with low round-trip latency, ideal also for recording solutions and audio processing. The Dante PCIe-R soundcard is can be used to displace legacy MADI point-to-point hardware with an advanced networked solution.

For testing, design, and development of Dante-enabled products, Audinate offers the Dante Brooklyn II PDK, which includes the Dante API toolkit and SDKs for integration into PC and Mac software, the latest Dante Controller software, and up to 4 Dante Virtual Soundcard licenses for installation on Windows or Mac and OEM Portal Access.

Access to Audinate's OEM Portal provides manufacturers with documentation and software, plus technical design information and schematics, design and configuration tools, and support information.

The Dante Brooklyn II Gigabit Ethernet module is another solution for easily integrating Dante into new and existing products, featuring a powerful FPGA engine and MicroBlaze CPU that includes its own Linux environment for custom embedded applications. A single Brooklyn II module provides a complete, ready-to-use Dante interface, and can equip a networked audio device with as many as 64 channels of bi-directional digital streaming.

For manufacturers looking for a higher level of integration in applications for low-channel count Audinate provides the Dante Ultimo chip, a highly cost-effective but feature-rich Dante solution, ideal for end-point products (e.g., powered speakers, amplifiers, wall plates, and break-out boxes designs). This chip solution, also firmware upgradable, has been designed for quick and easy integration, and is supported by a comprehensive network-side API and a range of control interfaces. The complementary Ultimo PDK is a comprehensive development platform including hardware, software, documentation, and a technical support package. It features a range of audio and control interfaces (I^2S digital audio, UART, SPI, I2C and GPIO control headers), integrated clock and codec chips, flexible signal routing, USB serial

Software for Dante

Dante solutions are complemented with the Dante Virtual Soundcard and Dante Controller software solutions, supporting the latest Apple Mac OS X and Microsoft Windows operating systems, including Windows Server. These Audinate applications enable users to instantly connect a computer to any Dante network using the computer's Ethernet port to send or receive audio from Dante-enabled devices on the network. The Dante Virtual Sound Card supports sample rates from 44.1 to 192 kHz, for both ASIO and Core audio. Just recently, Audinate also released its Dante Via software, a new tool to expand audio-over-IP networks to a wide range of computer audio devices and applications, without the need for dedicated Dante hardware.

console, and configurable user interface elements such as LEDs and push buttons. Power can be supplied via 5 VDC power supply/USB/Power-over-Ethernet.

Finally, there's the Dante HC reference design, on which OEMs may build upon to create cost-effective AV products. The Dante HC reference design is ideal for AV equipment products requiring high channel capacity (e.g., audio matrix routers, large format consoles, public address and evacuation systems, and large-scale DSPs). This solution supports up to 512 × 512 redundant bi-directional uncompressed audio channels (128 × 128 at 176.4/192 kHz) on a single Xilinx FPGA with automatic device discovery, one-click signal routing, and user-editable device and channel labels. Dante HC also offers a wide selection of interface options including SPI, I^2C, RS232, and configurable GPIO. A powerful onboard microprocessor enables local control and management without the need for any additional CPU.

As Ellison adds, "Audinate recognizes the importance of time to market. Audinate supports its OEMs through its global technical solutions teams to help with issues that could arise during the design, development, and test their Dante-enabled products. Dante OEM partners have access to an OEM portal which contains detailed documentation needed to integrate Dante into the audio products."

"Because the interfaces are all well-defined, we have had some customers develop a whole new range of Dante-enabled products in just a few months," he adds.

For more information about Dante or other products from Audinate, visit www.audinate.com.

Resources

Audinate, "The Three Pillars of Audio Networking," Whitepaper, http://go.audinate.com/resources/assets/pillars-audio-networking-interoperability-f.

Dante-Enabled Product Catalog, Audinate, www.audinate.com/products/dante-enabled.

Focusrite, RedNet Overview, http://us.focusrite.com/ethernet-audio-interfaces/rednet.

Institute of Electrical and Electronics Engineers (IEEE) Standards Association, "1733-2011 - IEEE Standard for Layer 3 Transport Protocol for Time-Sensitive Applications in Local Area Networks," https://standards.ieee.org/findstds/standard/1733-2011.html.

Magma, www.magma.com.

J. Martins, "AES67-2013 - The New Networked Audio-Over-IP (AoIP) Interoperability Standard," *audioXpress*, January 2014, http://audioxpress.com/assets/upload/files/StandardsReview_AES67.pdf.

———, "Audinate Dante (Part 1): Making Digital Audio Networking Easy," *audioXpress*, February 2014, http://audioxpress.com/assets/upload/files/DanteP1AXFeb2014.pdf.

———, "Audinate Dante (Part 2): Audio Networks Fast to Market," *audioXpress*, March 2014, http://audioxpress.com/assets/upload/files/DanteP2AXMar2014.pdf.

Other World Computing (OWC), www.macsales.com.

RH Consulting, "The Death of Analog and the Rise of Audio Networking," Network Audio Whitepaper, Audinate, www.audinate.com/rise-of-audio-networking.

Sonnet Technologies, www.sonnettech.com.

K. Walsh, "Audio Over IP," Audio Engineering Society (AES) Convention e-Brief, presented at the 133rd Convention, 2012.

———, "Dante and AVB Networking," Audio Engineering Society (AES) UK 24th Conference: The Ins & Outs of Audio, June 2011.

Measuring Reverberation Time (RT)

This Sound Control article focuses on reverberation time (RT). Here we provide some historical information and details on how to successfully predict and measure RT.

Photo 1: This Brüel & Kjær Sound & Vibration Measurement A/S Graphic Level recorder represents the type of instrument used for recording sound decays a few decades ago.

By
Richard Honeycutt
(United States)

Wallace Clement Sabine's formulation of the statistical theory of reverberation in the late 1890s is considered to mark the birth of modern acoustical science. The reverberation time (RT) of a room is the time required for a sound to decline by 60 dB from its initial level, after the source of sound is interrupted.

In practice, RT is measured by determining the time required for sound level to decrease by 10, 15, 20, or 30 dB from the level measured about 5 ms after the interruption of the sound source. This 5 ms initial delay is used because high-level early reflections can corrupt RT measurements. The smaller decay range is used because in many rooms, the noise level is so high that an uncomfortably high initial sound level would be required for the sound to decay by 60 dB and still remain above the background noise level.

The measurements obtained are multiplied by the ratio of 60 dB to actual measurement range (e.g., 30 dB), and are designated as T10, T15, T20, or T30. The decibel level of the reverberant sound linearly decreases with time. In this way, if the sound level decreases by 30 dB in 1 minute, the RT (or RT60) is 2 minutes. T30 is also stated as 2 minutes, as the measurement-time-correction factor is included in all forms of RT measurement. (T10 is more often called Early Decay Time, or EDT, and is closely correlated with people's perception of the reverberance of a room.)

RT Theories

Sabine's theory states that the RT of any room can be predicted from a knowledge of the room's volume and total acoustical absorption. For the prediction to be accurate, two conditions must be met:

- (1) The sound in the room must be diffuse, meaning that at any point in the room, there is equal probability of reverberant sound arriving from any direction.
- (2) The acoustical absorption must be evenly distributed among all surfaces: wall, floor, and ceiling.

Given these conditions, the RT is the same at any point in the room.

In 1930, Carl Eyring published a new equation for calculating RT. His formula was derived theoretically, and may be more accurate than Sabine's in "dead" rooms. The coefficients of absorption (α) of material surfaces are different in the Sabine and Eyring approaches, but Eyring α values can be calculated from Sabine α values. The Eyring and Sabine results agree for live rooms.

Later, Heinrich Kuttruff introduced a correction to the Eyring formula. And in 2012, Uwe Stephenson of HafenCity University in Hamburg presented a paper at InterNoise, in which he compared the Sabine, the Eyring, and the Kuttruff approaches, as well as some others.

Real rooms virtually never have a diffuse sound field or evenly distributed absorption. Deep under-balcony spaces in auditoriums and theaters act as separate, acoustically coupled spaces having different RTs than the main seating area. Balcony areas often have a lower RT than the main floor, for the same reason. In most theaters and concert venues, a large proportion of the acoustical absorption is provided by the audience, providing a more absorptive "floor," compared to the ceiling and walls, which are usually acoustically hard. And audience members seated near the stage or loudspeakers are exposed to a higher direct-to-reverberant sound ratio than the average audience member, resulting in a lower measured RT.

Measuring RT

To measure RT, one must have a sound source, a sound level meter, and some way to keep track of time. Sabine used human-blown organ pipes, his ears, and a stopwatch. He learned how to blow the pipes so that the sound level would be about 60 dB above the background noise, and he clicked the stopwatch when he could no longer hear the sound reverberating.

Over the years, balloon bursts and starter pistol shots have been used as sound sources. While these suffice for a general estimate, they did not permit measurement of the frequency-dependent behavior of RT, until computer analysis became available in the latter decades of the 20th century. An interrupted octave- or third-octave-band-limited noise source was used in the mid-20th century, with a graphic level recorder (see **Photo 1**) tracing the level vs. time after the sounds cut off. The results of such a measurement are shown in **Figure 1**. The "wiggles" in the RT curve result from the use of random noise as the test signal. A much smoother, more linear curve could have been obtained by averaging multiple measurements.

Figure 2 shows the Schroeder curve created with a computer analysis of a room's impulse response. (The Schroeder curve is obtained by reverse-integrating the decay curve, and is the actual curve used for RT determination, since it is smoother than the decay curve, due to the small bumps and dips being averaged out by this mathematical operation.) The short blue line from the left circle marker to the center one is the

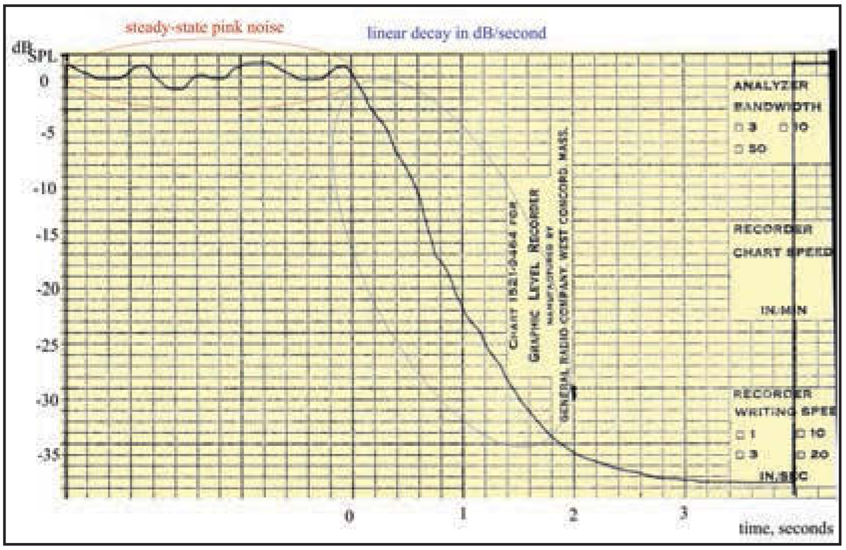


Figure 1: This is a record of the sound decay under the balcony of an auditorium.

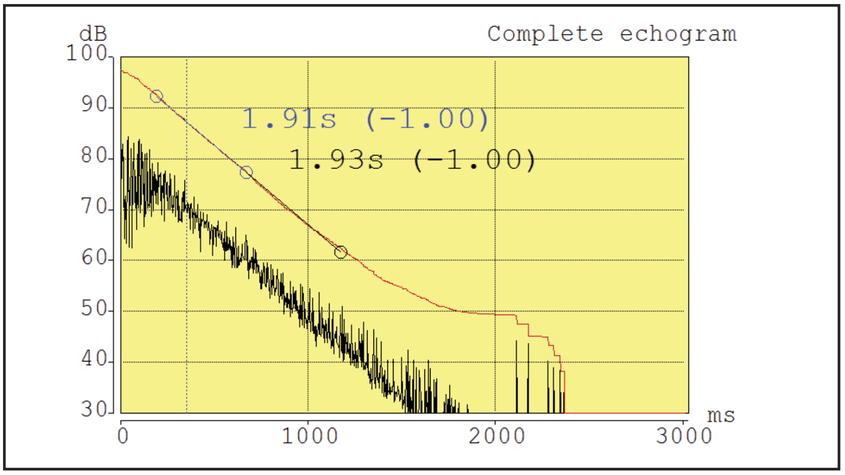


Figure 2: In this computer-generated RT graph, the ragged black line is the decaying random noise signal. The smooth lines represent T15 (blue) and T30 (black).

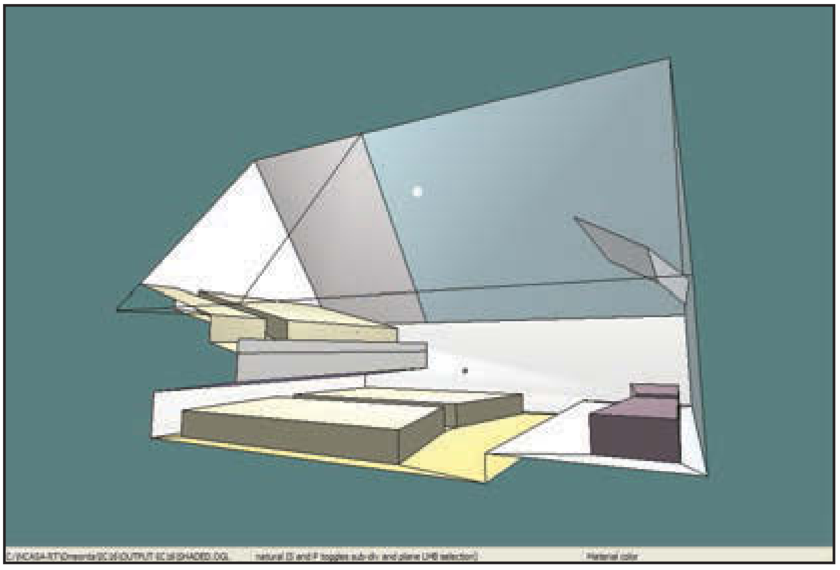


Photo 2: This rendering depicts a computer model of the auditorium in which the RTs presented in Figures 3–8 were determined.

T15 curve. The black line between the left and right circle markers is the T30 curve. In both figures, the curvature of the RT characteristic at longtime values results from the effects of noise in the room.

In a "Sabine-like" room, the location of the source is primarily determined by the need to keep the measuring microphone at least one "critical distance" (d_c) away from the source. The critical distance is the distance from the source at which the direct sound level equals the reverberant sound level. It is a function of the source directivity and the total absorption in the room. (In a 100′ × 73′ × 24′ room with a 2-kHz RT of 2 seconds, d_c for an unamplified talker is about 13′).

In a real room where absorption may not be evenly distributed among walls, floor, and ceiling, two commonly used locations for the source are (1) as near as possible to the horizontal and vertical center of the room, and (2) near one corner of the room. These locations excite room modes well enough to give an accurate measurement.

Before RT is measured, the background noise in the room much be reduced as much as possible. Often, this means vacating it of people and turning off the HVAC system. If the room is served by any background music or paging systems, these should be disabled as well. One noise source that is easy to overlook is masking noise in offices. Masking systems

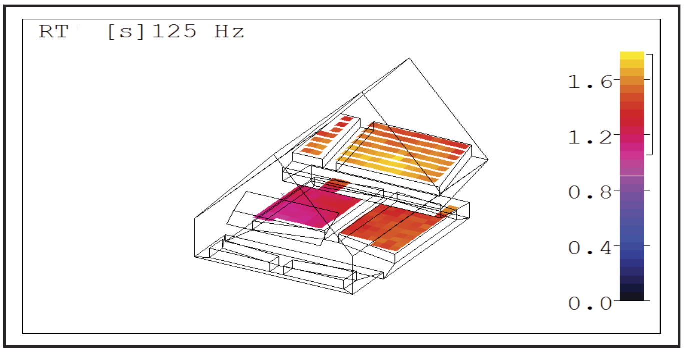


Figure 3: The auditorium's 125-Hz RT is mapped here, with a centered source 20′ above the floor.

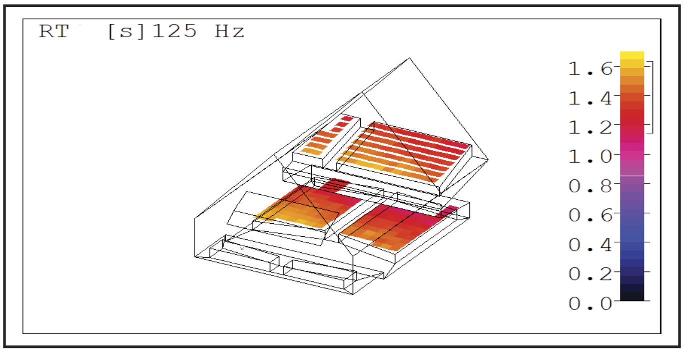


Figure 4: This is the 125-Hz RT with a source in the lower house-right corner.

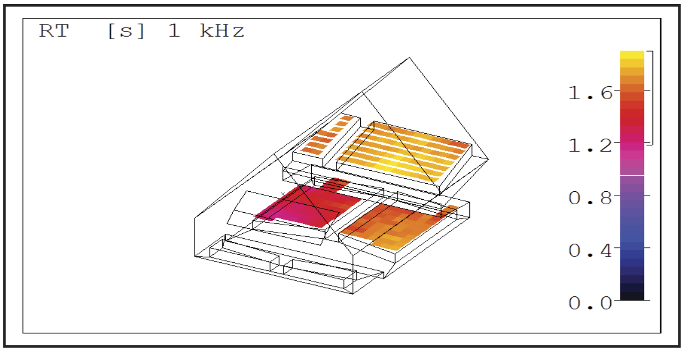


Figure 5: The 1-kHz RT is shown here, using a centered source.

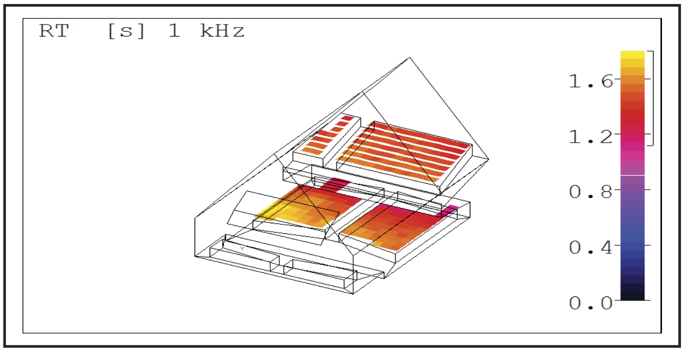


Figure 6: This is the 1-kHz RT with a corner source.

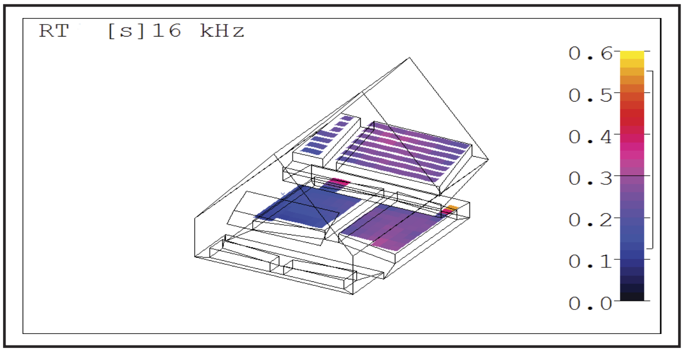


Figure 7: Here is the 16-kHz RT, using a centered source.

