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Doing It Differently

Time moves quickly. We are already feeling the aftermath of 2014's first two major industry shows. This is also a year when *audioXpress* is completing its transition to an expanded publication that addresses the needs of the audio engineering community—not only for those who have fun listening to music (there are plenty of magazines doing that) but mainly for those who imagine, create, and work with audio technology.

This year began with the 2014 International Consumer Electronics Show (CES) in Las Vegas, NV, introducing innovations on all fronts. It was also the largest CES in show history. While some companies introduced products based on users' needs, it appears many consumer electronics companies still prefer to throw hundreds of new ideas at the wall to see what sticks. I guess a major electronics show like the CES is the ideal place to test those ideas, but sometimes we have to wonder why the successful companies that only introduce market-ready products don't even need to attend the CES.

Yes, we miss seeing Apple at trade shows and we miss the inspiring clear vision of the late Steve Jobs. Apple is one those companies with products that are the perfect combination of state-of-the-art technology and innovation that are available for purchase exactly as advertised. And while the company was not in attendance, Apple's products still dominated the 2014 International CES. It is no surprise that many great ideas and reference designs were designed to complement the iPad, the iPhone, and even the new Apple Mac Pro workstation.

IK Multimedia promoted its iRing wireless sensors to control music apps (or any other apps) using only gestures. We've also seen great photography peripherals for the iPhone and many new charging and home-automation solutions. There are even iOS-device-controlled robots and drones. And of course, no audio company could ignore the huge market created for wireless speakers and headphones. Many were especially designed for Apple's mobile devices, leveraging Apple's push for Bluetooth Smart 4.0 and AirPlay technologies. Apple also effectively revitalized the worldwide home audio market.

Wireless speakers, headphones, soundbars, integrated A/V receivers and audio systems are experiencing impressive growth rates, according to recently published market reports. Bluetooth products, in particular, continue to bolster the wireless speaker market, offering the convenience of portability, while multi-room audio based on Wi-Fi is also on the rise. Among the 20,000-some products introduced at the 2014 International CES, there were a significant number of new headphones and earphones.

After every CES, we should also acknowledge those sparks of inspiration from obscure companies and the truly exciting technology announcements. For example, cars connected to mobile networks—actually talking and seamlessly interfacing with our mobile devices.

It's always difficult to understand why, but clearly, in the middle of all the Internet-connected toothbrushes and forks, speech-recognition watches, and curved television screens, some innovations make complete sense and leave us asking ourselves "why did it take so long?"

João Martins
Editor-in-Chief

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COLUMNISTS

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

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

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

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

Prism Sound dScope Series III—New Automation Solution and Updated Soundcard Support

Prism Sound announced a package of important new features for the dScope Series III audio analyzer system used in worldwide R&D labs and production lines.

The new capabilities include improvements to the “Auto Sequence,” and easy-to-use “check-box” test automation system that provides a simple way to set up and execute a series of tests. It also enables users to produce high-quality reports in HTML and Microsoft Excel formats without writing any code. This update provides a powerful Edit mode that makes sequence construction and adjustment even faster so automated production line tests can quickly and easily be set up using the design engineer’s test configurations as a starting point.

For example, it is now possible to locate an existing test configuration file and directly import it as a new step in the test sequence. You can also rename, copy, delete, and re-order test steps.

The analyzer system also includes a “tweak” feature that enables a single parameter adjustment to be automatically applied to several test steps, avoiding the need for manually editing individual test configurations. For the Auto Sequence update, a new Dialog (or Form) editor enables easy construction of pop-up forms for user interaction (e.g., entering serial numbers and other information, confirming actions such as external test fixture or device adjustments, or confirming observations of the equipment under test).

Prism Sound pioneered sound card testing in the dScope Series III audio analyzer system by using a direct software interface between the test application (dScope) and the sound card’s Windows (WDM) driver, combined with the complementary function of the dScope hardware connected to the other side of the sound card. This method can be used to drive the soundcard from the dScope software audio generator and to measure the analog sound card output with the dScope’s analog measurement input. You can also drive the sound card with the dScope’s precision analog generator and determine the audio performance using the dScope’s software analyzer functions to measure the audio stream returned from the sound card.

The new software release adds the capability to support ASIO drivers and WDM, enabling a range of professional and consumer audio sound cards and interfaces to be tested. In addition, ASIO is a useful tool that guarantees bit-perfect transfer between the host computer and the audio device, which is beneficial when testing the interface device’s accuracy.

In addition, dScope’s Continuous Time Analyzer (CTA) is now available for use with sound card inputs in addition to the fast Fourier transform (FFT)-based analysis previously provided for sound cards. dScope’s unique FFT detector meters provide all the usual audio measurements (e.g., amplitude, noise, THD+N, and many others). The CTA time-domain analysis process mirrors the conventional notch filter plus voltmeter method commonly used in analog instruments.

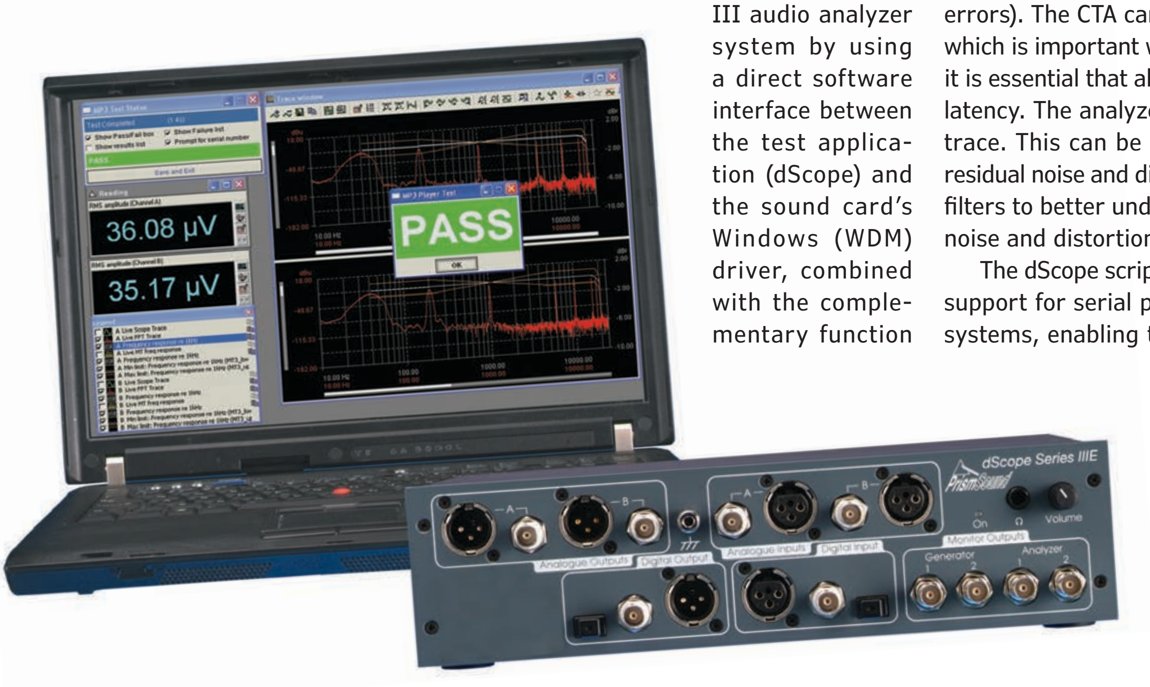
The CTA has several advantages: it is faster than the FFT analyzer, especially when performing sweeps; it can make peak and quasi peak measurements; and perhaps most importantly, it can be used to detect occasional glitches (e.g., sample errors). The CTA can measure interchannel phase, which is important with computer interfaces where it is essential that all channels experience the same latency. The analyzer can also generate an output trace. This can be used to measure and plot the residual noise and distortion after the measurement filters to better understand the composition of the noise and distortion components.

The dScope scripting environment now provides support for serial port control on 64-bit Windows systems, enabling the control of external devices using RS-232/4-22. This could be used to control external switching, DC power supplies, or other equipment.

The new release is available for download from Prism Sound’s website.

Prism Sound
www.prismsound.com

The dScope Series III is a comprehensive and powerful measurement system for analog and digital audio generation and analysis. Its features include digital audio carrier parameters, acoustic transducers and Windows sound devices. An audio signal generator and audio signal analyzer in one system, the dScope Series III is primarily a software application with a dedicated hardware interface simply connected to the host PC using a USB cable.



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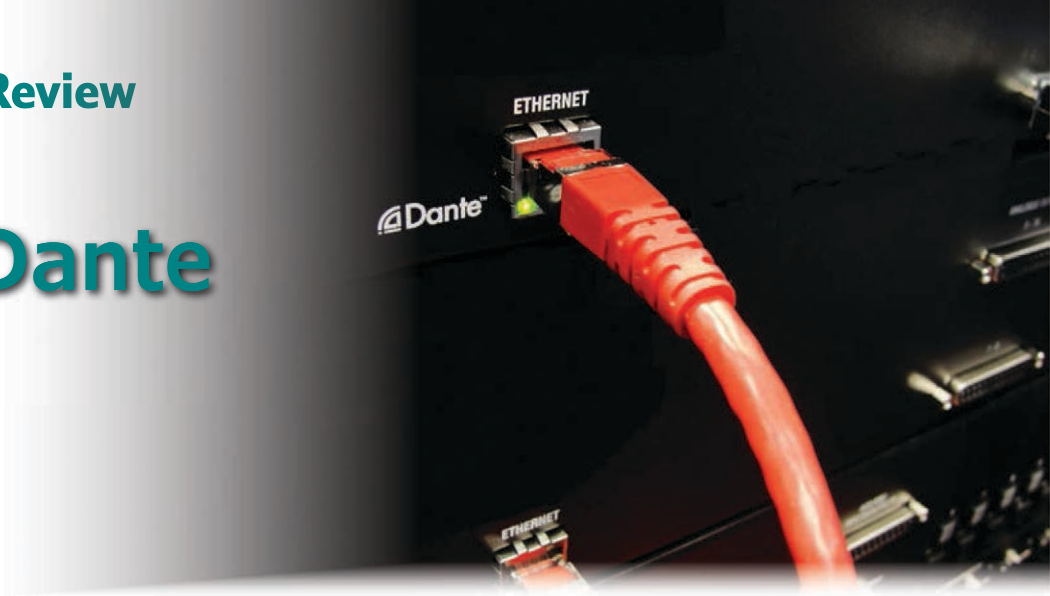


Audinate Dante (Part 2)

Audio Networks Fast to Market

The story of Audinate and the Dante audio networking implementation is one of the most interesting examples of perseverance and focus in the industry. It demonstrates that the evolution to an industry standard comes from having a clear vision of the market's needs and fulfilling those necessities with practical implementations of converging technologies. In the second part of this article series, we address how Audinate managed existing industry efforts and commercial requirements to achieve marketing success.

"We are not driven by what we think is right but what our customers want. At the end of day, that's what drives the market."
—Lee Ellison, Audinate CEO

By
João Martins
(Editor-in-Chief)

Founded and based in Sydney, Australia, Audinate was recognized as one of the 50 fastest growing technology companies by the Deloitte Technology Fast 50, Australia 2013. Deloitte Technology Fast 50 ranks the fastest growing public or private technology companies based on their percentage revenue growth over the previous two years.

As David Myers, COO of Audinate, explains, "The strength of Audinate's technology offering continues to grow along with our customer base. Our wide portfolio of solutions allows our customers to maximize the interoperability of their products and realize revenue potential from the latest network innovations—making Audinate the solution provider of choice for pioneering Audio/Video equipment manufacturers."

Built on existing networking protocols and standards, Audinate's Dante technology became a "plug-and-play" networking solution for OEMs, integrators, and the audio industry in general.

Audinate's Dante solution has been licensed by some of the largest pro audio industry players. The company is starting to leverage the technology to make inroads in the A/V market and the broader commercial installation space.

Audinate was founded in 2006. After more than three years of intensive research and development by the company's founders—who are all leading computer networking experts—they found a solution to the problem of transporting high-quality audio and media over standard TCP/IP computer networks. Their discovery helped solve the problem of long, variable delays through the network and provided a way to tightly synchronize multiple audio

A simple example of Dante implementation is the A820-NIC-DANTE card for the Shure SCM820 automatic mixer, offering multichannel audio networking over a single Ethernet connection and computer-based playback, recording, and signal routing via Dante Virtual Soundcard and Dante Controller software.

outputs. Audinate also took a novel approach to simplifying network setup and management, integrating conventional network routers and using simple network configuration concepts.