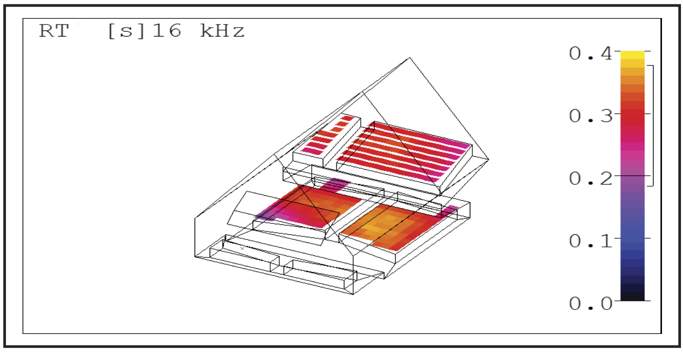


Figure 8: The 16-kHz RT is shown here, with a corner source.

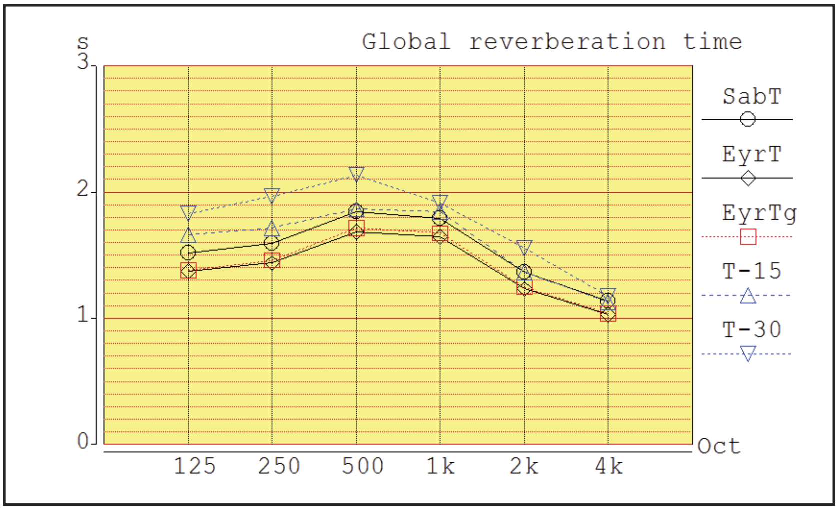


Figure 9: This graph compares three predictions of octave-band RT versus frequency: Sabine, Eyring, and Eyring geometric (Tg), with T15 and T30 measurements.

should not be used in auditoria or conference rooms, but sometimes they are, and they should be turned off before RT measurements are made.

Many RT-measuring instruments include a function that checks the level of background noise before RT measurement is commenced. This facilitates setting the RT sound source level high enough to allow proper decay before the noise floor is encountered. Even when such a function is used, impulsive noise can still contaminate a measurement by beginning a new attack/decay cycle after the RT measurement has begun.

RT in a Non-Sabine-Like Auditorium

Photo 2 shows a rendering of a "non-Sabine-like" auditorium. **Figures 3–8** illustrate the variation of RT throughout this auditorium. For **Figure 3**, **Figure 5**, and **Figure 7**, the source was centered horizontally, at a height of 20′. The height at the ceiling peak is 57′. For **Figure 4**, **Figure 6**, and **Figure 8**, the source was near the house-right front corner near the floor.

As you can see from the figures, source location affected the RT at specific locations, although the average RT in the room was affected very little by source location, except at the highest frequencies. Frequency made a significant difference in the measured RT.

Specifically, the RT in the balcony measured somewhat higher than on the main floor, a common result of the balcony area not being fully coupled to the rest of the room, and thus behaving as part of two coupled volumes. Under-balcony areas with acoustical absorption on the rear wall are particularly subject to showing a lower RT than the rest of the room. Also, RT decreases with increasing frequency, due to most materials' absorption increasing with frequency. This effect is especially visible if you compare **Figure 7** with **Figure 3**.

Figure 9 shows the average RT of the entire room, along with the calculated values using the Sabine, Eyring, and "Eyring geometric" (Eyr Tg) equations. (The Eyring geometric calculation uses the area-weighted mean absorption coefficient, giving slightly different results from the original Eyring formula.)

In this plot, we see the maximum RT centered around the fundamental speech frequencies. This is a bit unusual, and is not good for speech intelligibility. The lower RT values in lower octave bands result from the use of gypsum wall board for most of the walls, and the lower RT at 2 kHz and above comes from the audience and the carpet being more acoustically absorptive at high frequencies.

If the RT were plotted for higher frequencies, the value would likely decrease further due to the air's absorption of sound at higher frequencies. This effect is especially significant in rooms having large dimensions.

The disagreement between Sabine and Eyring values stem from the different assumptions made in deriving the equations for calculating these predicted values. T30 being longer than T15, as in this case, usually results from concave curvature of the Schroeder curve caused by either background noise or strong late-arriving reflections.

Figure 10 illustrates the effect of measuring RT too close to the source, which in this case was located a bit toward house-right of the horizontal center of the room, 10′ above the floor. The region of very low RT can be clearly seen.

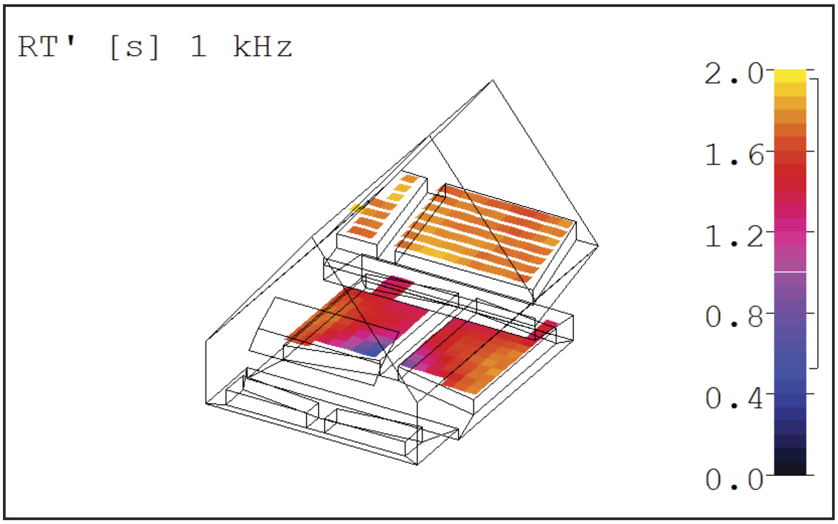


Figure 10: The very-low-RT area shown by the blue patch near the center results from the signal source being too near the area whose RT is being determined.

Simon Saywood is the owner of Analoguetube, Ltd., a London, UK-based company that manufactures classic versions of the 1950s Fairchild 660 and 670 mono and stereo limiters.

Recreating the Fairchild Legends

An Interview with Simon Saywood, CEO Analoguetube

By
Shannon Becker
(United States)

SHANNON BECKER: Your company, Analoguetube (www.analoguetube.com), received a lot of interest at the 139th Audio Engineering Society (AES) Convention. Tell us a little about Analoguetube.

SIMON SAYWOOD: Analoguetube has been in business for seven years when it was incorporated (formalized into a limited liability company) in 2008. However, its roots go back to 2002 when the re-development of the Fairchild 670 limiter began. Enter the golden age of electronics. The repair of vintage equipment is a tricky business, preserving the original function and tone of the equipment is key, while making it usable for the rigors of modern recording. Fortunately, there have been many advances in passive technology that help make this all possible.

It all began with the repair an original 670 that had been standing idle for many years. The repair and strip down of this original classic would have involved a power-up after inspection, but there were quite a number of audio transformers and components missing! Further investigation revealed the source of the problem had been faulty wiring around the tube power supply. I could see that this unit had not worked for quite a while, particularly as some of the original soldering had broken up around the solder pins of the E80F tube regulating the B+ supply voltage. Interestingly, it may have been a factory-born issue! While much of the unit had been modified over the years, some of the original soldering still remained. Eventually, after several months of investigation, four transformers, a bunch of capacitors and resistors, and the replacement of most of the wiring, the unit burst into life.

SHANNON: The AT-1 is a classic version of the 1950s Fairchild 660 compressor/limiter. The AT-101 is a recreation of the original Fairchild 670 stereo compressor/limiter. Why did you want to recreate the Fairchild designs?

SIMON: The Fairchild 660 and 670 compressors were extravagant audio pieces of their time—pieces that are wonderful in operation and complex in design. In operation, they exhibit interesting and dynamic ways of compression, particularly the 670, limiting a wide band of mid-frequencies and providing a spatial image as well as tone. Many original models still exist today as working units and they command a

The AT-1 is a beautiful and natural sounding Mono limiter that uses the original circuit, and all original parts including the transformers, controls, and the new generation 6386LGP triodes developed from the original GE6386 tube types.

The AT-101 is a recreation of the original Fairchild 670 stereo limiter. It is available in several colors including the blue unit shown here.

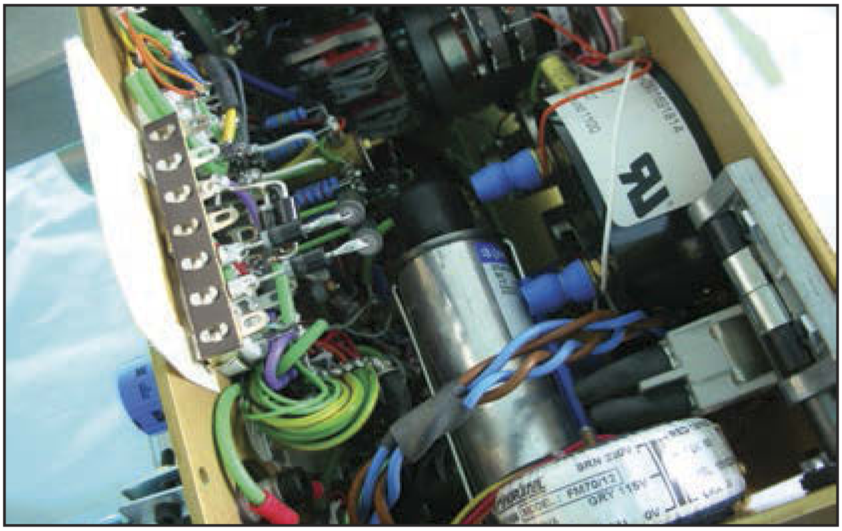
very high price. There are also similar versions on the market, but few can claim to be entirely hand-wired using point-to-point wiring throughout with no PCBs. I wanted to re-make these units to include Teflon wiring to the original wiring specifications, since many of the original units suffered from intermittent and decaying insulation caused by a combination of age and high temperature. Other issues included aging components and faulty transformers. Originally introduced at the beginning of the solid-state revolution, some conservative estimates make these units almost 60 years old!

SHANNON: What was your inspiration for building them?

SIMON: I was fascinated by the original designer Rein Narma and his pursuit of audio excellence. Rein built the first 10 mono units himself. The first one was made for Rudy Van Gelder, a recording engineer specializing in jazz. Rein later built a recording console for him. The second 660 was sold to Olmstead Recording in New York and the third was sold to Mary Ford and her husband Les Paul. Rein worked at Gotham Recording and then formed Gotham Audio Development Corp. with others. He built new amplifiers for the Ampex 300, a new product range for Gotham Audio.

After investigating this more closely, I could see there were many versions of the same 670, some with modifications, some with changes to the AC threshold and power supply, others with additions of PCBs, and so forth, but none that were all tube and fully wired as the originals were using the original wiring turrets of the 1950s. It was a challenge.

Here is a peek inside the AT-101.



SHANNON: Were you able to replicate the same sound as the Fairchild 660 and the Fairchild 670?

SIMON: There are many subtle features that form these super audio heavyweights, which made the recreation process all the more important as we tried to get as close as possible to the original performance, both functionally and tonally. Both units—the 660s and the 670s—are like cakes made with lots of ingredients. If you remove one part and replace it with something different (e.g., a modern equivalent), a small part of the tone or even the function may change. So it was very important to include everything that the originals had, right down to the original tube power supply.

SHANNON: You use original parts that include the transformers, controls, and the new generation 6386LGP triodes developed from the original tube types. How difficult is it to find the original parts?

SIMON: I try to use as much new-old-stock (NOS) tubes as possible but it's important to remember that part of the philosophy behind these new units is that they are all consistent with each other in function and tone and that everything is fully available off the shelf. There should be nothing that is difficult to get hold of and all the parts are current. The availability of some tubes was a problem, particularly with the original 6386 gain reduction tubes, which have been obsolete for decades. So in the early days (before 2008), I worked closely with JJ Electronics to help redevelop the 6386 tube, which was not an easy task. Lots of data spreadsheets went back and forth as did many hand-built triodes until we were happy. The original saturation transformers, running the regulated AC supply for the 6386 heaters, for example, took 18 months to remake.

SHANNON: Can you share some more details of the challenges involved with the design?

The AT-101 uses all original parts including the transformers, controls, and the new generation 6386LGP triodes, which were developed from the original tubes.

SIMON: While the 660 was the first model historically to come out, its bigger brother the 670 was the first to be redeveloped. This was mainly due to the popularity and developmental aspects of the stereo unit as actual work for this model began in 2002!

Some of the challenges in redeveloping these stereo units involved the re-introduction of the 6386 gain reduction tube. This particular type known as a remote cut-off tube owes its action in part to the structure of the grid, which provides a variation in gain with a change in bias. The control grid is wound with a wide spacing at the center and a narrow spacing at the top and bottom or sides of the winding. It was crucial to have this spacing correct as it affects the performance of the gain reduction. Other challenges included the audio transformers that were also key to the sound, so we used Sowter transformer in both models again constancy is very important here as we made lengthy comparisons between the original units and the completed prototype.

It also meant not replacing parts of the circuit with solid-state versions. We also did not use PCBs. There is point-to-point wiring throughout as per the originals, including all the original hot molded carbon composition controls and original stepped attenuators.

The Mid-Sid (lateral/vertical) switching was an important development to these new models as much of the original wiring was fed up to the front panel where all the switching was done on a dual wafer open-type switch. This has been changed so the lateral/vertical switching is now done through the use of four removable relays. No more complex wiring!

At the time, the stereo unit took precedence over the mono unit, which occurred before the launch at the 125th AES Convention in San Francisco, CA, in 2008. The AT-1/660 models came out several years later as more work was needed to understand these

This is a sampling of some of the parts that go into the AT-1 and the AT-101, prior to assembly.

units. It's interesting to note that the functionality and the tone is not the same as the AT-101/670 because they were fitted with different transformers and tubes. However, the GR stage (6386 tubes) remained the same for both types. The AT-1 mono limiters were launched in 2014 at the 137th AES Convention in Los Angeles, CA.

Analoguetube's philosophy behind production has always been that the limiters should be technically and historically accurate to the originals. This includes the use of all parts being freely available across the board with nothing difficult or obsolete used as this ultimately ensures a lifetime of use. In terms of typical functionality and operation, historically, the stereo AT-101/670 units behave in a slightly less aggressive way when compared to the mono units. The mono units have a little more gain and more presence in the mids than the stereo units perhaps even more functionality (with the additional bypass, stepped AC threshold and key insert switches), but when "driven" compare a little more to the stereo units! Typically both models are highly transparent in operation revealing a wonderful tone in operation, with tone as the key. Production time for each model is around 80 hours per unit, which includes lifetime warranty excluding the tubes.

SHANNON: What modern features have you added to the AT-1 and the AT-101?

SIMON: The unique features allow all types of instrument and sound to sit up front in the mix, sounding fatter and bigger bringing the material forward to add depth, dimension, and clarity to your music. The AT-1 glues the image together for all types of program, making this natural sounding compressor an indispensable tool. The AT-1 has several new features adding more functionality including a stepped AC threshold, external Key input, a stereo link, and a bypass.

The AT-1 also provides a repeatable stepped 21-position AC threshold control particularly useful when linking two units together and recalling tracks. The control has a 1%/div accuracy. The External Key input provides access to the compressor side chain input with a second balanced input that is particularly useful for De-essing, Ducking, and recording Stems. The AT-1 has a switch-able stereo link where you can link two units together for linked stereo operation. The link operation is the same as the stereo AT-101 where either left or right channel has control. Side chaining opens all sorts of possibilities. For example in De-essing one can heavily EQ the control signal i/p so only when selected frequencies appear at the ext key i/p, does the compressor kick in. Such a compressor can be used as a de-esser, reducing the level of annoying vocal sibilance in the range of 6 to 9 kHz. This arrangement involves a standard AT-1 compressor and an equalizer by feeding a 6 to 9 kHz-boosted copy of the original signal into the side-chain or external i/p input of the compressor. Finally, the AT-1 has a full bypass function where the unit can be switched fully out of the signal chain during operation. The CVT transformers are standard on all units. This is a world-wide transformer providing a regulated heater voltage of the critical 6386 tubes during operation as per the originals.

The AT-101 uses the original circuit and is technically and historically accurate to the Fairchild 670. The AT-101's features include a stepped AC threshold, a full Bypass function, and a lateral/vertical switch, which is a lovely function providing full sum and difference compression.

SHANNON: Tell us a little bit about your background.

SIMON: I was schooled in London during the early 1970s. I boarded from the age of 8 at a school that was considered by many to be at the forefront of learning, where the principles of learning took on a holistic view of the individual child's progress and which recognized their uniqueness, capacity, and potential. My secondary education was in North Wales where I continued to board till the age of 18 when I left school in 1980. I lived and worked in Cambridge for a few years with a lab equipment manufacturer before moving back to London to work as a technical support engineer at Dreamhire—a recording equipment rental company in Willesden, North London. This included Battery studios and Zomba records. In those days, there were many rental companies, particularly in London. Nine years later, I Joined HHB Communications as a Technical manager for its service dept. In 1998, I joined Metropolis Studios as a technical engineer looking after the recording equipment and studios.

SHANNON: How did you become interested in audio?

SIMON: My parents were very interested in music, and my father, in particular, was interested in hi-fi. My family's home audio equipment was a Quad system with a large electrostatic ELS57 speaker. Yes, in mono, but this was later upgraded to stereo. The system included an all-tube Quad 11 amp, control unit and an FM tuner. We played records on a Garrard 401 turntable with SME pick-up—all standard fair for the hi-fi interested. We loved music and listened to it every day.

I do remember from an early age being introduced to basic mechanical engineering and electronics with technical manuals in the family house. My father worked for various electronics companies including Marconi in Chelmsford. I remember many of his electronics projects and going to trade shows with him. My mother worked for an aircraft freight company where I was introduced to salvaging aircraft electronics. During the school holidays, I had access to hundreds of aircraft transceivers, instrumentation, and various bits of navigation equipment that were all dismantled including a "Black Box" that was de-soldered and investigated in a shed at the end of the garden.

Sitting side-by-side in the studio is an original Fairchild 670 stereo limiter and the AT-101 recreation of the 670 classic.

SHANNON: What's next for your company? Do you have any other projects in the works?

SIMON: More AT-1 and AT-101 limiters! I would like to expand as the company is quite small and offer a range of other studio products. I have a couple of projects in development at the moment, which will be out next year. One will be an audio companion for the AT-1 and the AT-101, but this will hopefully have some universal use. The other is a smaller mid-priced piece, which I hope to debut at the next AES show.