The Dante patent was filed in 2003, but the technology has been continuously evolving with new networking standards. Basically, as Audinate states: "If a piece of audio equipment is Dante-enabled, this means that it is capable of transmitting and/or receiving audio channels to/from other Dante-enabled equipment over a standard local area network running Internet protocols (TCP/IP, UDP/IP etc.)."

In addition to providing basic synchronization and transport protocols over Ethernet Layer 3, Dante provides simple plug-and-play operation, PC sound card interfacing via software or hardware, glitch-free redundancy, and support for routed IP networks. Dante implementations use the Institute of Electrical and Electronics Engineers (IEEE) 1588-2002 standard for synchronization, UDP/IP for audio transport, and are designed to exploit standard gigabit Ethernet switches and VoIP-style QoS technology (e.g., Diffserv model architecture).

In line with its commitment to be AVB compatible, Audinate has produced versions of Dante that use the new Ethernet Audio Video Bridging (AVB) protocols, including IEEE 802.1AS for synchronization and RTP transport protocols. It is committed to supporting both IEEE 1733 and IEEE 1722.

Existing Dante hardware devices can be firmware upgraded as Dante evolves, providing a migration path from existing equipment to new AVB capable Ethernet equipment. Recent developments include announced support for routing audio signals between IP subnets and the demonstration of low latency video.

The Licensing Approach

Since its early days, Audinate has concentrated on the distribution and licensing of Dante, helping OEMs minimize audio network implementation with ready to use solutions, including chip-level and hardware reference-designs in conjunction with technology partners such as Xilinx and Dayang OMS.

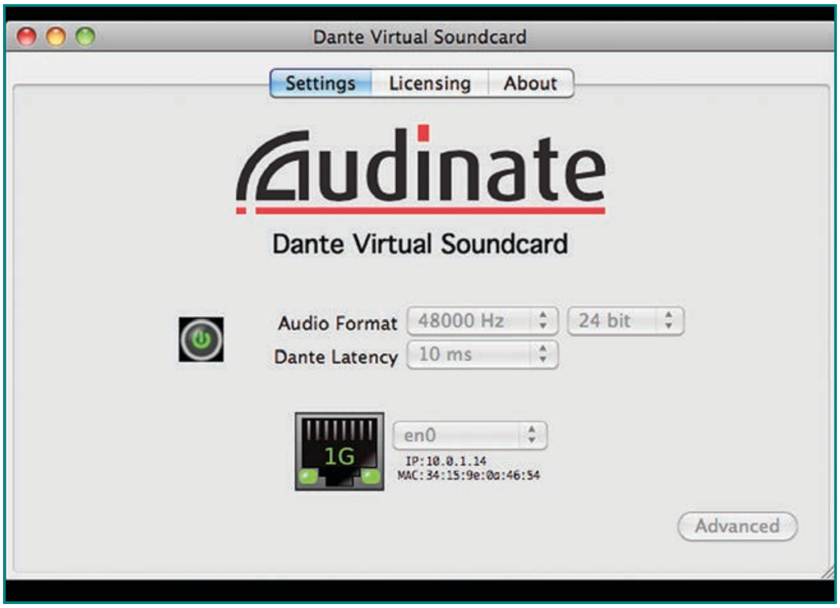
Independent of the complexity level of the implementation, Audinate provides all the essential implementation tools in terms of hardware, software APIs, and licensed applications (e.g., Dante Controller and Dante Virtual Soundcard) and support components.

Hardware-reference designs and implementation tools include the Dante Core Module (DCM) and the Dante Brooklyn II modules with support for 64 × 64 audio channels and the Dante PCIe Soundcard for PC or Mac, with support for up to 256 channels of uncompressed, 24-bit digital audio and/or up to 192-kHz sampling frequency on a Gigabit Ethernet interface.

A recent example of Audinate's commitment to easy implementation was the launching of the Dante Ultimo solution, a fully featured, ready-to-use Dante interface for networked audio products that is integrated into a single chip. Ultimo provides a cost-effective networked audio solution over a 100-Mbps Ethernet interface to a new range of audio devices with low channel count and high audio quality.

Ultimo also enables low-jitter clocks with ±1-μs time alignment between networked devices. The solution does not require specialized switches. It works with existing network infrastructure and offers all the well-established Dante features (e.g., automatic device discovery, plug-and-play networking, network-based firmware updates,

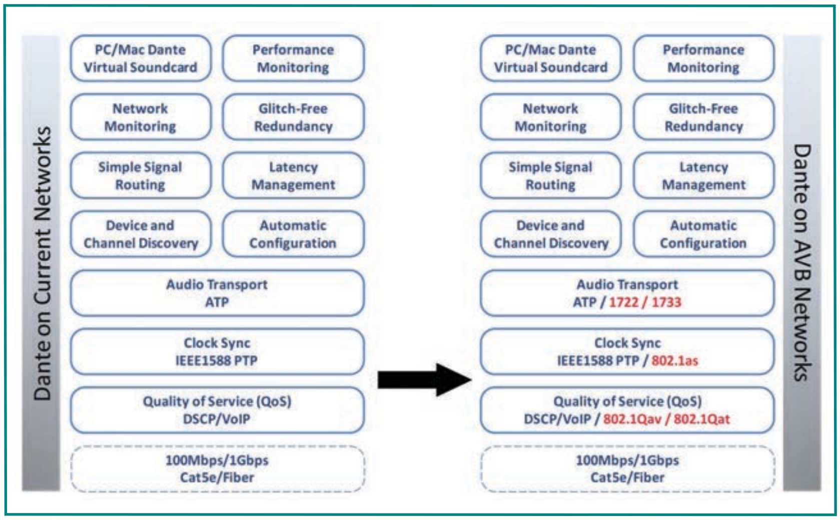
Audinate Dante is the first IP over Ethernet networking solution for the professional industry, capable of transmitting many channels of high bandwidth, uncompressed sample accurate digital audio with imperceptible latency, and high synchronization performance. When Solid State Logic started providing a Dante-MADI bridge to the studio market, it became clear the technology is now a "de facto" standard.



The Dante Virtual Soundcard software enables any PC or Mac to connect to a Dante audio network, using only the computer's Ethernet port to communicate with a network of other Dante-enabled devices. No special hardware is required. Audio applications recognize the Dante Virtual Soundcard as they would any standard ASIO, WDM, or Core Audio interface.

and customization of device names and channel labels).