Three-Way Loudspeakers with Active Crossovers

Some projects take longer than anticipated. A couple of years ago I decided to build a pair of three-way woofer-midrange-tweeter-midrange woofer (WMTMW) loudspeakers, using compatible drivers from Dayton Audio and an external, to the loudspeaker enclosure, active crossover network.

Photo 1: A view of the completed three-way loudspeakers on their "stools."

By
Ron Tipton
(United States)

The first step for this three-way project was to choose a tentative pair of crossover frequencies

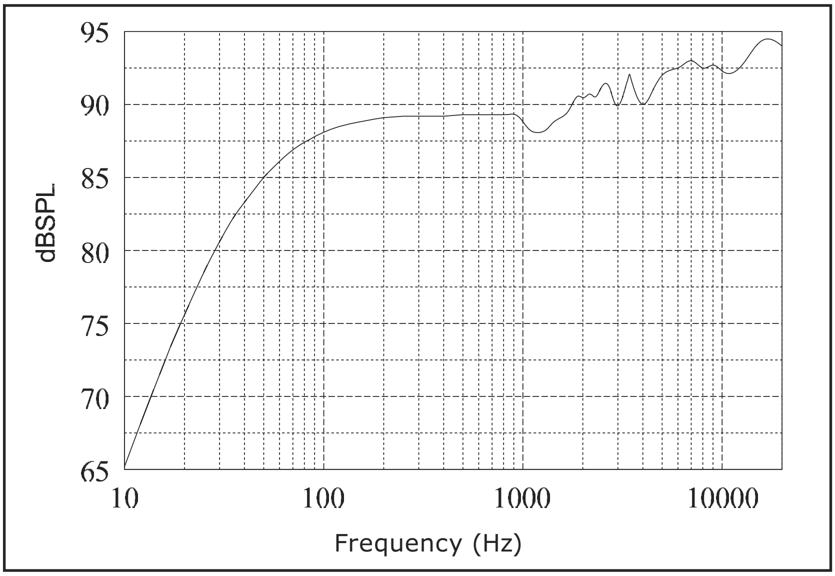


Figure 1: This is a BassBox Pro simulation of the three-way loudspeaker frequency response using the 850 and 3,450 Hz crossover frequencies.

and then look at the specification sheets to find the drivers. Eventually, the project morphed into using these as the left and the right loudspeakers in a 7.1 surround system (see **Photo 1**), but more on this later.

For drivers, I finally settled on the Dayton 8" Reference Shielded Woofer (295-366), the Dayton 2" Dome Midrange (285-020), and the Dayton Rear-mount 0.75" neodymium dome tweeter (ND20FB-4) with crossover frequencies at 850 and 3,450 Hz. A Bass-Box Pro simulation showed a fairly flat frequency response (see **Figure 1**).

The Design Process

Both the midrange and the tweeter have rear seals so the box design is set by just the woofers. Entering the woofer parameters into WinSpeakerz gave a closed-box Q of 0.7 and a system resonance of 49.7 Hz for an internal volume of 2.1 ft^3.

I wanted a narrow front width to minimize diffraction distortion so I chose an outside width of 9", just barely wide enough for the 8" woofers. Getting the WMTMW stack as close as practical gave me a height of 35″, which resulted in an outside depth of 11.5″.

Photo 2 shows the boxes under construction. I used 0.75″ 12-ply Baltic birch plywood for the front and rear panels and 0.75″ MDF for side panels, the top, and the bottom. I added the cross bracing to improve the side panel stiffness. I glued all the joints and then additionally sealed them with 25-year construction caulk.

The woofers and the midranges are front mounting so I used T-nuts to ensure the mounting was both easy and secure. But the tweeter is rear-mounting so it presented a challenge. Because I was using an external crossover, I needed three connector cups in the rear panel. So, I positioned the tweeter cup directly opposite the tweeter mounting hole. By using a flashlight and a long-blade screwdriver, I was able to mount the tweeter with four wood screws. I gave the completed boxes two coats of clear, satin polyurethane varnish before mounting the drivers and completing the wiring to the connector cups.

The woofers and the midrange are wired in parallel for about a 4 Ω driving impedance. Using the voltage divider method described by Joseph D'Appolito in his book *Testing Loudspeakers*, I measured the woofers' Q_{TS} as 0.72 for one box and 0.68 for the other—both measurements were close to the 0.7 design specification.

Photo 2: The three-way loudspeaker enclosures are shown before I attached the rear panels. Note the internal bracing, the T-nuts, and the black rectangles. These are 1/16″ thick pieces of barium-loaded vinyl to help reduce internal reflections.

The Crossover Networks

These are active circuits, that is, they use op-amps, resistors, and capacitors to realize the low-pass, band-pass, and high-pass filters for the woofers, midrange, and tweeters, respectively. Higher-order active filters are smaller and can have

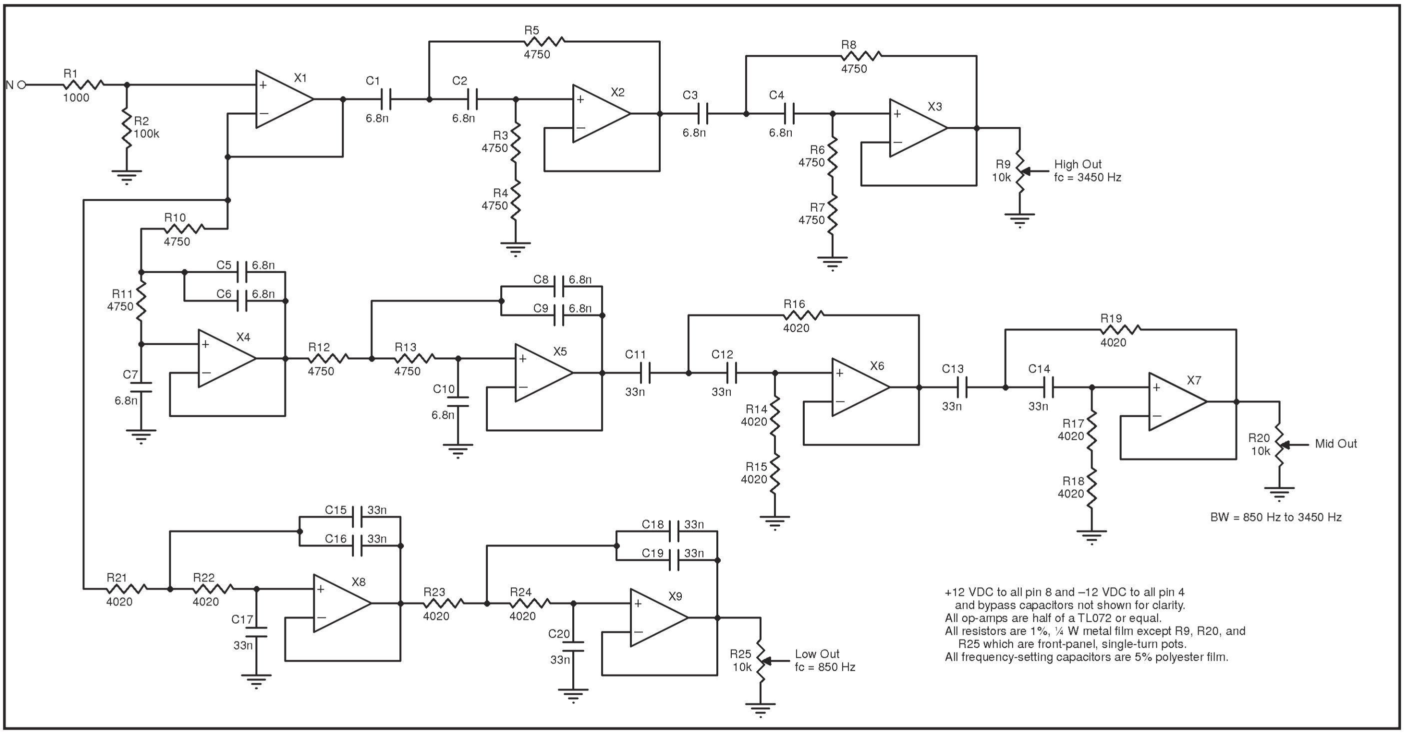


Figure 2: The active crossover circuit for the three-way loudspeakers has a Linkwitz-Riley response. I built this circuit on two mono boards from Elliot Sound Products.

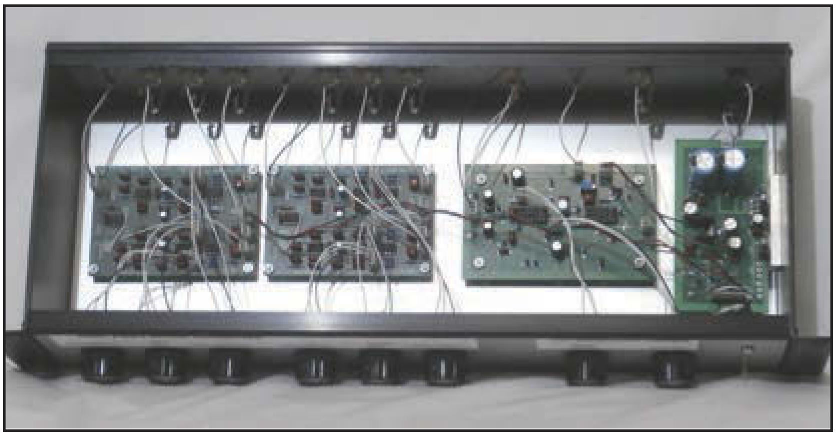


Photo 3: Here is an inside view of the active three-way crossover networks. The two boards on the left are the Elliot Sound Products' three-way mono networks. The power supply board is on the right.

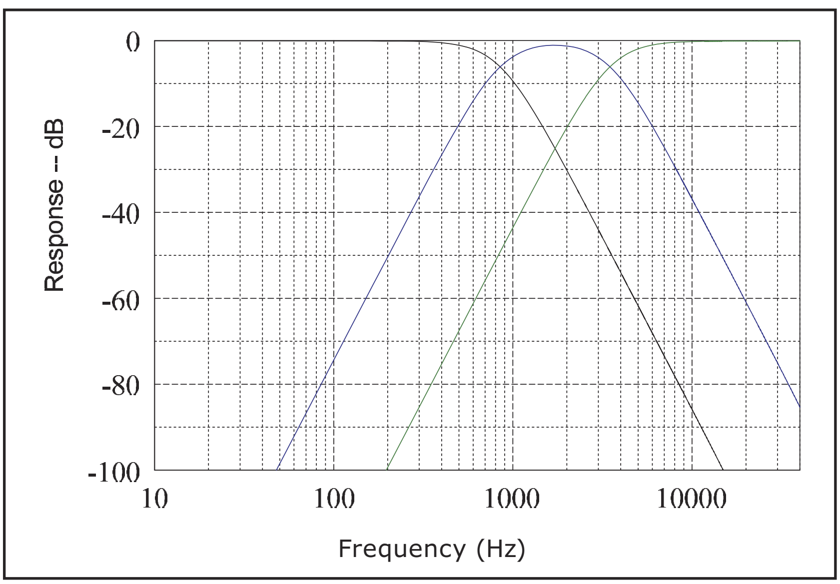


Figure 3: This is a SPICE simulation of the Linkwitz-Riley active three-way crossover network. The measured –6 dB crossover frequencies are within 10 Hz of the design values.

Photo 4: I put the woofer, the midrange, and the tweeter amplitude setup controls on the dual three-way active crossover front panel, but this is not necessarily a safe location, unless you live alone.

closer tolerances than their passive counterparts, especially when the active circuits are built with ±1% metal film resistors and ±5% polyester capacitors.

I chose a four-pole Linkwitz-Riley (24 dB/octave) design for good frequency-band separation and also because I decided to use the crossover circuit boards from Elliot Sound Products. Two of his boards are needed for a pair of mono, three-way networks. They are fairly inexpensive and easy to build.

Figure 2 shows the circuit diagram. **Photo 3** is an inside view of the completed unit with the two crossover boards on the left side. The power regulator is on the far right. It supplies well-filtered ±12 VDC to all the boards from a 24 VAC wall transformer. **Figure 3** shows the crossover SPICE simulation and the measured crossover frequencies were all within 10 Hz of the design goal. The in-between board is the dual voice coil subwoofer controller, a simplified version of the project described by Daniel Ferguson in "A 12" Dual Voice Servo Subwoofer System," (*audioXpress*, September 2004).

I don't have any children or other people with curious fingers who could play with control settings so I placed the crossover output amplitude pots (R9, R20, and R25 shown in **Figure 2**) on the front panel (see **Photo 4**). You may want to choose a different location!

The Power Amplifiers

A crossover network with six outputs needs six power amplifiers—three for each of the loudspeaker boxes. By the time I got around to working on this part of my project, I had decided to expand my original stereo design to a 7.1 surround system. I found that I could conveniently fit eight power amplifiers into a standard rack-mount box if I used a single IC for each amplifier. I chose the 56 W LM3876A because I already had them on-hand. I designed a circuit board for two amplifiers, and four boards just fit the available space (see **Photo 5**). The circuit board files and parts lists are included in the Supplementary Material for this article.

I set the voltage gain of each amplifier to 20 dB by my choice of R3 and R4 in the circuit diagram (see **Figure 4**). I have used this IC before and its harmonic and intermodulation distortion numbers are very low. It is rated for a maximum operating voltage of ±42 V but this is not necessary for good performance so I used ±30 V from the toroidal power transformer and filter capacitor supply shown at the front of the enclosure (see **Photo 5**).

Keeping the IC amps sufficiently cool presented some challenge. Visible near the top of **Photo 5** is the top edge of a 0.25" × 1" copper bar to which each of the IC amps is attached using insulators and a single machine screw.

The copper bar is thermally attached to the external, finned aluminum heat dissipaters with five short lengths of 1″ diameter copper rod soldered to the copper bar. The rods pass through holes in the enclosure's real panel and are fastened to the dissipaters with machine screws. This arrangement works fine for listening to music but is woefully inadequate for full-power operation.

The front panel is very simple—just the power on/off switch and a power-on indicator. The handle screws fasten the panel to the enclosure. They are among the parts I used when repurposing this box from a defunct commercial power amplifier. I used

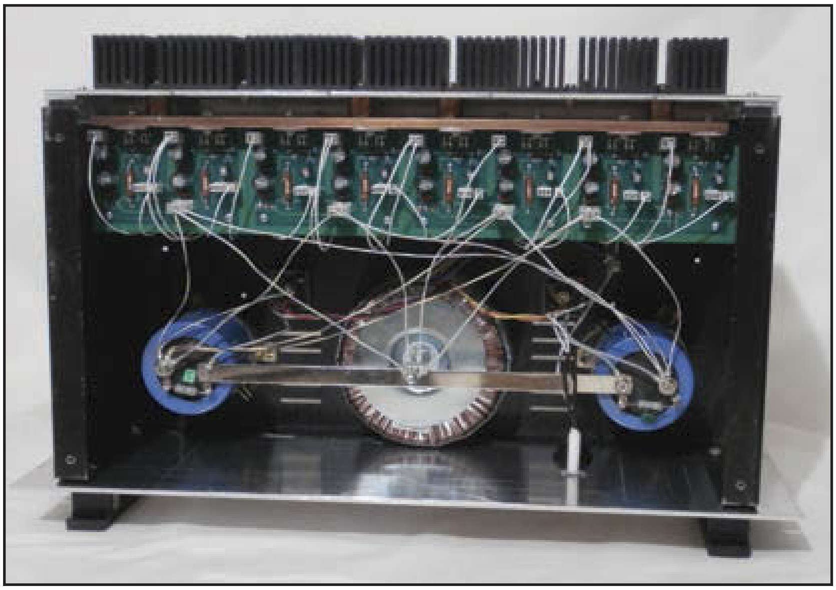


Photo 5: The eight-channel power amplifier inside view. The circuit board files, circuit diagrams, and parts lists are included in the Supplementary Material.

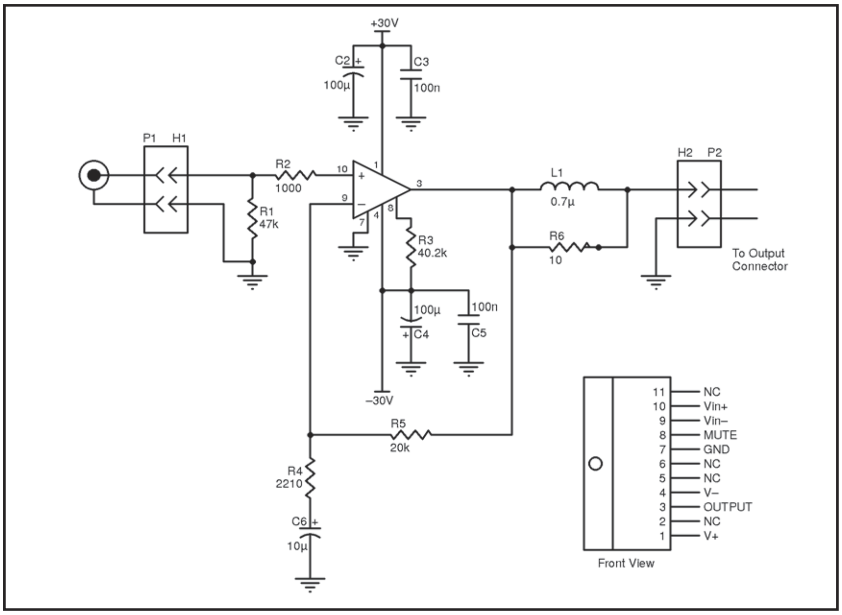


Figure 4: This LM3876A power amplifier circuit was used eight times in my three-way loudspeaker amplifier. The parts list is included in the Supplementary Material.

Photo 6: The finned aluminum heat dissipaters are surplus items. I don't remember the source but I expect you can find a similar part. I used 0.25″ mono phone jacks for the three-way loudspeaker inputs and outputs.

Project Files

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

Resources

J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Publications, December 1998. [Available at www.cc-webshop.com, Item #: AA-BKAA045]

C. Struck, "Group Delay," CJS Labs, groupdelay.pdf.

D. Ferguson in 12″ Dual Voice Servo Subwoofer System," *audioXpress*, September 2004. [audioXpress articles from 2001 to present are included on The Cache, www.cc-webshop.com]

R. Tipton, "Signal Dynamics and Loudness," *audioXpress*, May 2015.

Randy Yates, DSP Engineer, Ericsson, Inc., www.ericsson.com.

Sources

Project P09
Elliot Sound Products | sound.westhost.com/purchase.htm

BassBox Pro software
Harris Tech Audio | www.ht-audio.com

Dayton 8″ Reference Shielded Woofer (295-366), Dayton 2″ Dome Midrange (285-020), Dayton Rear-mount 0.75″ neodymium dome tweeter (ND20FB-4), and OmniMicV2
Parts Express | www.parts-express.com

LM3876A Class-D amplifier
Texas Instruments, Inc. | www.ti.com

WinSpeakerz software
True Audio | www.trueaudio.com

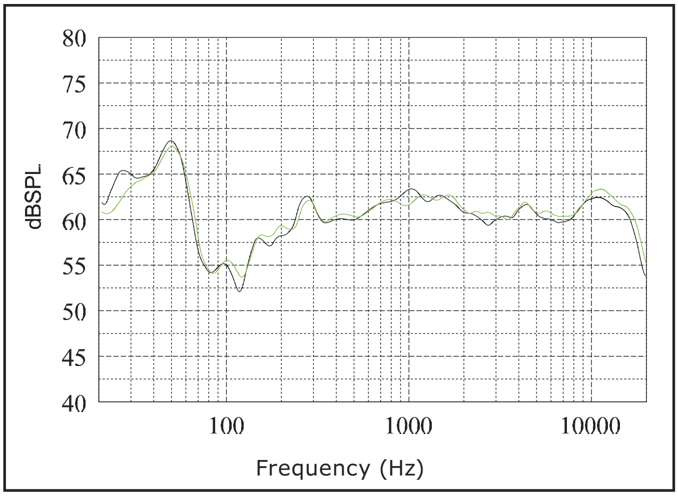


Figure 5: I measured the three-way loudspeaker amplitude response using the Dayton OmniMicV2 system. All room reflections are included, that is, the impulse response was not time windowed. The black line is the left enclosure and the green line is right.

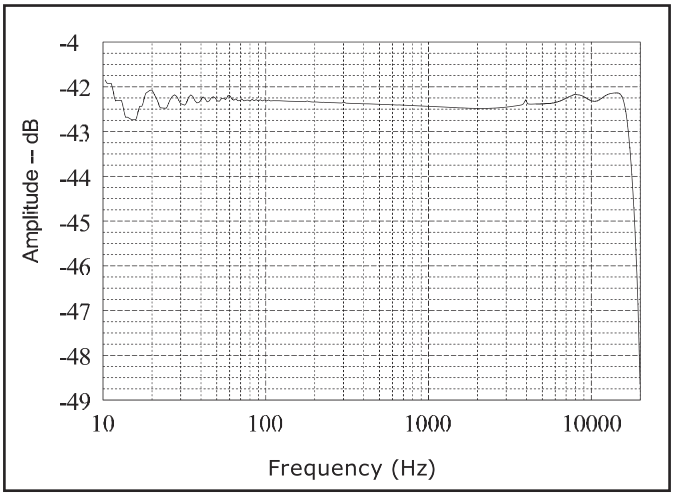


Figure 6: The measured frequency response of the short-duration "chirp" was produced by the OmniMicV2 system as the three-way loudspeaker measurement input. Note the rapid amplitude decrease above about 15 kHz.

0.25" mono phone jacks for the six crossover outputs and three-way loudspeaker inputs on the rear panel (see **Photo 6**). The other two channels, which I'm using for the center and subwoofer, have male RCA inputs and 0.25" phone jack outputs.

Testing the Three-Ways

My listening room is small for a 7.1 surround system, just 10.5' × 11.5' with a ceiling that slopes from 8' to 9' at the opposite side wall. I chose to test the three-way loudspeakers in place by adjusting the crossover outputs for the flattest frequency response including the room reflections. The Dayton OmniMicV2 system has this option.

Figure 5 shows the amplitude responses. They are not especially flat below 200 Hz because of the room size, shape, and the reflections. At this time, I have not made any effort to minimize the reflections, which is difficult to do at low frequencies. The sound quality is good so I will probably just accept what I have. The rapid drop off of amplitude above about 15 kHz is because the input test signal from the OmniMicV2 system also has an amplitude decrease in this frequency range (see **Figure 6**).

Figure 7 shows the group delays. It's important to note they are always less than 1 ms. Most authorities agree this makes the delay distortion inaudible. Group delay can have negative values—the short answer for this is because the group delay function contains derivatives. Randy Yates explains it this way:

"It's just basic calculus. Group delay is defined as G(omega) = −d theta(omega)/d omega. (See e.g., Proakis, 'Digital Signal

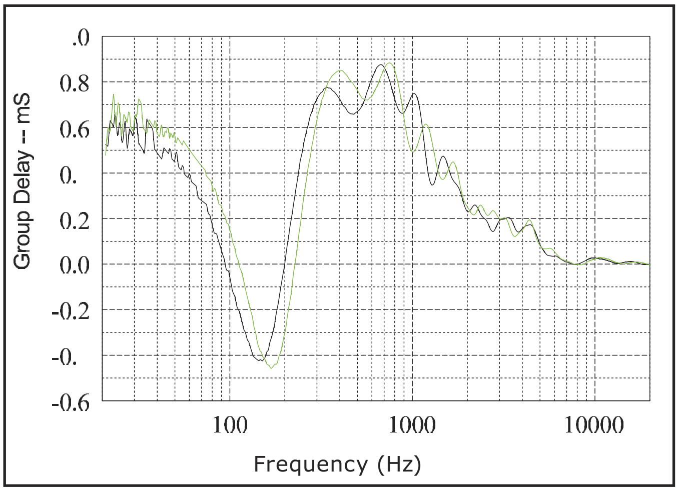


Figure 7: The three-way loudspeaker group delay was calculated from the phase and frequency responses measured by the OmniMicV2 system. The black line is left and green line is right.

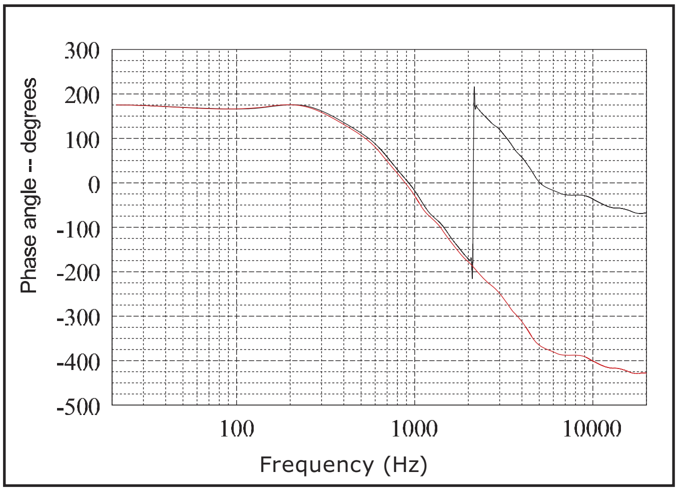


Figure 8: Here is the left three-way loudspeaker phase response. The black line is the measured wrapped phase and the red line is the unwrapped.

OmniMic

The Dayton OmniMicV2 system consists of the items shown in **Photo 1** plus the software CD, but not the laptop computer. Each microphone has a unique serial number and its calibration file can be downloaded from Dayton Audio's website (www.daytonaudio.com) using the serial number. Click on "Test & Measurement," then click anywhere in the OmniMic portion of the screen. Scroll down to enter your microphone's serial number.

OmniMic is a versatile acoustic measurement system. Its frequency response screen is shown in **Photo 2**. The frequency response is shown across the middle part of the screen with the impulse response shown below. The portion of the impulse response highlighted in red corresponds to the displayed frequency response. This length can be changed to include or exclude room reflections by dragging the red-black junction back and forth or by changing the number in the box under the Distortion tab at the top of the screen.

The required CD track or tracks for the test signal to your loudspeaker is shown just below the main menu bar. The "sine sweep" radio button should be selected for accurate frequency response measurements. The "all" button includes all room reflections. The "only to" button selects just the reflections in the red portion of the impulse response and "blended" blends the "only to" calculation at higher frequencies to the "all" calculation at lower frequencies. The "smoothing" selection ranges from 1 octave to 1/96 octave to vary the amount of detail shown. You can include the phase response by selecting its checkbox.

Frequency response measurements can be saved in frd format using the "File, Save" menu. Each line of the saved file will have frequency, amplitude response and phase (if it has been selected) separated by a Tab character. An frd file is just a text file and it can be imported into a text editor, a spreadsheet, or a graphing program.

This brief introduction is just an overview to get you acquainted with the software. It is fully described in its 61-page Help Manual, and a copy is included in the Supplementary Material.

Photo 1: The calibrated USB microphone, the USB cable, the microphone holder, the tripod, and the software are included with the OmniMicV2 system. The computer is not included. (Photo courtesy of Dayton Audio)

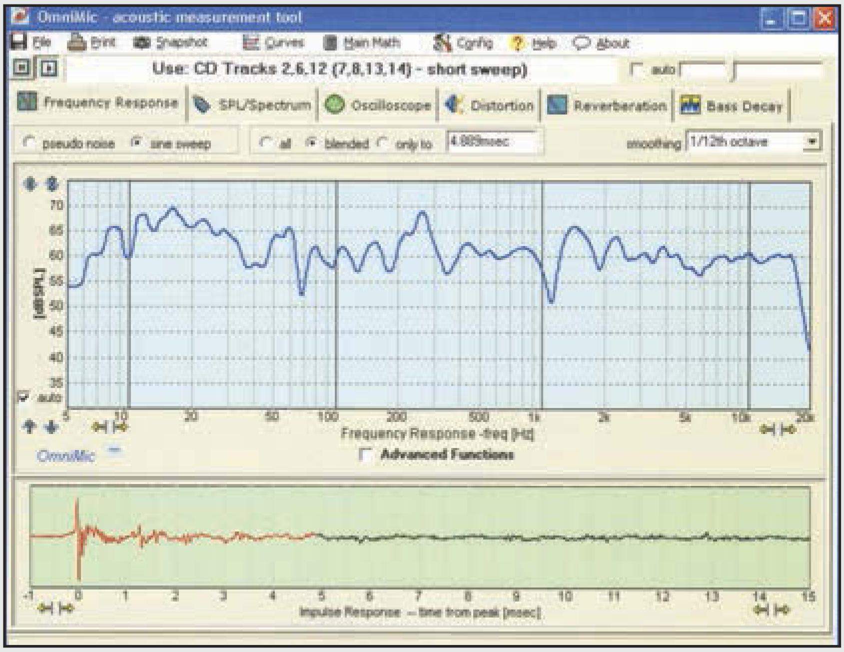


Photo 2: The main screen shows the Frequency Response selection. This screen allows complete program control, depending on which tab is selected. (Photo courtesy of Dayton Audio)

Processing'). Consider the signal: y(t,w) = sin(w*(t-T)) = sin(w*t + ((-1)*w*T)), where T is the delay through the filter. Clearly the second term in the sin argument is the phase (theta). If the time delay is constant, theta will be a decreasing function of omega. However, if the time delay T was not constant over frequency, then theta may actually increase instead of decrease, and thus –d theta(omega)/d omega would be negative. Just think of the slope of the function theta(omega) and you will see how the group delay can be negative without necessarily meaning the future is being predicted!"

The OmniMicV2 software measures the phase and can create a text file with each line having frequency, amplitude response and phase. How then did I get the group delay? I wrote a C language program to read the text file and computed the group delay from the frequency and phase using the equations given by Christopher J. Struck in his paper, "Group Delay" (see Resources). First, the phase is "unwrapped" using Struck's equations. Unwrapping is simply making the phase monotonic. The left loudspeaker wrapped and unwrapped phases are shown in **Figure 8**. The curves for the other loudspeaker are similar.

Then the group delay is calculated using his equation. The derivatives are approximated using a three-point slope from the frequency and phase data. Thus, the group delay can be calculated from simple differences.

I have included a copy of the program source code and the library files needed to compile it in psi2dlay.zip in the Supplementary Material. Yes, it's another DOS program using the PowerC compiler but it runs fine in Windows 2000, XP, and 7. For details, please see the readme.txt file in psi2dlay.zip and my "Signal Dynamics and Loudness" article (*audioXpress*, May 2015).

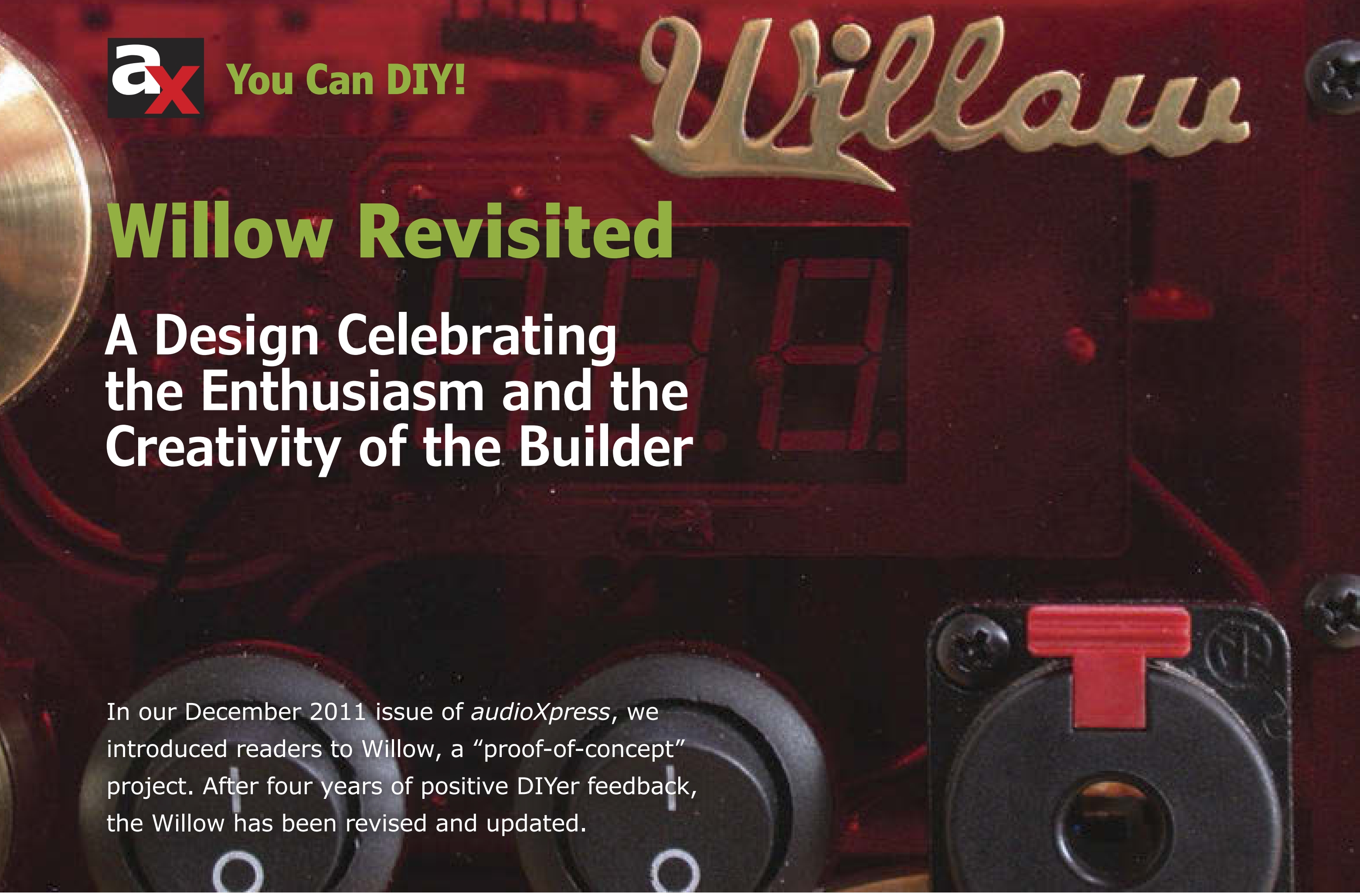
Final Thoughts

I could have made the amplitude response curves flatter by time windowing out the low-frequency reflections. This would have looked "prettier" but it would have been unrealistic because the reflections are there in the sound you or I would hear. But, I am especially pleased with the low group delay distortion. This may be a big factor in why these loudspeakers sound so good. Overall, I think it was a worthwhile project.

In case you are curious, I built the speaker "stools" shown in **Photo 1**. The tops are 2" × 12" red oak and the legs are 12" lengths of 1" diameter aluminum rod drilled and tapped on one end for the five 16" × 2" stainless steel flat head machine screws to attach the top. The rods are drilled and tapped on the other end for screw-in carpet spikes. I gave the rods a satin finish in the lathe with 150-grit sandpaper and applied two coats of clear, satin polyurethane varnish to the tops. The legs were set a 0.5" into the tops with a 1" Forstner drill bit.

About the Author

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at the 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.



You Can DIY!

Willow Revisited

A Design Celebrating the Enthusiasm and the Creativity of the Builder

In our December 2011 issue of *audioXpress*, we introduced readers to Willow, a "proof-of-concept" project. After four years of positive DIYer feedback, the Willow has been revised and updated.

By
Robert Nance Dee
(United States)

It's funny how things go, Willow was a project that I never really expected to take off. What I had hoped for back then was that the DIY community would pick up on the design and build it into something that would fit specific requirements and tastes. From the correspondence I've received, many of you have done just that and much more than I ever expected. I can't relay how happy that makes me!

I was working on a different design when I got an email from a DIYer filled with enthusiasm and lots of questions about Willow so I decided to change tracks and revisit and update the design, especially now that the HA-5002 buffer is obsolete.

I wanted to keep the main amplifier circuit profile, but I also wanted to improve the power supply and the input to the buffer circuit. I especially wanted to make the amplifier as quiet as possible and have it be flexible so that those who want to avoid microcontrollers could still experiment with the design. After all, this is an amplifier for the DIY community and I enjoy seeing what can be done with my designs (see **Photo 1**).

The Buffer

The first issue I addressed in the revised design was the buffer. For this version, I chose the LM49600, which didn't have the open output problems of the HA-5002 that some builders experienced.

To keep the noise down, I redesigned the input bias circuit. First I used all low-noise regulators in the buffer power rails. Second, I used low-noise surface-mounted capacitors and resistors here, and throughout the design's audio section, to keep the signal path short and noise free. I also updated the buffer input using the negative 15 V rail to adjust the small final bias voltage, thus avoiding the variable carbon trim pots from the original design.

The change worked well. Although there was a slight difference between the two buffer biases, I could balance the offset between them to the point of only a few millivolts bias. You can individually bias each buffer by experimenting with the bias resistors and omit the negative 15 V variable resistor, but the benefit of the variable rail is that it's an easy fix if you check the bias voltages after a period of break in and need to tweak things.

Photo 1: The revised Willow offers a unique cube design.

I liked the sound of the original JFETs so much that I kept that portion of the circuit true to the original design. What I tried to do was give you a basic layout that they could work from, but if you build this design as it stands, I think you'll find it is quite good.

The Power Supply

The power supply for the Willow's revised design is completely new (see **Figure 1**). I used all low noise regulators anywhere they were involved in the signal path. The other thing I did was take special care in balancing the input transformer. In past designs, I have simply showed different AC inputs leaving the choice up to the builder. The problem with that approach is that noise, especially hum, is tough to track down and keep out of circuits.

If you hear a hum that varies between left and right channels when you shut off your amplifier with your headphones on, then you very possibly have imbalances between the rails and the DC outputs.

Balancing the power supply, as I did here, eliminates those problems and keeps hum to a minimum. I laid out the circuit boards using a single point connecting the power supply and the auxiliary circuits (input relays, etc.) to the signal portion of the preamp itself.