Applications targeted for Ultimo include powered speakers, microphones, speakerphones, amplifiers, paging stations, personal monitoring systems, AV wall plates, recording interfaces, analog/digital, break-in/break out, and even musical instruments. Product development kits, in partnership with IED, Attero Tech, and Stewart Audio are also available.



The Dante architecture includes Ethernet Audio Video Bridging (AVB–IEEE 802.1) support, according to Audinate.

About this recent initiative clearly aimed "more toward prosumer than consumer" applications, Audinate CEO Lee Ellison revealed the implementation will expand from the 2 × 2 channels currently supported. "The path to 4 × 4 channels will open the doors for a lot of different products, when you hit that target at a very low cost. If we look at a network it's the stuff at the edges that there is more than anything else."

The Good Thing About Standards

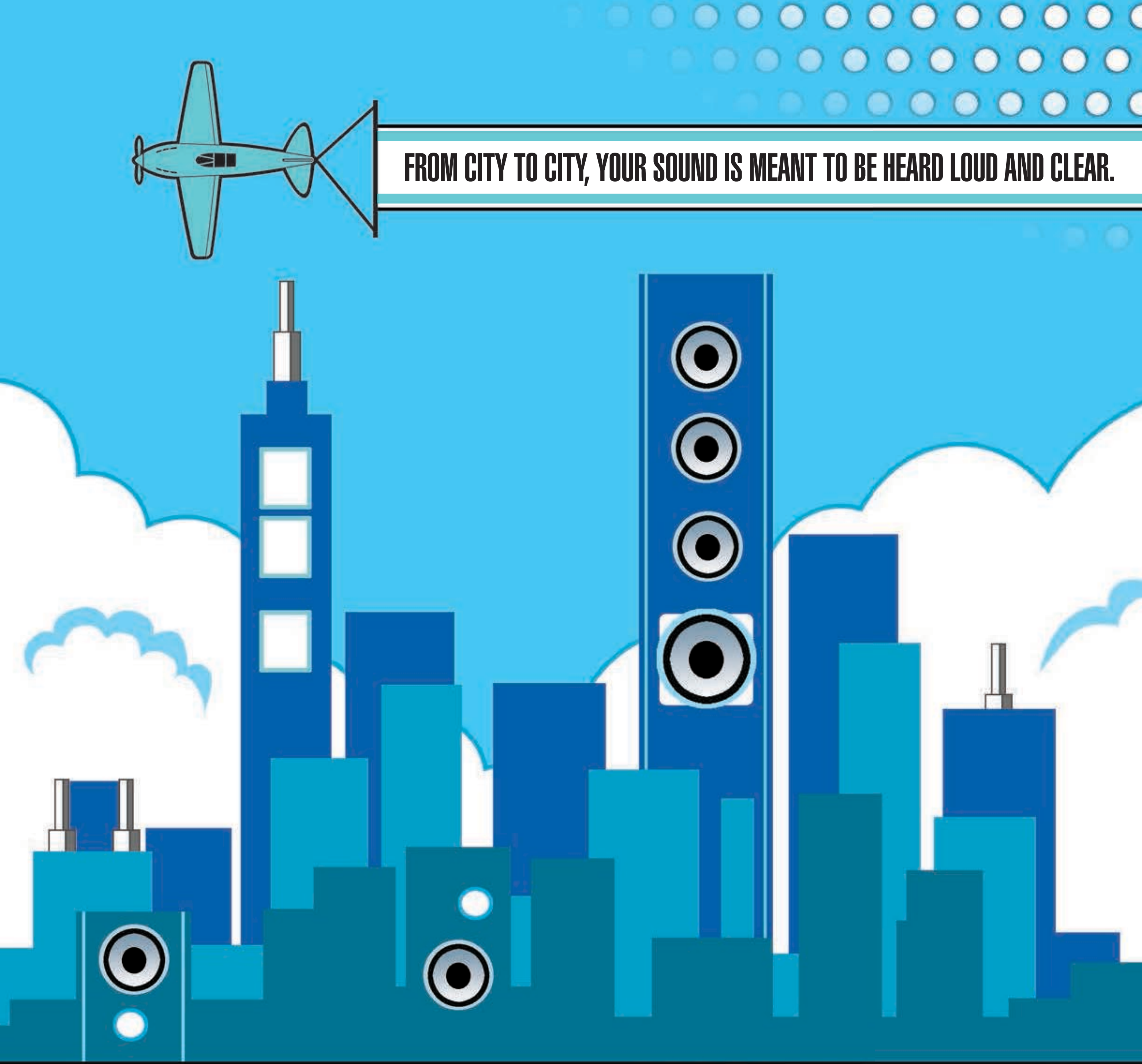
Another early strategy followed by Audinate was to join and contribute to the most relevant audio industry efforts within the Audio Engineering Society (AES) or the IEEE Audio Video Bridging Task Group (IEEE 802.1-AVB). Audinate has announced that Dante will be AVB compliant as these standards are ratified, and is a Promoter Member of AVnu, the industry group that seeks to promote and certify solutions based on the new AVB standards. In fact, within the AVnu Alliance Audinate already demonstrated Dante and AVB running side-by-side on the same network interface.

In our interview with Ellison, we discussed in detail the position of Audinate in regard to the "continuously moving" environment, as illustrated by the recent publication of the AES67 interoperability standard (*audioXpress,* January 2014) or the AES X-210 project promoted by the Open Control Architecture (OCA) Alliance, aiming to develop a media networking system control standard for professional applications.

Naturally, we started with the AVnu Alliance and the AVB developments to which Ellison confirms, Audinate is totally committed to support, but not necessarily embrace.

"AVB is a suite of standards and people like the concept of having a standards-based approach. It gives them a comfort level. There are products that I use everyday in my life that aren't totally 100% open, like setting up a meeting on my Mac and someone sends me something back on a Windows machine, talking on a iPhone and the other person on Android… It's not necessarily bad. There is a company that stands behind it, that completes it, and adds features and wraps that around it.

"Dante is just not a QoS or a time sync protocol. Dante is really the complete wrapper and tool kit around all of that. So an example I like to equate us to is Red Hat. They provide a Linux suite and tools in a complete solution so that companies don't have to fix all the things that exist in open source and can get to the market quickly. And they are already well over a $1 billion company now. Well, what we try to do is to give our customers a wrap around those



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Four Audio was one of the first manufacturers to bring a two-channel Audinate Dante Breakout box to the market, based on the Ultimo chipset. It is a cost-effective solution for getting analog signals in or out of a Dante audio network without much infrastructure. It is also ideal for speech or music rooms or multiroom installations in buildings.

tools too. AVB is one of the tools that customers are requesting. We are not driven by what we think is right but what our customers want.

"At the end of day, that's what drives the market. AVB changes some things in terms of the network clock to the switches and as a result we have to work within an avenue to get those types of things tested. AVB, as a standard, has the potential to be interoperable but is not automatically interoperable. If we implement something in an island, there is a likelihood that things will not be quite compatible when they initially connect.

"The AVnu organization mission is to promote the interoperability and compliance testing so that the end-devices will be implemented. Initially there is an avenue in layer 3 and we also fully believe that, in the long run, routeability is going to see more and more user cases required. It's not an argument about layer 2 or layer 3... It's really about IP and that's what will be required in the market. That's precisely what drove AES67."

In reference to AES67, we asked Ellison, what this changes for Audinate.

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Bosch | www.bosch.com

SCM820 Automatic mixer
Shure, Inc. | www.shure.com

From 2 × 2 channels to complete multichannel solutions for commercial installations, Lectrosonics also offers Dante network audio interfaces and breakout boxes for its range of ASPEN DSPs.

"AES67 is an interoperability standard that will use RTP transport. RTP is a common transport used in VoIP, in IPTV or in videoconferencing. Most of the video going forward will be RTP as well, for AVB and non-AVB. What this means is that to support AES67 in Dante we would have to modify the transport layer to do that. That's about it. As I said earlier, we are not trying to force anything on any customer. What we are trying to do is provide a solution that they know works. Not just a technology that isn't proven.

"By the way, we have been doing this for about 10 years already. This stuff isn't easy and, if you think about it, the AVB organization has been doing the standards for about nine years. It takes a while…and I think that will be true with AVB as well as with AES67. Our job is to make that robust for our customers. We are polling customers now and finding out which direction is going to go.

"I don't think networking technology is about being "all-in". It's a competing environment. There is a misperception on the market about Dante versus AVB. We are a supporter of AVB. Do we think it will be the dominant standard? It's very unclear. Maybe it will or it won't. We don't know. We think its going to have a lot of momentum in automotive. We thought it was in pro audio and we will have to see if that takes place or not. Now… with AES67 is that going to get momentum? We don't know. But if there is momentum and if our customers want us to provide a solution, we will implement those capabilities for our customers."

Discussing the Open Control Architecture (OCA) Alliance initiative (OCA came from the Audio Engineering Society's AES-24 protocol architecture), where Audinate is one of the founding members with Bosch Security Systems, we've learned that both companies worked in close cooperation. "We have been working with Bosch for over four years. Bosch has adopted Dante in their products and they have combined it by putting the Open Control Architecture on top of it. They call it Omneo, for Bosch products," Ellison said.

"So, Omneo has a foundation of Dante networking with an Open Control Architecture on top. The OCA Alliance is trying to define a suite of profiles that manufacturers can use so that, at least for basic functionalities, you can send control commands from one manufacturer to another. We are able to transport control information over Dante and that would be an example of a layer of control. If the OCA becomes implemented on a series of products, we will be able to use Dante to transmit that control information.

"In the past, control was not so much a technical problem. It was more a manufacturer specific problem and I think that some companies realized that there is something they do in how they manage their systems that is special, and that there are some basic functions, like mute and gain that aren't that special and that the gross of the industry would benefit from being able to have more interoperability at a control level."

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The Bare Essentials of Tube Compression

By
Miguel Marques
(Portugal)

Rockruepel is a German pro audio equipment boutique manufacturer founded in Düsseldorf in 2004. Company founder Oliver Gregor hand builds every unit sold under the Rockruepel brand. When Gregor began his career building studios and handling repairs, his expectations regarding equipment quality and functionality were never met, which inspired him to create Rockruepel. His first achievement was the launch of the comp.one variable-μ stereo compressor in 2011.

The main philosophy behind this compressor was to strip it down so every part was strictly necessary for operation, while using some of the best components available. After this initial success, Rockruepel took the concept further in 2013 with the introduction of the comp.two, an even more straightforward yet powerful all-tube stereo compressor.

The comp.two is a completely hand-built 3U rack in a thick aluminum housing. Out of the box, it feels like a quality piece of gear. The sturdiness of every touchable part inspires confidence and it feels like a unit that has been built to last. As a dual-mono or stereo compressor, there's a set of controls for each channel in the front panel, which are arranged in a symmetrical disposition to the center of the unit. In the middle, two large Hoyt volume unit (VU) meters take up most of the front panel, with four two-position switches to activate true bypass—operating independently for each channel—stereo linking the channels and a power switch for the unit. The two twin center knobs are used to select the compression mode per channel. Users can choose from "amp mode," "flat" compression mode, or one of the three sidechain-filtered modes.

"Amp mode" enables the use of the comp.two's I/O stages without actually compressing anything. In extreme distortion settings, there will be some noticeable compression action. In these settings, the VU meter will show some gain reduction, which is caused by the tubes saturating and distorting.

For the sidechain filtering modes there are three frequencies available for the high-pass filter. These 6-dB per octave high-pass filters can be used at 54, 74, and 110 Hz.

Being a variable-μ compressor, there's no ratio control available as the amount of compression applied depends on how much signal enters the compression circuit. According to the user manual, the comp.two's compression ratio can be measured from anywhere between 1.5:1 and 10:1. An input and output fader are available for each channel on the extremes of the front panel. The faders are in the form of a knob but they act like a fader. They aren't stepped potentiometers or switches and they range from no signal to maximum I/O levels. For a typical high-end bus compressor, we were expecting to see stepped switches everywhere as it helps recalling the unity. But after using the unit for a few days, we could totally understand and endorse the designer's choice. There will be more about this in the "In Use" section of our review.

Each channel has three lower knobs that show the traditional threshold and attack and release controls, each with its dedicated scale. The threshold in this compressor is also not stepped. Attack and release times are the only controls that are configured using stepped switches, with 10 positions each. Reading the manual, we found this unit's available attack times range from 3.5 to 70 ms and the release times range from 200 ms to 4s. On the unit, they're just labeled F, 2, 3, 4, 5, 6, 7, 8, 9, and S. However, the manual has a complete description of exact attack and release times.