The signal portion of the main board shown in **Photo 2** is basically contained in a small area of about 2″ × 3″ (50 mm to 75 mm). Very few

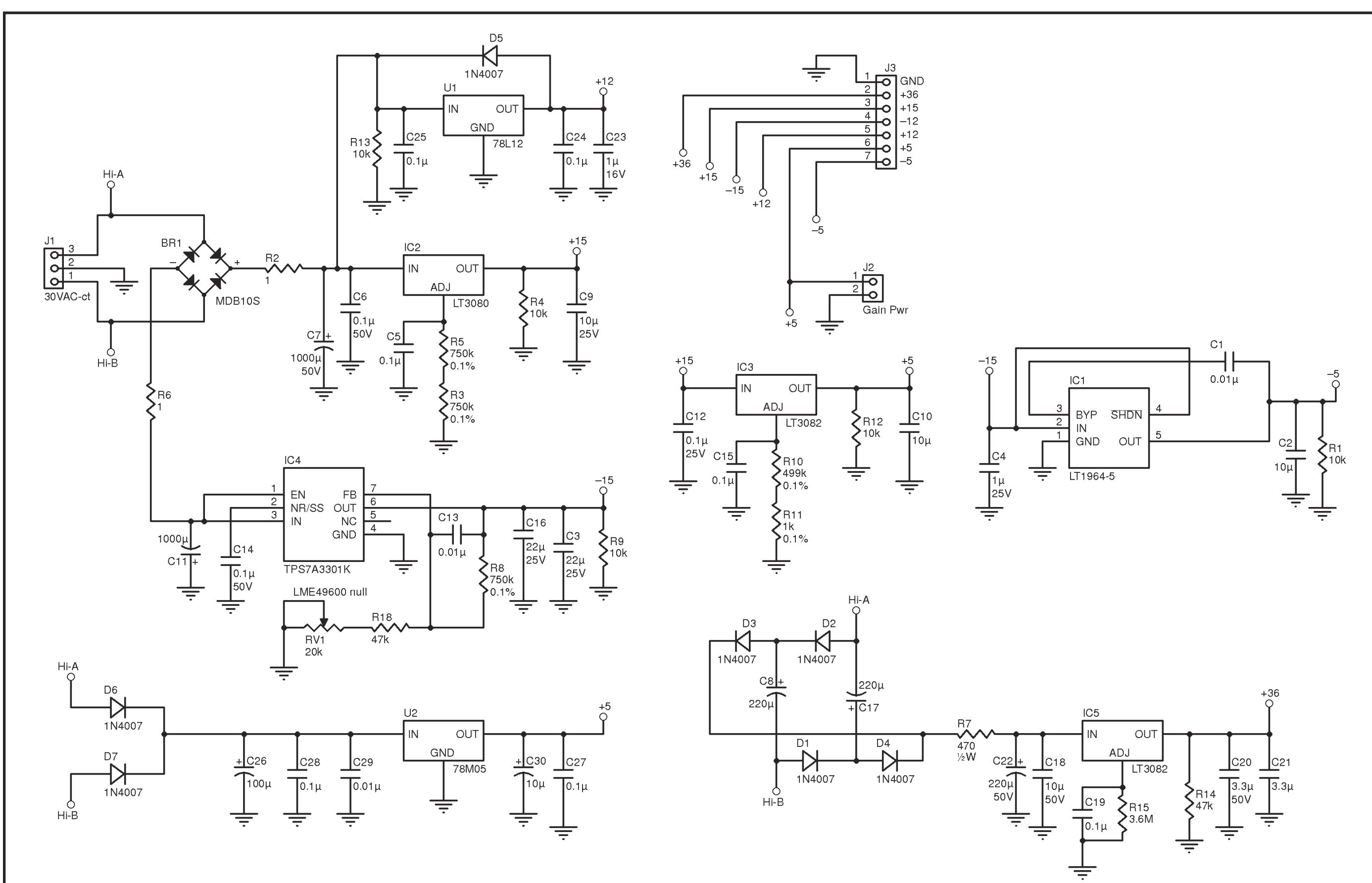


Figure 1: The revised Willow has a completely new power supply.

amplifiers have such short paths. I carefully chose each component for its audio excellence.

Low noise and compactness make a final design that rivals many high-end commercial units in a package the DIYer can build. **Photo 3** shows the 250 kHz square wave output. (The upper trace is the amplifier. The lower trace is the source.)

The Silent Microcontroller

I took special care to shut off the gain microcontroller. It goes to sleep after the level is set and displayed. The AD5282 digital 50 kΩ pot has only the I²C leads (SCL, SDA) on the amplifier board. The microcontroller and the three digit LED display are on their own board mounted to the front panel (see **Figure 2**). The display comes on for a few seconds to show which one of the possible 256 levels has been selected (see **Photo 4**). Then, it shuts off so it emits the lowest noise possible.

The AD5282 is on the amp board, in close proximity to the capacitors and the JFETS in keeping with short signal paths. All the control electronics are on the front panel where they don't lengthen the signal path. All this careful attention to circuit layout, along with component choices, pays off in a dead-quiet amplifier and minimum signal path lengths. Even the headphones have a relay that shuts off the leads to them by a front panel switch when not in use.

Looking at the PCB top trace, the red copper pour is the signal ground and the bottom blue is power supply ground (**see Figure 3**). A jumper from J3 to J4 on the amplifier board connects the two at this single location only. The red top copper show just how small the actual amplifier is with only the leads from the inputs outside the pour itself. The inputs have their own copper pour (bottom blue on the top edge of the board). The power board bottom copper (**see Figure 4**) shows how I ran the low noise ground plane.

Figure 5 shows the revised schematic for the updated Willow design. Portions of the circuit isolated in red dotted line boxes are not in the circuit. I made a small board for the LED circuit and mounted it to the top of the acrylic case, each LED is associated with an input. I used an SMLW56RGB1W1 tricolor LED for this application emitting a red, blue, and green light. The fourth input is a combination of the three (white light). The diode area in red is for those who want to use the buffers in other designs, tubes for example. It holds the input to the buffers at a safe level, we do not need it in this circuit.

The AD5282 is quite a chip, it contains all low-noise metal film resistors and is so superior to

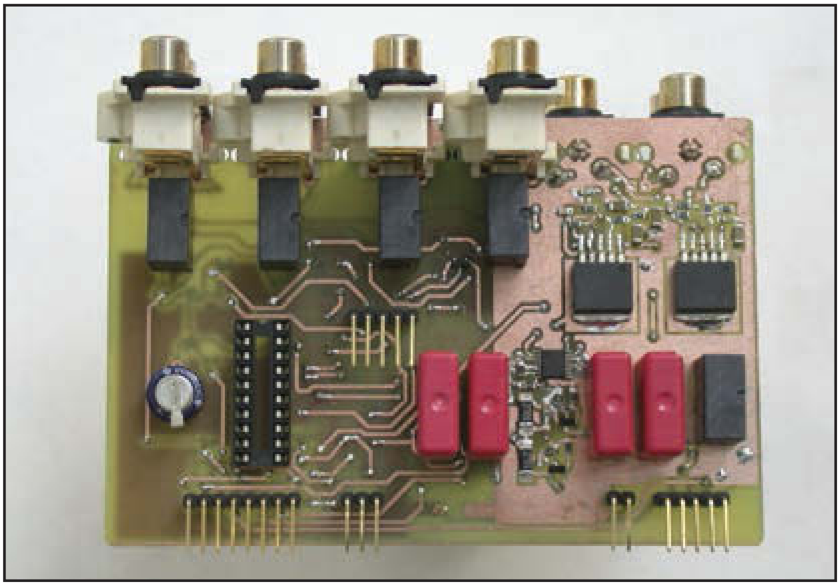


Photo 2: The signal portion of the main board is basically contained in a very small area.

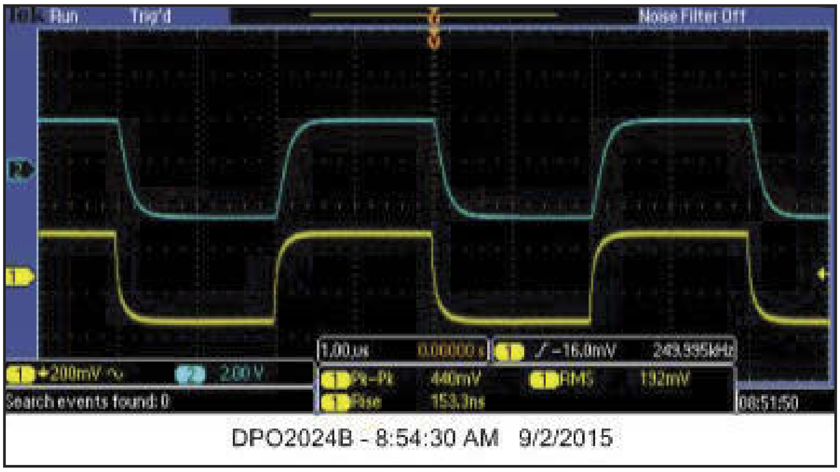


Photo 3: This is the 250 kHz square wave output for our revised Willow.

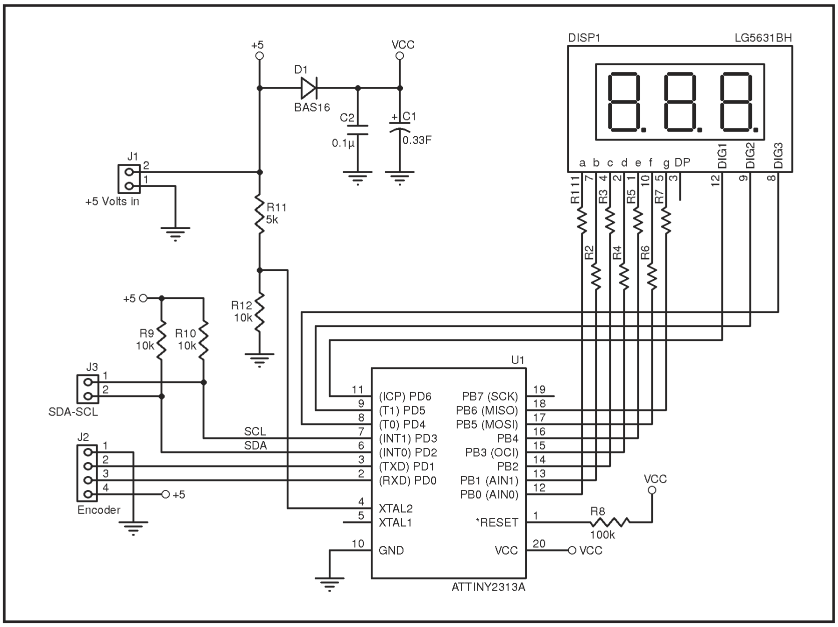


Figure 2: This is the gain display for the Willow's updated design. A new revised schematic is also available in the Supplementary Material.

Photo 4: The display comes on for a few seconds to show which one of the possible 256 levels has been selected.

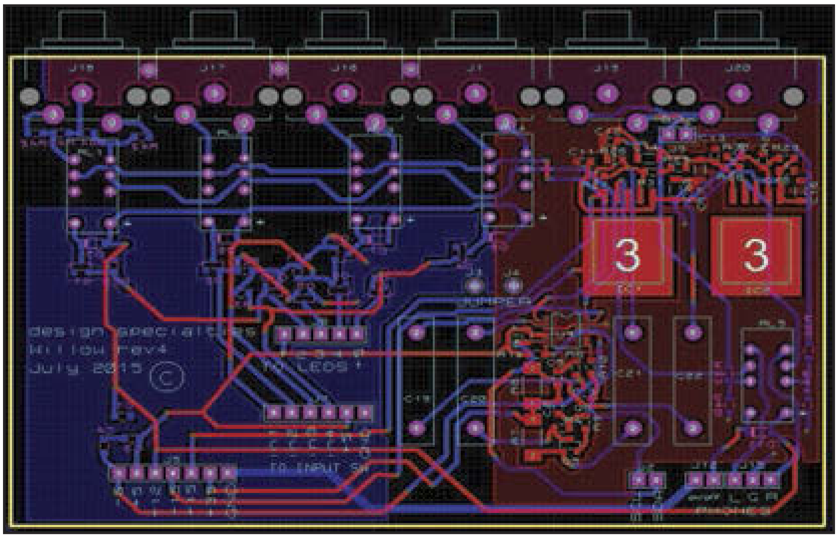


Figure 3: Looking at the PCB top trace, the red copper pour is the signal ground and the bottom blue is power supply ground.

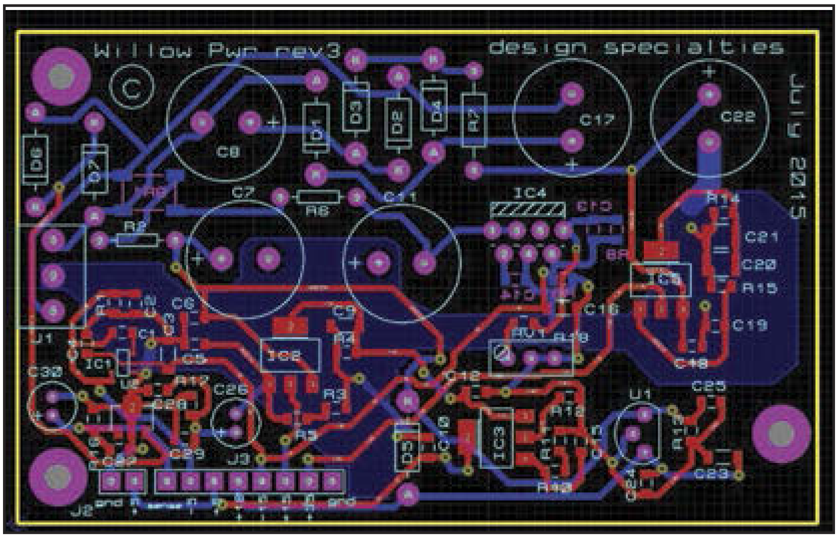


Figure 4: The power board bottom copper shows how I ran the low noise ground plane.

carbon pots that I do think it makes a difference. Using it keeps the signal path tight and completely carbon free.

You certainly can use switched pots with ganged resistors here. I find the better ones overly expensive, but this is your amplifier and I understand how you may prefer different components. The problem is that ganged and carbon pots either sit off the main amplifier board or take up a lot of room on it, which introduces noise along with contacts that degrade over time.

Chips such as the PGA2311 are built around op-amps. I avoid those as they influence my design and ability to contour circuits to my specific liking. Also the PGA2311 is not in the same league as the AD5282 series of chips. The AD pots are that much better.

Project Files

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

Resources

Design Specialties, www.dsgnspec.com.

R. Nance Dee, "The Willow Preamp: A High Slew Rate JFET Amplifier," *audioXpress*, December 2011. Available online http://audioxpress.com/article/The-Willow-Pre-amp-A-high-slew-rate-JFET-amplifier.html.

Sources

AD5282 Potentiometer
Analog Devices, Inc. | www.analog.com

SMLW56RGB1W1 Tricolor LEDs
ROHM Semiconductor | www.rohm.com

LM49600 Audio buffer
Texas Instruments, Inc. | www.ti.com

About the Author

Robert Nance Dee is a retired electronics engineer. He received his BS from the State University of NY, where he was nominated for the Chancellor's Award for Student Excellence. He has worked on large frame military computers and has several medical instrument patents. He enjoys electronics, mechanics, clock and watch making and precision machining. He and his wife Nancy live in the Western Catskill Mountains of NY where he is presently restoring a massive E. Howard Tower Clock in the Delhi, NY village square.

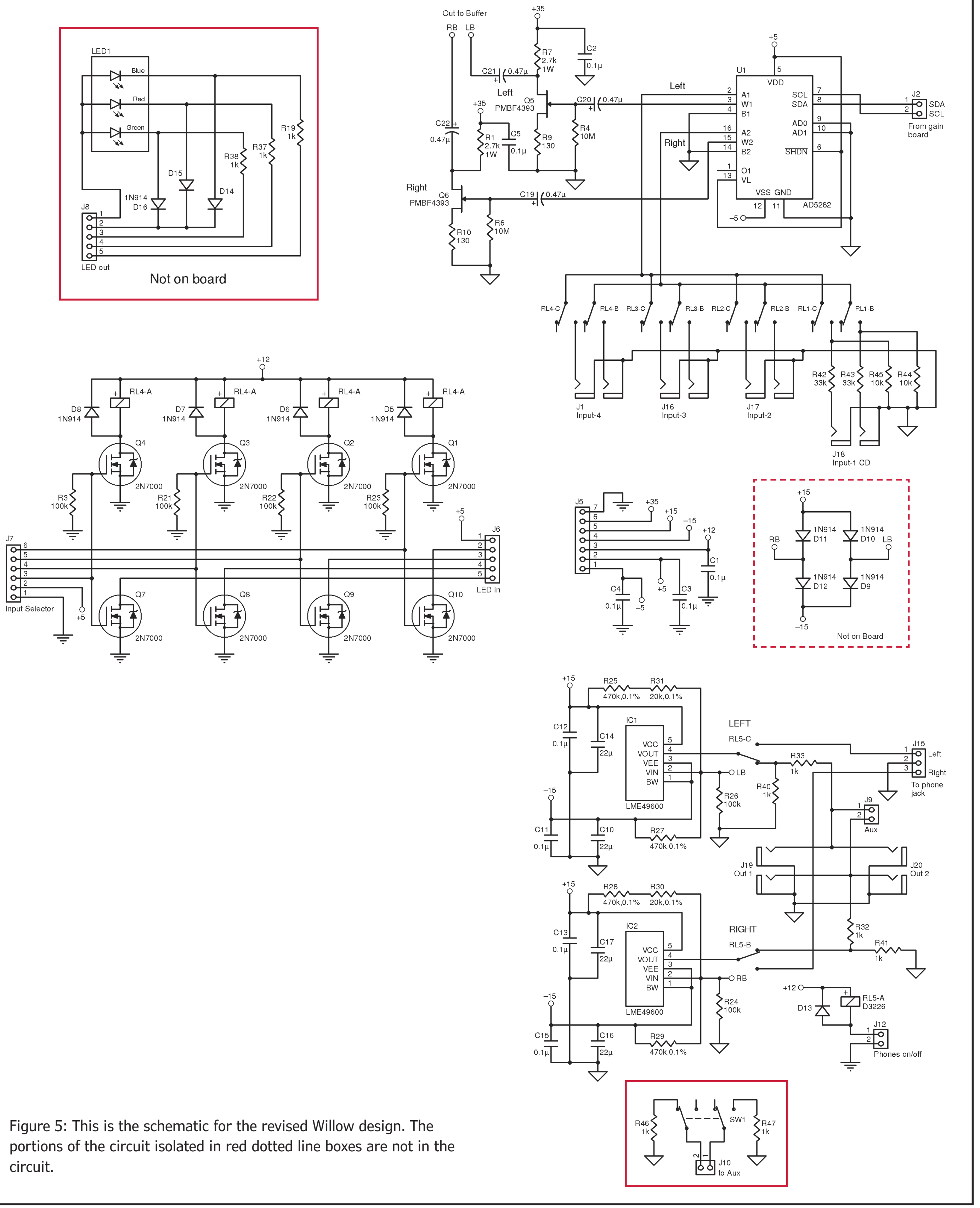


Figure 5: This is the schematic for the revised Willow design. The portions of the circuit isolated in red dotted line boxes are not in the circuit.