A Simple Approach

The comp.two's back panel is extremely simple, with both inputs and outputs on XLR connectors and a power International Electrotechnical Commission (IEC) socket. Despite the back panel's simplicity, it all looks very appealing due to the thickness of the aluminum and the overall great build quality and finish. Almost in an Apple kind of way, the comp.two has that same polished look and feel of the aluminum Mac Pro towers, only much sturdier.

For those who know or tried the comp.one, there are considerable differences between the first model and the comp.two. There's obviously a major cosmetic difference, as the comp.one had two retro-looking pinup girls on the VU meters. The comp.two meters have a plain simple look (no pinup girls). Personally, we prefer the sobriety of the comp.two's VU meters, although we are certain there will be those who will miss that unique approach in the comp.one design. The comp.two's knob caps are also different from the ones used on the comp.one. The comp.two has rounder larger knobs that are used in place of the thinner and rectangular ones chosen for the original model.

Differences between the original model and the comp.two are not restricted to cosmetic changes. The comp.one had a special operation mode called "Ruepel Mode" that enabled the user to dial in considerable amounts of distortion that basically converted the comp.one into a very expensive distortion unit.

The Ruepel mode used a rectifier tube and some extra circuitry to achieve that effect, an option deliberately removed on the comp.two. According to the manufacturer, the comp.two is now "even more reliable and stable" without that part of the circuit.

The Rockruepel comp.two is an all-tube hand-built stereo compressor made in Germany. This comp.one successor is a more refined and straightforward unit.

In any case, even though this distortion-specific mode is not available on the comp.two, we can still obtain considerable amounts of distortion from of the unit by just overdriving its input and output, with or without compression.

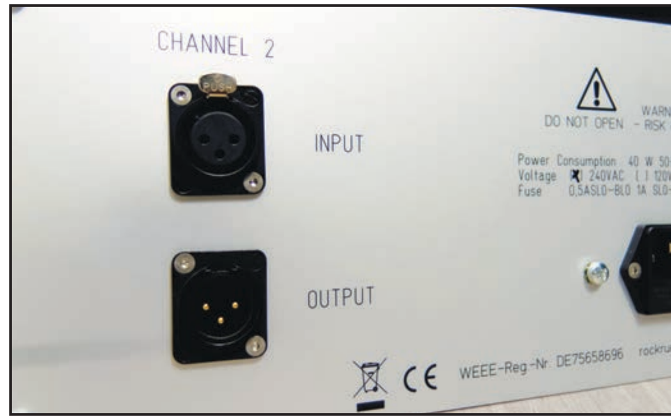
Another difference between the comp.one and the comp.two, resulting from that "simpler" approach, is that the new design costs less than the original, now-legacy unit. However, the Rockruepel comp.two is still an expensive unit (it costs $5,499).

Specifications

The beauty of this compressor design also lies in how few parts it contains. For each channel, there are six basic components on the audio path: two

The comp.two's front panel seems like a standard variable-μ compressor with input, output, threshold, and attack and release settings. This unit also features Hoyt volume unit (VU) meters.

The comp.two can work as a saturation or a distortion unit, bypassing the compression. It also has three side-chain filtered modes for removing lows on the detector circuit.



The back of the unit features the balanced connectors in XLR plus the International Electrotechnical Commission (IEC) power socket.

Resources

ALPS Electronic Co., Ltd., www.alps.com.

ELMA, www.elma.com.

Rockruepel, www.rockruepel.com.

Sowter Audio Transformers, www.sowter.co.uk.

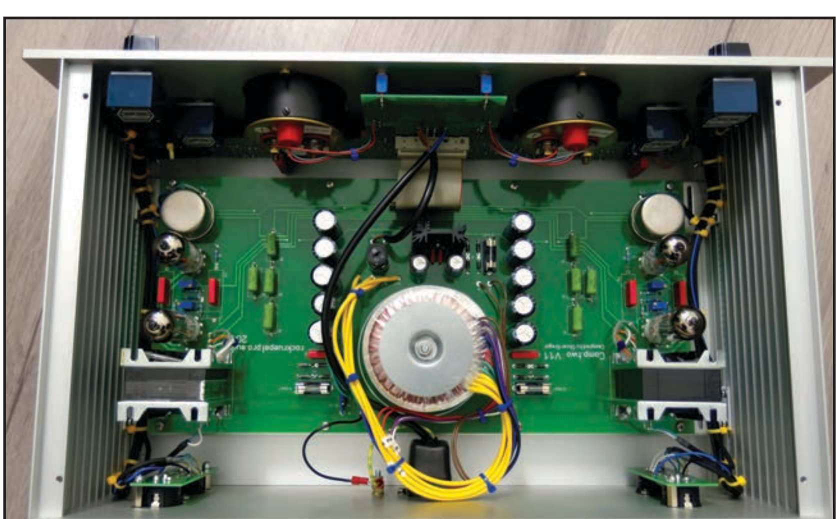
WIMA, www.wimausa.com

Source

6CG7 Preamplifier vacuum tube
Electro-Harmonix | www.ehx.com

capacitors, two transformers, and two tubes. What started as an Altec 436-inspired design has been stripped down to those six main components. This obviously makes the comp.two a totally different design from the Altec, therefore we will not even pretend to compare the units. However, we mentioned the Altec because the manufacturer referenced it when confirming that the idea for the comp.one/comp.two can be traced back to that design, but quickly diverged based on Gregor's own ideas.

Opening the unit for the first time, we noticed several top-quality parts. Rockruepel uses ALPS potentiometers for the I/O faders, WIMA capacitors, ELMA switches for attack and release times, and Sowter transformers. The comp.two contains two 6CG7 tubes and two 6N3P tubes. The first one is an Electro-Harmonix medium-micro twin triode tube, the later is a new old stock (NOS) Russian medium-gain dual-triode model.



The beauty of the comp.two is the simplicity of its design. The main audio path can be traced to six main components: two capacitors, two transformers, and two tubes.

Even though the comp.two only contains quality parts, Gregor states that this is only second to a great circuit design. Nonetheless, it is still impressive to see the quality parts in the comp.two. It all feels, looks, and is very high-end.

Looking at all the comp.two's specifications and considering this is an all-tube design this is a quiet, low-noise, and low distortion piece of equipment—unless we purposely saturate or distort the signal, obviously. Compared to a modern solid-state discrete design, most of the comp.two's specifications aren't that impressive. But as far as tube processors go, this is one of the cleanest units we've seen, assuming the unit's excellent specifications are not only due to the simplicity and the good design but also because of the high-quality parts used in it.

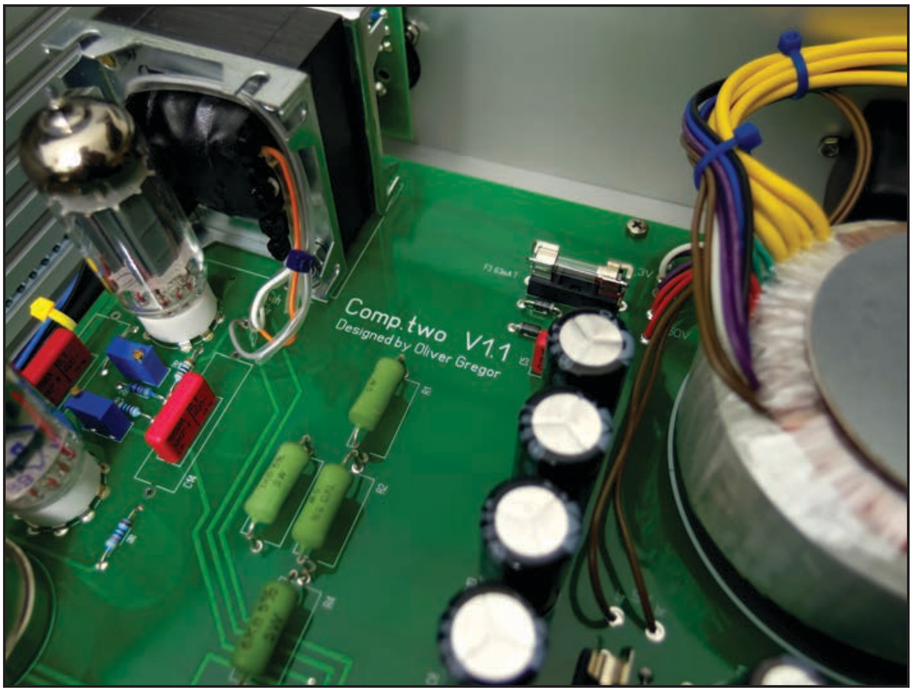
During our review we only had access to a German user manual. The English version is in the works and should be available when this review is published. In any case, it was nice to see the care the manufacturer took while measuring the unit and presenting its specifications in a very complete, detailed, and accurate form.

In Use

Initially, the comp.two may appear to be simple to use. Most parameters should be pretty straightforward to any professional user. There's an I/O level, a threshold, and a set of attack and release

Manufacturer Specifications

Frequency response: 20 Hz to 20 kHz (±1.25 dB)
Noise floor (10 Hz–30 kHz): –86 dB Q
THD+N (1 kHz at 4 dBU): –68 dB
Maximum I/O level: 24 dBu
Input impedance: ~17 kΩ
Output impedance: ~600 Ω



Rockruepel founder Oliver Gregor designed and hand built the comp.two.

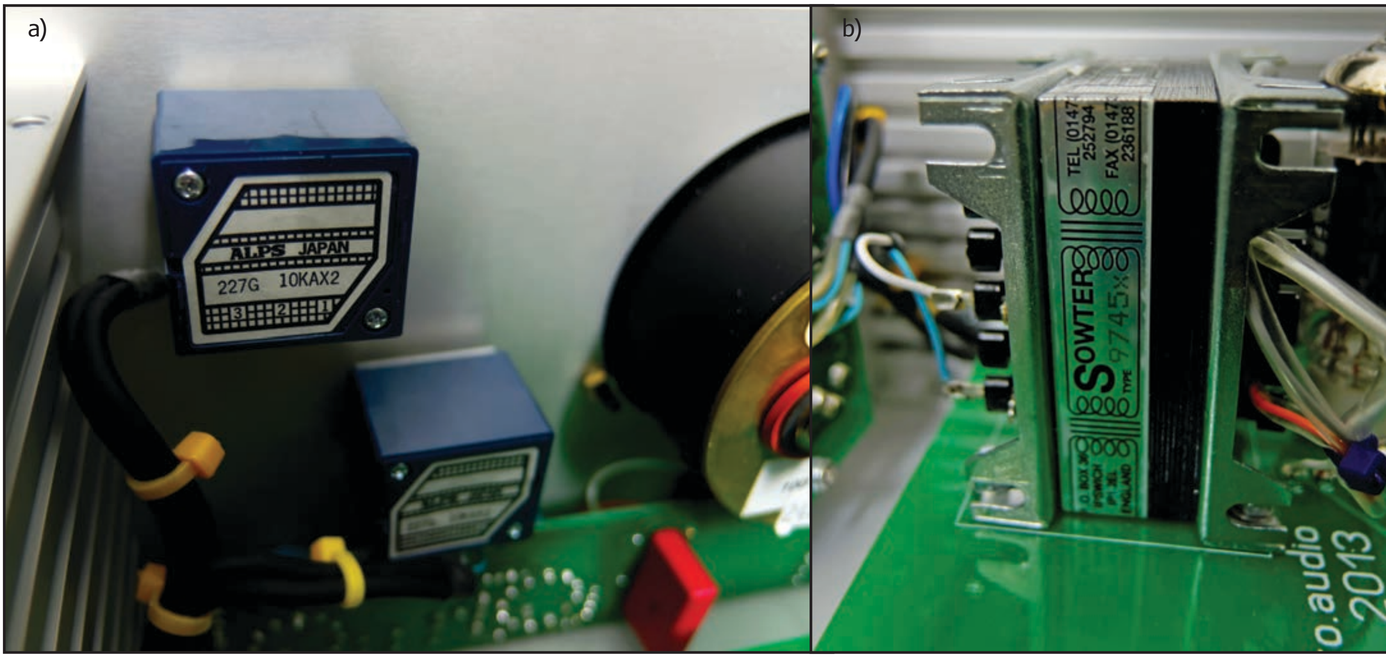
time constants. And there's a gain reduction meter to check how much compression is occurring at any moment. It seems fairly easy and similar to many other compressors on the market. But, we would be wrong to think that because there's a lot more to the comp.two than you may think.