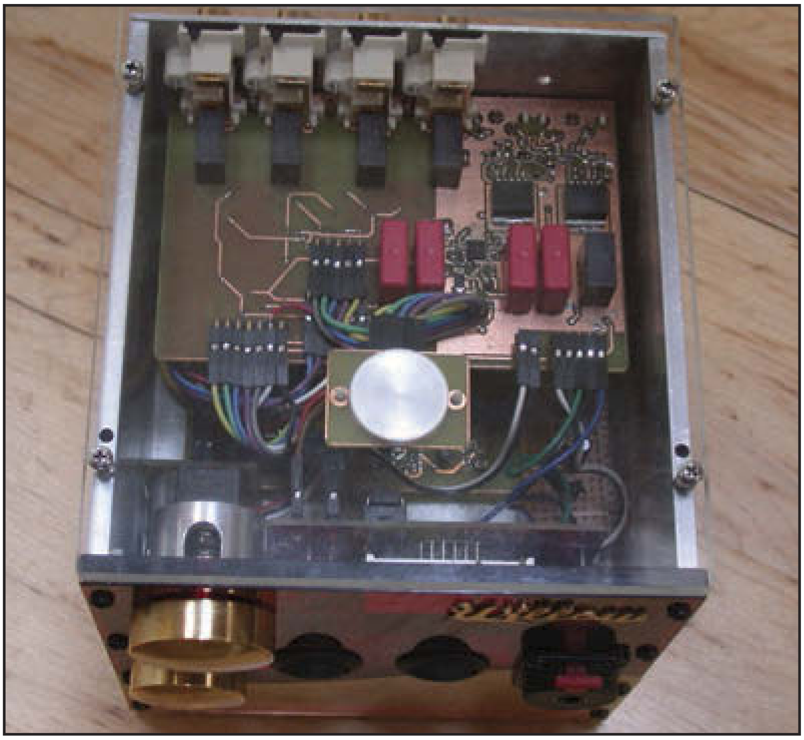


Photo 5: Space is efficiently utilized in this tight cube design.

The Case

I also built a new case for this preamplifier. The most efficient utilization of space is a cube (see **Photo 4** and **Photo 5**). This one has added benefits. The power transformer is located in the bottom of the case. It is separated from the power board (next tier up) by a metal shield above it and the metal surrounding it.

The front panel incorporates a metal lower plate of brass and copper and an acrylic upper plate in red to enhance the display's three digits and still surround the transformer with a metal shield. The main board occupies the highest location in the case, completely isolating it from the transformer. There is a ground from the main board at the outputs to the case itself. The three-legged mains line also has its ground leg going to the case. The power supply board has no grounds to the case and a single-point ground to the main board, again, at pins "J3" and "J4" of the main board.

The specs for this amplifier should include a gain of 20.7 dB with the distortion and noise at less than 0.15%. This means the amplifier should be noise free. Mine is dead quiet with the inputs shorted—no hissing or low frequency hums. In fact, I can't tell whether the amplifier is on or off using headphones.

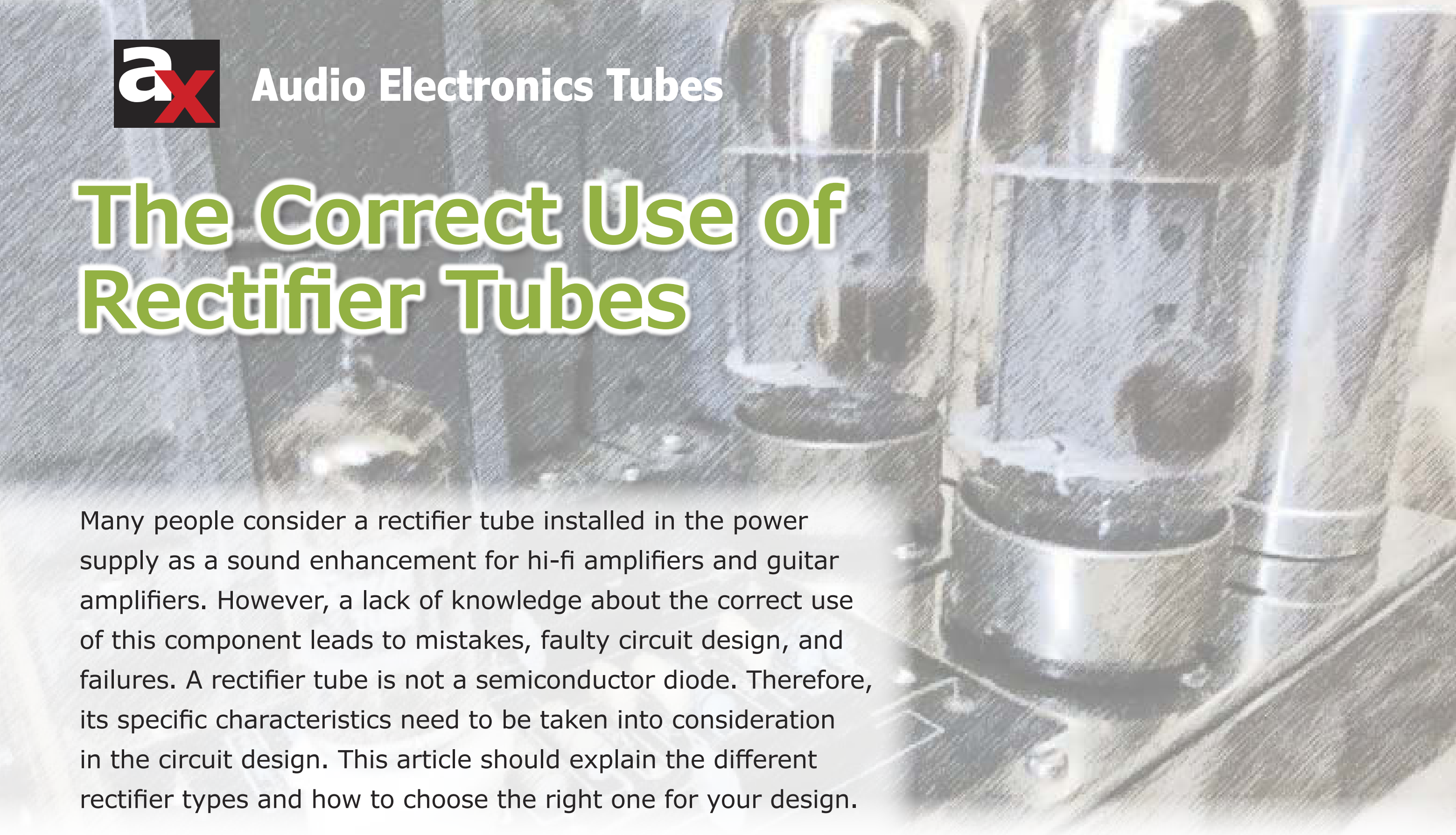
The bills of material (BOMs) are included in the Supplementary Material on the *audioXpress* website, which will enable you to build Willow using the same components used in this design. But, I encourage you to experiment on your own.

Redesign Final Remarks

Any design has trade-offs but what is gained here I find worth it. There's no reason you can't incorporate the updates with the original design if you so desire, nothing is etched in stone.

I won't go into verbose nonsense on the sound, I'll let the builders and listeners decide for themselves if these changes make a better sounding amplifier. I will say that if you like tube amps give this a try. It has a smoothness but with great detail. Do the changes make a better overall amplifier? Yes, I believe very much so. I could give you a lot of specs but what does that really mean and how does it equate to sound, musicality, or the sheer joy of building something that rivals many amplifiers regardless of price?

Author's Note: I have updated the gain display board and added a new bill of materials (BOM). They can be found in the Supplementary Material section of the audioXpress *website and on my website.*



The Correct Use of Rectifier Tubes

Many people consider a rectifier tube installed in the power supply as a sound enhancement for hi-fi amplifiers and guitar amplifiers. However, a lack of knowledge about the correct use of this component leads to mistakes, faulty circuit design, and failures. A rectifier tube is not a semiconductor diode. Therefore, its specific characteristics need to be taken into consideration in the circuit design. This article should explain the different rectifier types and how to choose the right one for your design.

By
Gerhard Haas
(Germany)
Original article published in German in *Elektor Roehren Sonderheft 1*, December 2005.

Before there were so-called dry rectifiers, the discussion about what type of rectifier to use was irrelevant. Basically, the only choice was vacuum diodes. For demodulators and small signals, cuprous oxide rectifiers were used.

In this case, the rectifier tubes are made with a monovalent copper oxide with the chemical formula Cu_2O. This oxide forms crystals. If a metal contact is put on the crystal, a semiconductor effect is generated between the metal and the crystal that can be used for rectification purposes. With about 0.2 V, the threshold voltage is very low. However, the current as well as voltage capacity are not particularly high. So it was this component would not typically be used as a rectifier for mains operation. It was mostly used in measuring devices or as a high-frequency rectifier tube.

Dry Rectifiers

Originally, the term dry rectifier meant a rectifier on a semiconductor basis. Previously, selenium, a relatively rare element, was the material of choice. A selenium cell can withstand a reverse voltage of about 30 V and features a 0.4 V threshold voltage. For higher voltages, more single cells need to be connected in series. However, the surge current stability of selenium rectifiers is not very high because the power load is only 50 mA/cm^2. If damaged, several rectifier cells could blow, creating a garlic-like odor. The low-threshold voltages were a big advantage at low currents and voltages of up to 30 V because the losses remained minimal. On the other hand, these threshold voltages add up if higher voltages need to be rectified. For safe operation with an AC voltage of 250 V, you need up to nine cells, depending on the design, which results in a total threshold voltage of 3.6 V.

With bridge rectifiers, where two diodes are used, you would have a two-fold voltage drop. However, a silicon rectifier has a 0.6 V threshold voltage per diode. Depending on the current, the voltage drop can rise to just over 1 V, which is also true with currents of more than 1 A. The silicon rectifier has a steep characteristic curve so that even with high currents the voltage drop increases only slightly.

The selenium rectifier has a relatively flat characteristic curve showing a steady increase of the voltage drop with increasing current. This effect is used in simple battery chargers for car batteries.

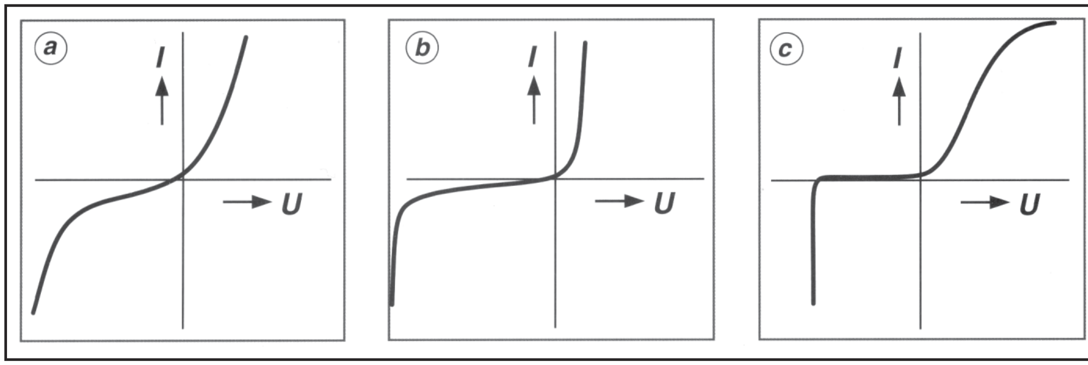


Figure 1: These are the typical forward curves of the selenium rectifier (a), the silicon rectifier (b), and the tube rectifier.

With an empty battery, the current is high and so is the voltage drop at the rectifier. This results in an automatic current limitation. If the battery voltage rises, the current decreases and so does the voltage drop at the rectifier. In this way, the differential voltage between the transformer winding and the battery voltage increases once again, which leads to an increase of the charging current. By making full use selenium's physical characteristics, you can create a simple charging current control without any series resistor or electronic control circuit. Moreover, the copper resistance of the transformer winding is taken into account when it comes to current limiting. In the past, this principle has proven itself millions of times.

Wet Rectifiers

In addition to dry rectifiers, there are also wet rectifiers. Two electrodes immersed in an electrolyte ensure that the current can only flow in one direction. However, a wet rectifier has the disadvantage of gassing and the electrolyte is gradually consumed. The wet rectifier is basically nothing more than an inversed electrolytic capacitor. An electrolytic capacitor tolerates about 1 V of reverse voltage before becoming more and more conductive. When applying the correct polarity, it blocks the current.

In principle, one could also make a rectifier with an electrolytic capacitor, but that does not make sense as it would soon burst because of gassing. In some craft books from the 1960s and earlier, there were building instructions for wet rectifiers because tube and dry rectifiers were relatively expensive and only available in a limited selection. Today, building a wet rectifier is an interesting experiment, however, it has little technical value.

With every rectifier, the maximum permissible forward current (conducting state current) is specified. Likewise, the reverse current is listed in the datasheets. The higher the ratio between forward and reverse current, the better the rectifier. The lower the junction capacitance and the faster the junction can be discharged, the lower the peak negative phase sequence current when closing the rectifier. Herein lies the superiority of silicon-based rectifiers.

Rectifier Tube Characteristics

Rectifier tubes have their very own characteristics. **Figure 1** shows the typical forward curves of the respective rectifier types. The selenium rectifier's characteristic curve, shown in **Figure 1a**, is constant throughout the forward and in inverse directions. The silicon rectifier's outlet curve, shown in **Figure 1b**, is very steep, which means you can expect a low voltage drop even when there is a high current. The characteristic curve is very flat in the stop band. Once the maximum reverse voltage is exceeded, the current rises and the rectifier will be destroyed.

The rectifier tube's characteristic curve, shown in **Figure 1c**, slowly rises in the pass-band. If the voltage at the rectifier exceeds 0 V, the current starts to flow. At a certain point, the cathode reaches the saturation point so the current hardly increases. In this case, there is a danger of destroying the emitting cathode layer. Once the maximum reverse voltage is exceeded, there will be flashovers inside the tube. The ever-present residual gases inside the tube ionize and become conductive. This can result in bluish sparks inside the tube, possibly leading to its destruction.

Selenium and silicon rectifiers require a minimum voltage for the diode to open at all. The outlet characteristics of the widely used two-way rectifier diode EZ81 (or the 6CA4), shown in **Figure 2**, indicates that current can flow through the tube diode from the zero point.

As nice as it is at 150 mA, the voltage drop rises to an impressive 20 V, which needs to be considered when dimensioning a power supply equipped with rectifier tubes. **Figure 2** also shows currents of more than approximately 75 mA, which are drawn as dashed lines. Currents higher than this point must flow in pulsated operation because of the half-waves during rectification of the line voltage.

Figure 3 shows curves that represent the fundamental ratios at the rectifier and the charging capacitor. In contrast to silicon rectifiers, rectifier tubes tolerate current spikes to a limited extent. For example, the typical and inexpensive 1N4007 rectifier diode can sustain a maximum reverse voltage of 1,000 V and a constant current of 1 A with a single surge current of up to 50 A for a 60 Hz half-wave without any component failures. However, this would be absolutely deadly for a rectifier tube. The momentary surge current stability of silicon rectifiers means no major measures for limiting the current need to be taken.

Normally, the transformer winding's copper resistance is sufficient. At the moment of switching on, the connected charging capacitors are empty. So, at the first sine half-wave, a considerable amount of current flows from the transformer and the capacitors will be largely loaded. The current is lower with the second half-wave. If a silicon diode survives the first surge, the power load data for continuous operation is applicable because it is significantly lower than the initial current.

Rectifier tubes are different. At the moment it switches on, the cathode is cold and no current flows. Once the cathode temperature rises, current slowly starts to flow. This is safe for the charging capacitors, but today it no longer plays an important role.

Normally, the loading capacitors are electrolytic capacitors that have experienced more than 100 years of evolutions. They are used in switching power supplies, computers, gated power supplies in TV sets, screens, welding machines, transformerless charging devices, electronic transformers for halogen lighting, CNC machines, and much more, where they are exposed to high surge currents and voltages. Modern electrolytic capacitors can withstand the surges for years, provided that the limiting data of the manufacturers is observed. Therefore, rectifier tubes can be used to conserve electrolytic capacitors.

Dimensioning of a Power Supply

During the dimensioning of a power supply, some rules apply to rectifier tubes but not to silicon rectifiers. For instance, the missing surge current strength of the tube must be taken into consideration as does the voltage drop per tube diode when dimensioning the transformer. The tube manufacturers prescribe the maximum capacitance of the charging capacitor so that the recharging current spikes per sine half-wave are not too high. Whenever the rectifier opens, a charging current

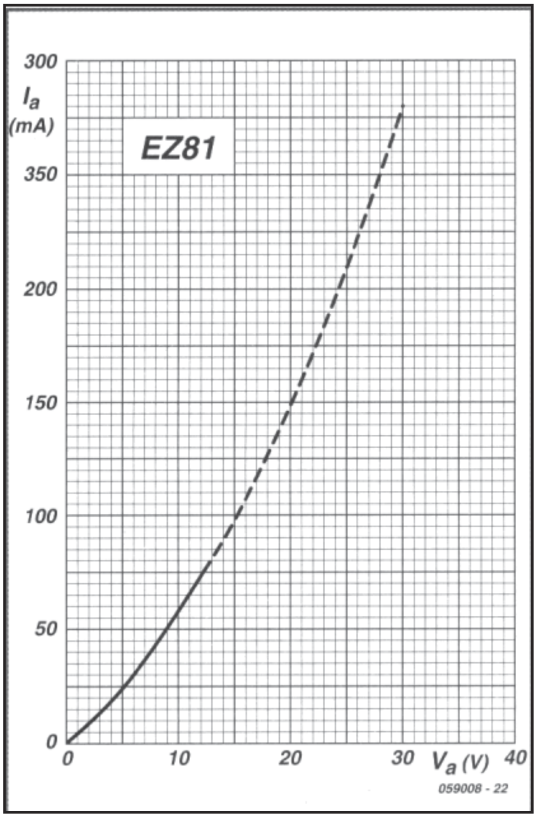


Figure 2: The two-way rectifier diode EZ81 indicates that current can flow through the tube diode from the zero point.

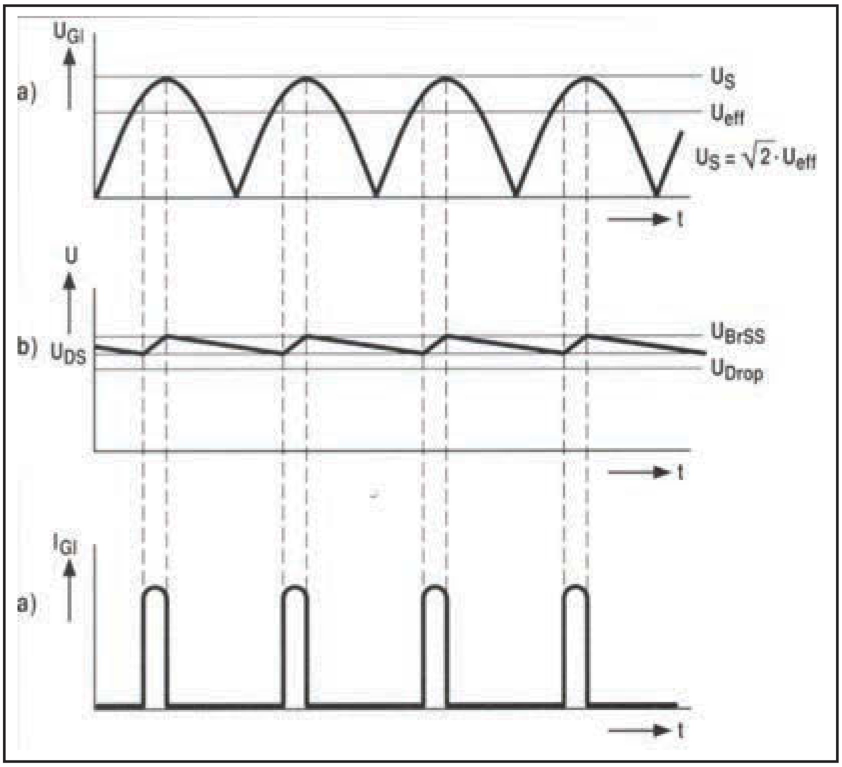


Figure 3: These curves represent the fundamental ratios at the rectifier and the charging capacitor.

spike is generated.

Figure 4 shows the fundamental circuit. The maximum current is limited by the cathode's emission capability, which is expressed in mA/cm^2 and is typically 75 mA/cm^2. If the allowable limits are repeatedly exceeded, the cathode layer will be irretrievably destroyed. Therefore, the charging current spike needs to be limited by means of resistors. The datasheets contain specifications for this. In innumerable old circuits, the current limiting is achieved via the copper resistance of the transformer winding, which is also mentioned in the datasheets.

About the Author

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

Figure 4, taken from a historical tube datasheet for the EZ81 (or the 6CA4), shows the total resistance necessary for the copper resistance of the transformer winding and, in case it is too low, of additional resistances in the anode line. In this case, the copper resistance R_P of the primary winding and of the secondary windings (R_S) as well as the turns ratio are taken into consideration.

When a transformer is calculated and built according to up-to-date methods, it is done so that the losses inside the transformer are as low as possible. Therefore, the copper resistance of the winding becomes relatively low for the anode voltage.

Older transformer designs featured a much thinner wire, which was intentionally chosen because of the rectifier tubes, guaranteeing the current limiting due to the copper resistance. However, two effects will occur. The warming up of the transformer winding will be higher and it will heat the entire transformer more than is actually necessary. The other effect is that the copper resistance will increase with the temperature.

The copper resistance increases per degree Celsius by the factor 0.0039/°C. This may seem low, but let's look at an example. At first, the DC resistance of the copper winding has a room temperature of 20°C. When installed, the ambient temperature inside the device is 40°C. During operation, the winding itself warms up by 40°C because of iron and copper losses. This results in a temperature increase of the transformer winding by 60°C. Assuming that the entire transformer winding has a cold resistance of R_C = 100 Ω with 20° C. The increase in temperature is 60° C, this leads to a hot resistance R_W of:

$$R_W = R_C + 0.0039 \times \Delta t \cdot R_C = 100\ \Omega + 0.0039/°C \times 60°C \times 100\ \Omega = 123.4\ \Omega$$

Therefore, the resistance of the copper winding has increased by at least 23.4% during operation. If this is not sufficiently taken into account when designing the power supply, depending on the load current, a significant operating voltage difference between a cold and hot transformer must be considered.

Non-Constant Load

The next problem is the non-constant load. Preamplifiers, as well as single-ended type-A power amplifiers don't show any significant differences in current consumption between idle and full load. However, it will be more critical with push-pull AB amplifiers. In this case, the rectifier's long and bent characteristic curve and the transformer's heating have a major effect. The more power is pulled (i.e., at high modulation), the higher the voltage drop will be at the rectifier diode and the transformer winding.

The effect of the voltage drop at the rectifier tube will be immediately noticeable, however, the drop at the transformer winding will be delayed. If the amplifier is warm, the operating voltage will be slightly lower, which will have an effect when using power triodes. In this case, the fluctuation of the anode voltage is much higher than with pentodes. Furthermore, it must be taken into account that the emission capability of the cathodes reaches the maximum after 10 to 20 minutes, depending on the tube size (e.g., power tube, preamplifier tube). Then, a lot of theorizing can be heard about the warm sound of the amplifier without people being aware of what influences this effect.

Another effect of the smooth characteristic curve of the rectifier tube together with the current-limiting resistors is that the operating voltage varies depending on the modulation, which is the case with push-pull AB amplifiers. Therefore, the internal resistance of a power supply operated with rectifier tubes is relatively high. This leads to a shift of the operating points of the amplifier stages, in particular, those of the power amplifier. This is desirable for guitar amplifiers, where some sort of compression effect will arise. The result is that when the string is plucked for the first time, the power amplifier reaches its limit range and the operating voltage drops. When the input drops, the power amplifier slowly comes out of its limitation and the operating voltage increases again. In this way, a long sustain is achieved (i.e., the guitar tone will last for a long time with high volume instead of quickly decaying). This effect has been used for decades in rock music, however, it can also be generated in another way.

Figure 5 shows a typical rectifier tube circuit. Normally, the tube is used as two-way rectifier. The resistances R_S complement the copper resistance of the transformer winding to achieve the value necessary for safe operation. Since it is not possible to include the datasheets of all tubes, you should get the necessary data of the tube you plan to use before mistakenly pairing a dimensioned transformer with the wrong tube wiring. One of the most common mistakes is when the transformer voltage and the transformer output are incorrectly calculated for the use of a rectifier tube. For a two-way rectifier, a transformer winding with twice the voltage is used that features a center tap connected to ground. Only one partial winding of the mains transformer is used per half-wave (see **Figure 4**). Here, transformer builders talk about 50% Duty Cycle (DC) of the winding parts.

Let's calculate an example using the EZ81. The DC voltage at the charging capacitor is supposed to be 300 V with a power consumption of 100 mA. The transformer in a warm operating condition should have 61.7 Ω for winding each half of the winding. For the charging capacitor, we would use the standard value of 47 µF, since for the EZ81 a maximum value of 50 µF is admissible. Now, various aspects need to be considered to determine the necessary transformer voltage and the transformer output for this winding. The DC voltage and the actual AC voltage are linked by the factor $\sqrt{2}$. In conclusion, if a DC voltage of 300 V is required, the alternating voltage of the transformer must be lower by that factor, because the rectifier and the charging capacitor together form a peak voltage rectifier. Therefore, the transformer voltage should be:

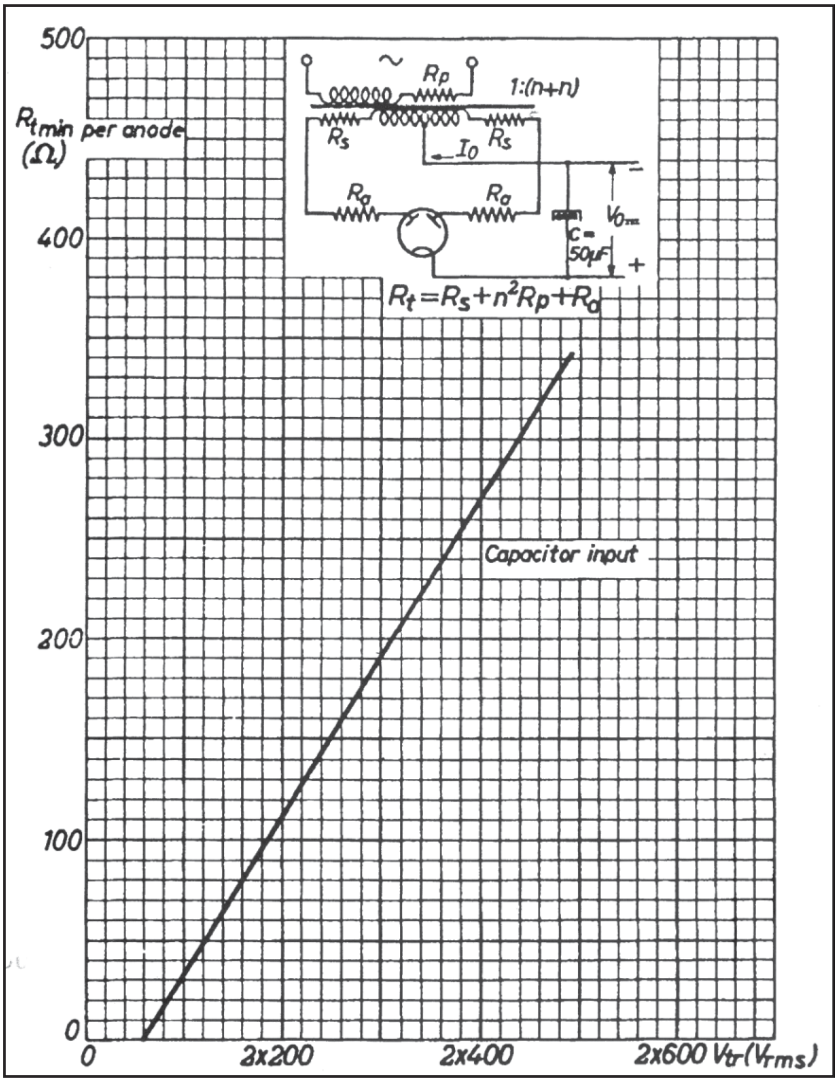


Figure 4: Taken from a historical tube datasheet for the EZ81 (or 6CA4), this shows the total resistance necessary for the copper resistance of the transformer winding and, in case it is too low, of additional resistances in the anode line.

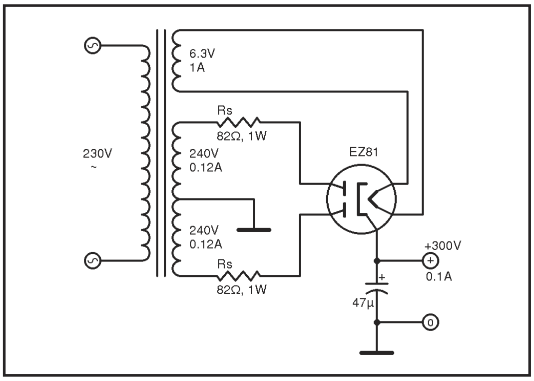


Figure 5: This is a typical rectifier tube circuit.

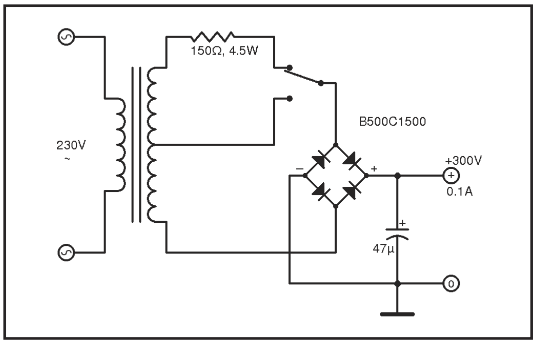


Figure 6: This is a a circuit that simulates the behavior of a rectifier tube.

$$U_{TR} = U_{DC} : \sqrt{2} = 300\text{V} : \sqrt{2} = 212.2\text{V}$$

The load curve of the rectifier tube EZ81 shows that with 100 mA, a voltage drop of 15 V occurs. Therefore, a transformer voltage 212.2 V + 15 V = 227.2 V would be necessary. Now, the voltage drop at the resistors R_S need to be taken into consideration, too. According to the datasheet, the total resistance for each diode and with a transformer voltage of 250 V should be 150 Ω. Since the entire transformer winding has been calculated with 123.4 Ω at operation temperature, the result for the half winding is 61.7 Ω. Therefore, 88.3 Ω still need to be connected upstream. The nearest standard value with 82 Ω is used, which is tolerable. With 100 mA, this results in a voltage drop of 8.2 at the resistor. In this case, it is assumed that the transformer builder has calculated the winding so that the required voltage is present at the load specified and at operating temperature, which is standard practice.

During the calculation of the transformer winding, the copper resistance at operating temperature is also taken into account so that you can calculate with the full voltage. In purely mathematical terms, each transformer winding must have 212.2 V + 15 V + 8.2 V = 235.4 V. The transformer voltage is rounded up to 240 V, because the tubes also have tolerances and by using series resistors the voltage can easily be divided. However, a subsequent increase in not possible. Since an arithmetic dissipation of 0.41 W is generated at the resistor, it is better to use one with a resilience of at least 1 W. You need to take into account that they do not come up 8.2 V × 0.1 A = 0.82 W but only half the amount, since only one half-wave flows through the resistor and the latter is loaded for only half the duration. To be safe, you should use resistors with a resilience of 2 W. You also have to consider the dielectric strength of the resistors. For safety reasons, the selected types must feature a dielectric strength of at least 500 V.

In principle, the rectification increases the transformer's effective voltage by the factor $\sqrt{2}$. Since there is no perpetual mobile, the transformer's AC current must theoretically be increased by the same factor. In fact, it is sufficient to calculate using a factor from 1.2 to 1.3 since the electricity is not drawn in full sine half-waves but only for as long as the rectifier diode is open (see **Figure 3**).

A mains transformer can always provide more than the nominal current for a short time without being damaged. Furthermore, the electrical power is not drawn via the full sine half-wave so that the duty cycle of the winding is not 100%.This means the transformer must feature the data of 2 × 240 V and a least 0.12 A. This results in a minimum output power of P = 2 × 240 V × 0.12 A = 57.6 W. However, since only 50% of transformer windings are switched on, you could divide the 57.6 W by 2 which results in 28.8 W.

For safety use the factor 0.7 (i.e., 57.6 W × 0.7 = 40.3 W). So, to build the transformer you must use the necessary data with the minimum required transformer output: 2 × 240 V/0.12 A, 50% DC because of the two-way rectifier and 40.3 VA.

With an AC calculation, the term volt-amperes (VA) is used instead of watts (W), which is basically the same. The wire of the transformer winding is designed for the maximum current so that the voltage drop at the winding is not too high. However, the core size may be smaller because of the duty cycle of 50%.

One Particularity

A particularity during the operation of rectifier tubes should not go unmentioned. It is the filament. A rectifier tube is heated by a special coil and are available as directly heated types where the cathode and the filament are the same. Then, there are types that are indirectly heated but have the cathode and the filament connected inside the tube (e.g., the GZ34 or the 5AR4).

The third type of tubes have their cathodes and filaments separated as with the EZ81. Therefore, the rectifier tube's filament must never be connected to other tube filaments. This would destroy directly heated amplifier tubes. With indirectly heated ones, the dielectric strength between the filament and the cathode is limited.

In most cases, the dead end is reached at 200 V or far below. Therefore, the rectifier tube must have its own filament coil at the transformer. This filament coil needs to be particularly insulated from all other windings! If there is an internal voltage flow, the transformer can be damaged and further consequential damages cannot be excluded. Depending on the application of the type of tube, it must be ensured that the voltage difference between the filament and the cathode does not exceed the maximum permissible value. It is absolutely essential to observe the specifications in the manufacturers' datasheets.

Figure 6 shows a circuit that simulates the behavior of a rectifier tube. The mains transformer is calculated normally, which is customary practice for a silicon bridge rectifier with a voltage drop of about 1.5 V. The resistance of 150 Ω provides the voltage drop at the rectifier tube as well as the voltage drop at the series resistors necessary for the tube. The resistor is dimensioned so that approximately the same voltage drop as the tube is achieved.

Still using the EZ81 as an example, approximately 15 V must drop at 100 mA, which results in a resistance value of 150 Ω. With these values, about 1.5 W are converted into heat. So for safety reasons, a resistance with a resilience of 4.5 W is used.

The switch S allows switching between the simulated "rectifier tube operation" and silicon rectifier. To achieve a DC voltage of 300 V, the transformer must have approximately 236 V with the 150 Ω resistor, without the latter the voltage is 214 V.

Builder's Choice

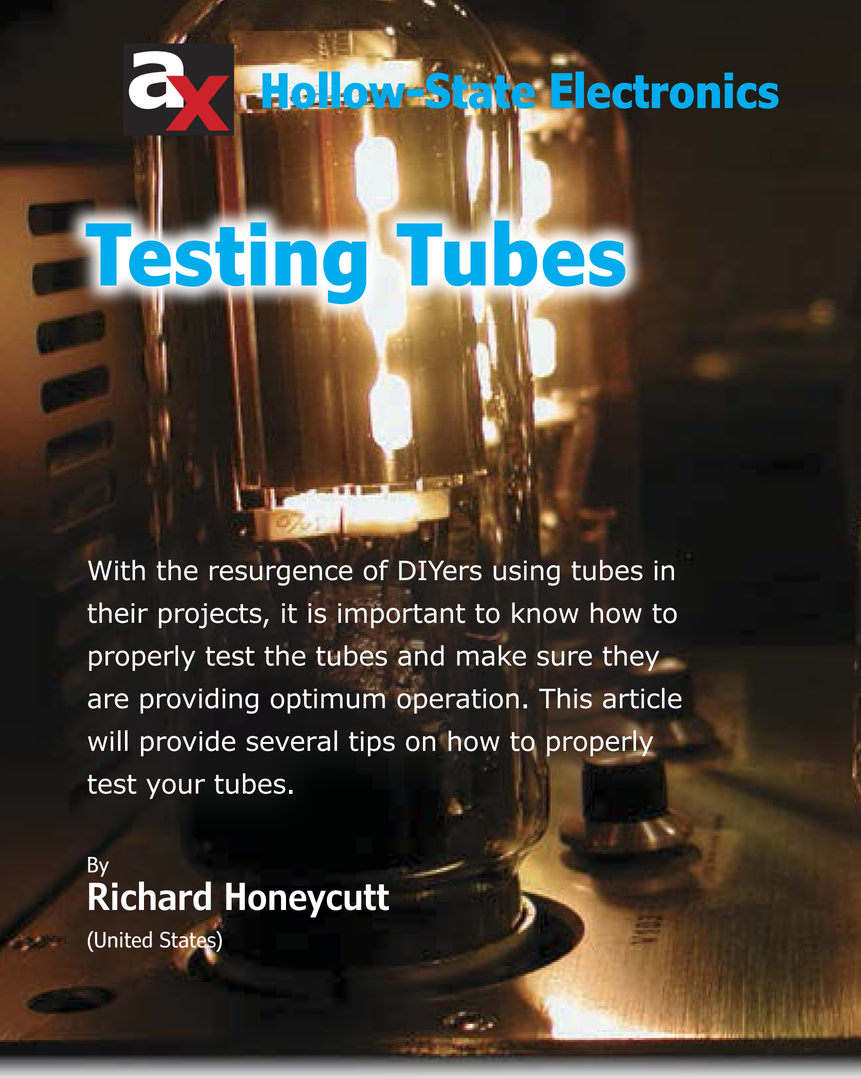
Finally, using a tube or a semiconductor as a rectifier is a matter of taste. In any case, semiconductors are more cost-effective, they feature lower losses, are wear-resistant, and resistant to surge currents. In our rectifier tube example, we have a transformer output of approximately 47 VA, because 6.3 VA are added for the heating of the EZ81. With silicon rectifiers, the transformer rating is only 24 VA. Furthermore, a simple high-voltage winding is sufficient, which saves production costs. Overall, the nonexistent heating power of the tube and the low voltage drop of about 0.7 V compared to 15 V per each tube diode, plus the voltage drop at the current-limiting resistors have their effects.

Rectifier tubes are subject to a similar wear as that of power tubes. In this case, the thermal load and the power conversion are higher than they are with preamplifier tubes. Moreover, the effort to be made in the transformer is higher than silicon rectifiers may require. Those who swear by the rectifier tubes must remember they are significantly more expensive than silicon.

You also should consider the follow-up costs resulting from the routine tube replacement. The smoother response behavior because of the smoother cut-off characteristics could sway you to use rectifier tubes. However, this can also be achieved with low power loss when using modern diodes with corresponding wiring.

Finally, there is still an important note. Booster diodes, which used to be installed in color and black and white TV sets equipped with tubes, are absolutely not suitable for mains rectifiers! Even if the high dielectric strength, the high maximum peak current, and the low price may seem appealing, this type of diode is only designed for pulse operation in the microsecond range!

When choosing a rectifier tube, it must be suitable for the intended operation, and it must be stated that it can be used with a mains transformer. In this case, the limiting values specified by the manufacturers must be respected. These are, above all, the maximum permissible transformer voltage, the maximum capacitance of the charging capacitor, the minimum current-limiting resistance, and the way the heating operates. Only then, can a successful and safe operation be guaranteed.



Testing Tubes

With the resurgence of DIYers using tubes in their projects, it is important to know how to properly test the tubes and make sure they are providing optimum operation. This article will provide several tips on how to properly test your tubes.

By
Richard Honeycutt
(United States)

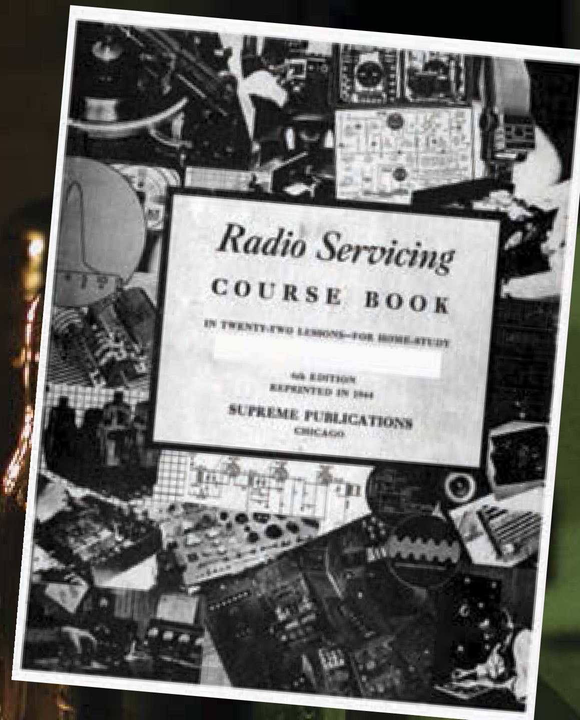


Photo 1: Home-study books like this one introduced many hobbyists—and future professionals—to electronic servicing.

About 1960, my dad gave me a copy of the *Radio Servicing Course Book* by Supreme Publications (see **Photo 1**). While studying this practical text, I found to my surprise that the authors considered capacitors (which they called condensers) to be the most likely component in a radio to fail. I, like probably most other folks at the time, considered tubes to be the first suspects, mainly because of their resemblance to light bulbs, which often had to be replaced.

Even if tubes were not the least reliable component in consumer electronics at that time, they were the only ones that could be tested and replaced by untrained owners of the equipment. (The other user-replaceable part, a cartridge fuse, was seldom found in consumer electronics at that time.) Thus, a market developed for replacement tubes and convenient tube testers that were available to the general public. Tube testers such as the one shown in **Photo 2** were commonly found in drug stores and even the occasional grocery store or "dime store." As you can see, several drawers full of replacement tubes were available for purchase in case the suspect tube(s) tested as defective.

Not surprisingly, tube testers made as point-of-sale merchandising devices were not very technically sophisticated. Most equipment owners figured that since a good tube glowed, a bad one wouldn't. This simplistic line of thinking led to continuity-type tube testers, which only verified that the filament would pass current. Such testers were nearly useless, since the "glow" test could give the same information without the need for special equipment.

Test Purpose

So, how should a tube be tested? That depends on the purpose of the test. If the test is to determine whether a tube is bad or good (likely to work properly in the equipment), the tester must meet less stringent requirements than if tubes are being matched for high-fidelity output stages or for instrumentation amplifiers. With this fact in mind, let's see what methods can be used to test a tube.

The most reliable way to test a tube for proper operation in a specific piece of equipment is to replace it with a new or known-good tube of the same type. If replacing the tube restores proper operation, the old one can be assumed bad. There are two shortcomings of this method: if more than one tube is bad, replacing the tubes one at a time may not definitively indicate which ones are defective, and even if only one tube is bad, you have to have a complete spare set of tubes to test the equipment this way.

As I mentioned earlier, continuity-type tube testers are useless unless the filament is burned out, which has been an uncommon failure mode since the 1930s. So the next type of tester we'll discuss is the "tube checker." This device treats all tubes as low-power rectifiers, connecting all elements except the filament together to act as a plate. The correct connections for each tube type are made via switches. A fairly low "plate" voltage is used, and a meter

Photo 2: Drug-store tube testers like this one were not very useful, except for selling tubes.

measures the current, which is a fraction of the normal emission. The meter may be calibrated in percentage or simply as "BAD----GOOD." Tube checkers are sometimes called emission testers, because the meter reading is a crude indication of the cathode emission in tubes with directly heated cathodes. For tubes with indirect heating, the measurement is contaminated by leakage.

Emission Testers

The true emission tester is a more refined development of the idea behind the tube checker. It uses switches to connect the cathode to ground, and all the grids and the plate to B+. A meter is connected in series with either the cathode (yielding what has been called a "cathode conductance" tester), or the plate ("plate conductance tester"). **Photo 3** shows a Sencore Mighty Mite, a popular emission tester. This model checks tubes near their normal operating voltage level, allowing it to identify marginal tube defects that may be missed by testers operating at lower levels. It also checks for grid leakage, which is a common failure mode. Whereas some emission testers, and even some dynamic conductance testers, have a 2 to 5 MΩ leakage sensitivity, the Mighty Mite has a leakage sensitivity of 100 MΩ. The schematic diagram of the Sencore Mighty Mite is shown in **Figure 1**.

Most emission testers also include a short-circuit test, which performs a continuity test between elements. Usually this involves an AC voltage with less than 65 V applied between pairs of elements, and a neon lamp used as an indicator of shorts. This is an especially important test for tubes used in equipment exposed to a lot of vibration, such as guitar amplifiers (or car radios, in the olden days).

To test a tube using the Mighty Mite or a similar tester, the user looks up the tube type in a book supplied with the tester. The book tells the user which socket to plug the tube into, and gives the switch settings for testing. The FUNCTION switch is then set to SHORT TEST, and switch D is rotated through all positions, from H-K to A. If the short indicator lamp glows, the tube has high internal leakage or a short, with certain exceptions detailed in the user's manual. Next the FUNCTION switch is set to EMISSION, and switch D is set to the position indicated in the book. Emission quality can be read on the upper scale of the meter. Finally, the FUNCTION switch is set to GRID LEAKAGE. The meter reading appears on a GOOD-?-BAD scale.

When using an emission tester to assess the quality of a rectifier or power output tube, you can set the tube at one click below its rated heater voltage (e.g., 5 V for a 6.3 V filament, etc.) and see if the emission still indicates good. If not, the tube may not perform well when asked to provide a high output current. Tubes that fail this check may also suffer from weakened internal supporting structures due to long usage and might catastrophically fail if not replaced.

You can test preamp tubes for microphonics by gently thumping the tube while testing emission and leakage. If the meter needle vibrates when the tube is thumped, the tube is probably microphonic.

As a group, emission testers have several shortcomings:

- They cannot measure key characteristics of tubes, such as transconductance.
- Many models do not test under real load, voltage, and current conditions.
- The test uses only DC conditions, not the AC signals encountered in normal operation. This fact can result in failures such as low gain or transconductance not resulting in a BAD indication.
- Tubes with grids may show deceptively high

Photo 3: This Sencore TC142 MightyMite V emission-type tube tester was common in service shops for many years, and used ones can still be found online.

emission because of *hot spots* in the cathode that emit additional electrons which are picked up by the grids during testing, but are shielded by the grids during normal operation.

• Grids will be forward biased to some extent; some fine control grid wires are limited in their ability to withstand this. This hazard to the tube can be mitigated by applying the emission test for as short a time as possible—end the test as soon as the meter reading stabilizes.

The main advantage of an emission tester is that, of all types of tube testers, it provides the most reliable prediction of impending tube failure. If emission is at 70%, transconductance and gain can still approach normal levels. This behavior requires judgment on the part of the technician, because most tubes with somewhat low emission will operate properly in the circuit.

Mutual Conductance Testers

Mutual conductance is an old term for transconductance (g_m), which is the ratio of a small change in plate current to a small change in grid voltage that causes the plate current to change. Mutual conductance testers test the tube by applying the tube's normal plate voltage and grid bias voltages and an

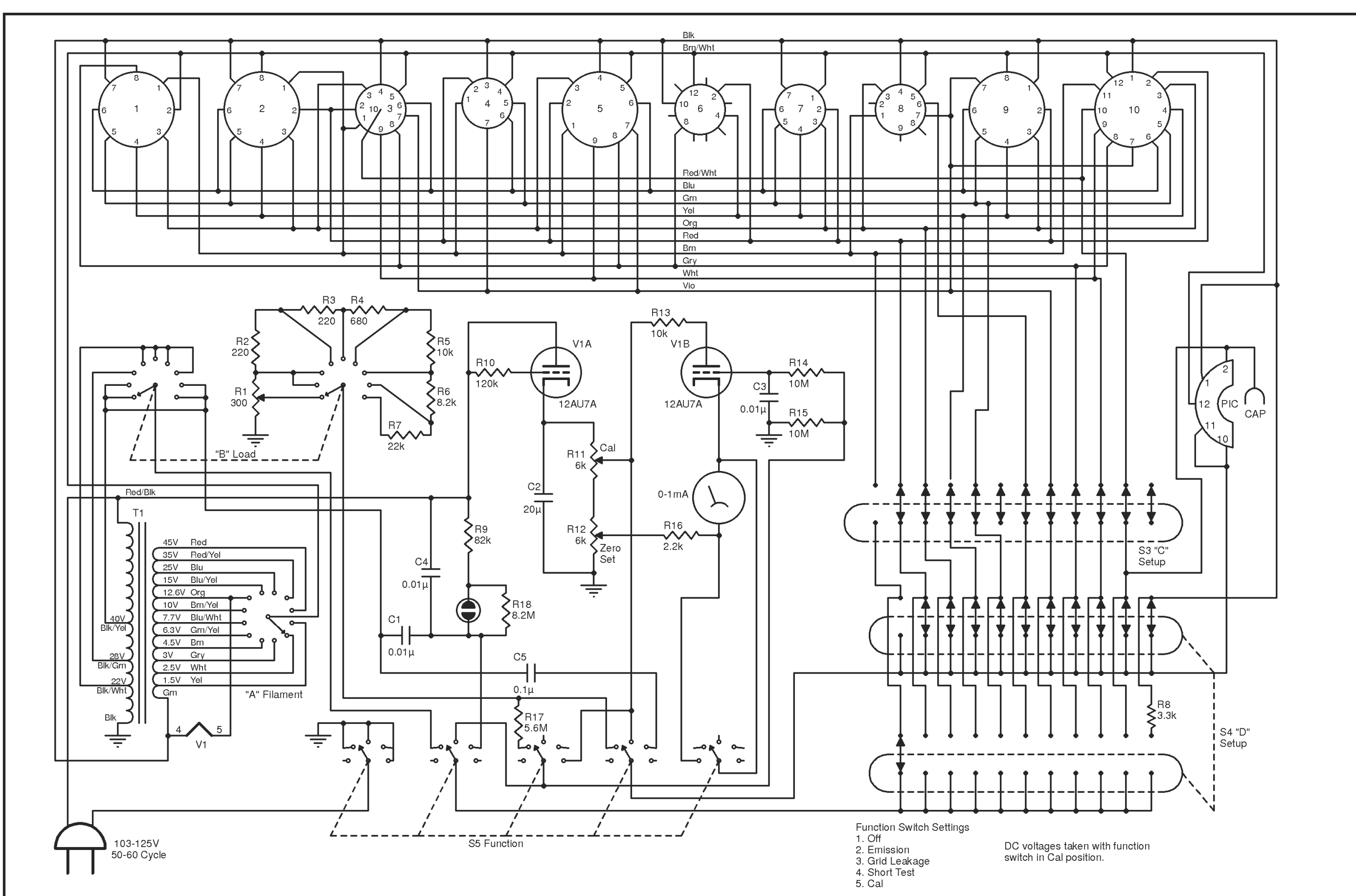


Figure 1: This is the schematic of the Sencore Mighty Mite.

AC voltage to the control grid, then measuring the plate current. The transconductance in microsiemens (previously called micromhos) is displayed on a meter. The measured value can be compared to the value specified in a tube manual.

The Hickok 539C tester shown in **Photo 4** is a fine example of a mutual conductance tube tester. The operation procedures are conceptually much like those for an emission tester, but the switches actually set up the voltages applied to the tube terminals under direct user control. Rather than supply a book listing tube setups for various tubes, the Hickok testers incorporated a scrollable chart. Many of the Hickok testers also included a line-voltage meter and a control for adjusting it, since the mains voltages were less stable in the past than they are now. Hickok testers dominated the market for mutual conductance testers.

Hickok also made numerous testers for the military, and these did not bear the Hickok logo. One of these, the AN/USM-118A/B tester shown in **Photo 5**, was the military version of the 1234A Cardmatic tester, which used punched cards rather than switches to set up the tests. A card was required for each tube type to be tested.

Mutual conductance testers are preferred over emission testers for matching tubes which will be operated in pairs. They are also useful for the experimenter who wants to identify similar-looking tubes with markings that are no longer readable. For example, a 12AX7 looks almost identical to a 12AY7, but has a transconductance of 1,600 μS, vs. 1,750 μS for the 12AY7.

Dynamic Conductance Tester

The dynamic conductance tester is a variation of the emission tester with similar advantages and disadvantages. This type of tester was made by several different manufacturers, and provided an indication related to transconductance, but not a true measurement of transconductance. The similarity of name ("dynamic conductance" vs. "mutual conductance") was mainly designed to avoid infringing on the patents and the trademarks for the mutual conductance tester, held by Hickok Electrical Instrument Company.

DC and AC Parametric Testers

Parametric testers enabled the user to apply the DC and AC voltages to the tubes' elements as required for measuring the tube parameters specified in a tube manual. Such testers were quite sophisticated, and found use mainly in laboratories in which new tube types were developed, and as QA testers for factories.

Photo 4: This 1940s-vintage Hickok Model 539C mutual conductance tester has been highly respected by professionals and hobbyists for many years.

Photo 6: The military version of the Hickok Cardmatic tester provided simple setup via punched cards and made accurate transconductance measurements.

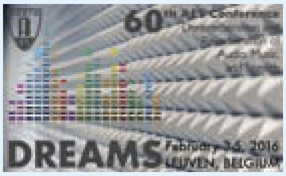
Today's hobbyist or professional in hollow-state audio can often find serviceable tube testers available online for a price not much higher than the original selling price. Manuals for these devices are also often available, and many of them contain schematics and/or troubleshooting and calibration instructions. But several new-production testers are also available, and these will be the subject of a future article.

Resources

M. Beitman, Radio Servicing Course Book: In Twenty-Two Lessons for Home-Study, Supreme Publications, 6th edition, 1944.

Industry Calendar

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



February 3–5, 2016
Audio Engineering Society (AES) 60th Conference on Dereverberation and Reverberation of Audio, Music, and Speech
Leuven, Belgium
www.aes.org/conferences/60

The Audio Engineering Society (AES) 60th International Conference will be held February 3–5, 2016, at the Leuven Institute for Ireland in Europe. The conference will focus on Dereverberation and Reverberation of Audio, Music, and Speech and aims to provide a forum for researchers working on the modeling, control, removal, and synthesis of acoustic reverberation. The interdisciplinary nature of the topic will cover disciplines such as room acoustics, psychoacoustics, and signal processing; hence, the conference is targeted at researchers working in any of these disciplines.

February 9–12, 2016
Integrated Systems Europe (ISE) 2016
Amsterdam RAI, The Netherlands
www.iseurope.org

Currently one the most successful and fastest growing shows in the world—and jointly promoted by InfoComm and CEDIA—this European event is becoming better every year for the audio industry, with a focus on Audio over IP (AoIP), wireless audio, immersive installations, and the latest in electroacoustic solutions. Due to the convergence between consumer and professional technologies in the electronic systems integration markets and the expansion of the installation market worldwide, ISE has become the largest showcase of IP-based technology solutions, home networks, and building management.

For the first time, ISE 2016 will be held for FOUR days. The sold-out exhibition floor of the 12th ISE will provide exhibitors and visitors with a friendly environment for doing business, showcasing innovation, and helping shape the future of the industry.

February 10–12, 2016
Audio Engineering Society (AES) 61th Conference on Audio for Games
London, UK
www.aes.org/conferences/61
www.audioforgames.net/2016

Sponsored by Dolby, DTS, Genelec, AudioKinetic, Tazman Audio, and other companies dedicated to audio for games, this three-day conference will be a unique opportunity for all those interested in game-audio production, platforms, and tools now evolving toward new 64-bit mobile platforms. The event takes place at the Royal Society of Chemistry, in the center of London, and promises to be a unique opportunity to learn more about interactive experiences, real-time synthesis, and spatial audio.

February 22–25, 2016
2016 GSMA Mobile World Congress
Fira Gran Via, Barcelona, Spain
www.mobileworldcongress.com

A decision to attend this gigantic and chaotic event should not be taken lightly. But, it is a must-see if your job involves developing technologies for mobile devices and you actually have to work with the telecom giants. The Barcelona event is also evolving toward wearables and smart technologies in general, competing with both the CES in Las Vegas, NV, and all the European electronics trade shows, which is one reason why many audio companies and professionals are adding the Mobile World Congress to their calendars.

Article Update

Our March 2015 *audioXpress* issue featured an article by George Ntanavaras titled "Build a Voltage and Current Peak Detector." The detector is a simple portable device that can help answer the question about peak voltage and peak current requirements and whether or not your power amplifier is capable of driving your loudspeaker without clipping.

Recently, Ntanavaras provided us with a modification to his design. He suggests removing the Diode D4 from the circuit of the peak detector. Then, insert a new diode D5 (type 1N5819) between the battery (+) pole and the 9 V pin of the circuit. This will offer better protection in case of accidental reverse polarity.

The file illustrating this modification and the complete article can be found at: http://audioxpress.com/article/You-Can-DIY-Build-a-Voltage-and-Current-Peak-Detector.html.

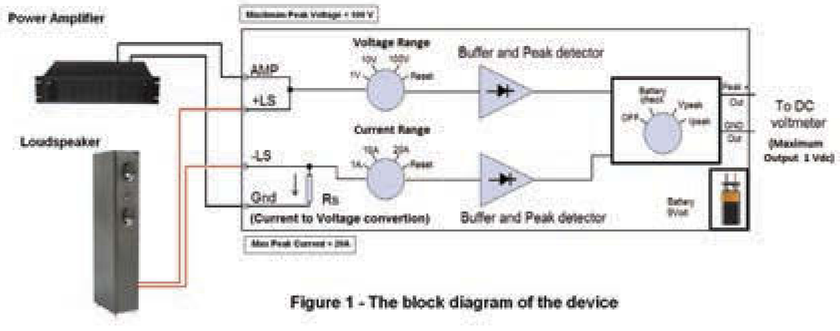


Figure 1 - The block diagram of the device