First, we observed some interesting interactions between both time constants. For example, when increasing the release time there's a noticeable effect on the attack time that also seems to speed up a little. It's not exactly the same as changing the attack time manually, but it does change its behavior a little. These interactions between both time constants, in our experience, add a lot more versatility to a compressor because it can adapt to any kind of material. It can be as "grabby" and as fast as we may possibly need and it can also handle a soft gentle program leveling with medium to slowish times. This compressor can sound pretty aggressive, if using fast attack and release times. It can be faster than what most people think a variable-μ compressor could be. It can easily distort and pleasantly create "pumping" on an electronic dance music track, any drum bus, or music for that matter. Despite that, the comp.two shines by doing gentle program leveling and bus compression, such as that used in mastering and/or mixing.

We haven't found many compressors in the market that can be as smooth as the comp.two. Its action can be so transparent and delicate that we would have no problems using it in any type of recording and musical genre.

We were immediately sold on this compressor based on its action alone. Then we started experimenting with the comp.two's internal gain staging and a whole new world of sonic possibilities opened!

The unit instantly became even better. The comp.two can range from super clean and pristine sounding to almost insane amounts of distortion that sound so good you will be tempted to distort everything. We must confess that we're not huge fans of distortion, but the sounds you can get from the comp.two would make it worth buying just for those. The clever part about it is you can get any stage—from clean to saturated or even very distorted sound—because the unit uses I/O faders with no detents or steps. It makes complete sense to use them because it's in fine-tuning the comp.two's input and output levels that we can precisely match the desired sound for our source. If we want clean-sounding compression, we'll probably lower the input signal a lot and compensate with the threshold and output to get the same amount of gain reduction and output level. Or, we can achieve just the right amount



A few of the Rockruepel comp.two's premium components include the ALPS potentiometers for setting input and output levels (a) and two Sowter transformers on each channel. The larger transformer is shown (b).

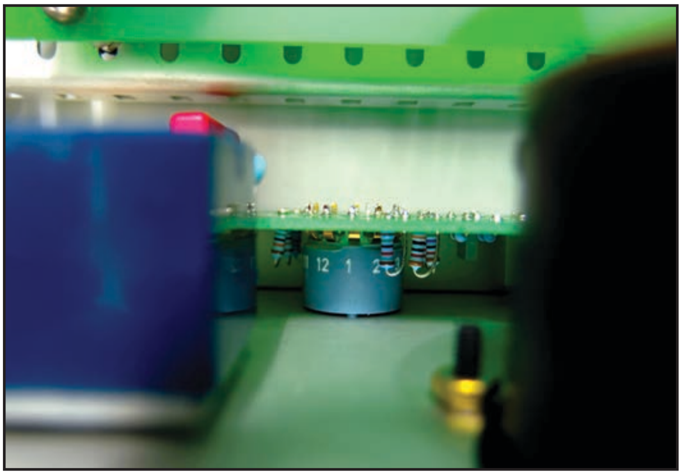
of saturation to make the entire midrange pop.

The comp.two's saturation is one of the most notable characteristics about this compressor. It can make anything sound larger than life, with or without compression, because we can use the comp.two in "amp mode," without ever making things sound "stuck" or saturated in a bad way. It's such a clean-sounding saturation that even the source's frequency response is not drastically changed.

We must reinforce the fact that we can make the comp.two to sound as pristine sounding as a tube compressor. Perhaps even cleaner sounding than what most people expect from an all-tube unit.

The truth is we've never come across a compressor with such a wide range of sounds. And all those "colors" from the comp.two are incredible. It's just a question of choosing the right one for the program or source. That's why having faders to set I/O levels is a great decision. A huge part of the comp.two's versatility and sound quality lies in correctly setting those parameters. That's why the comp.two is not a standard compressor.

With all that said, it's hard to find a particular application in which the comp.two would not excel. Want extreme distortion and pumping on a lead synth? It can do that. Or are you just looking for a variable-μ compressor that can provide the right amount of saturation and compression to make a pop vocal stand out in front of the mix? It also



High-end ELMA switches are used to configure the attack and release time constants.

does that. It's fairly easy to imagine using this unit for almost any stage of audio production, from recording a track to finishing a master. It really can do all that without imprinting the same sound over and over again.

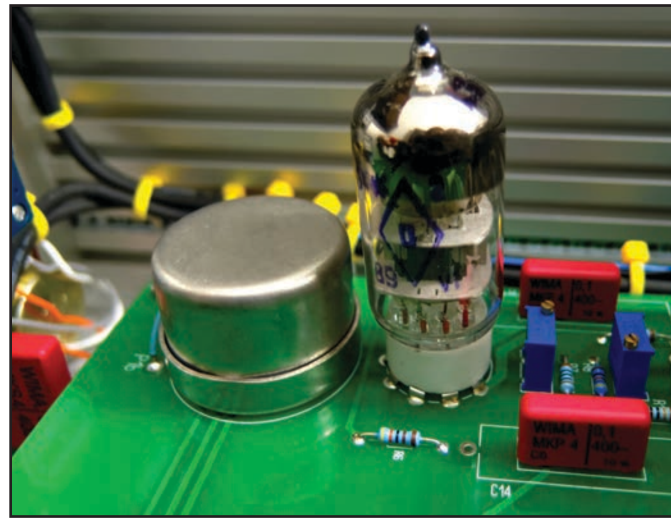
A League of Its Own

There are a lot of variable-µ compressors on the market and most of them are actually high-end pieces of gear. But the comp.two is, in our opinion, in a league of its own. It's true that this is a fairly expensive compressor. But after working with it for a couple of weeks in the studio, we believe that it's worth every dollar. It's one of the most versatile audio processors we've ever encountered. Especially when taking into account all it can do regarding compressing audio and also what it can do as a saturation/distortion unit.

We didn't find a single flaw with this unit. It's a design that's been refined to perfection. The attention to detail, the excellent build quality, and the use of quality parts turn the comp.two into one of the finest examples of a tube compressor. And the fact that it is so versatile while maintaining an excellent sound in every application makes it a worthy investment.

About the Author

Miguel Marques is a full-time mastering engineer who works in his own mastering studio in the north of Portugal. After earning a bachelor's degree in Music Production and Technologies, Miguel worked in commercial recording studios as a recording engineer. He has also written several technical reviews and articles for pro audio magazines. He recently finished his first book about audio engineering, which was released in Portuguese-speaking countries.



As an all-tube unit, the comp.two relies on one Electro-Harmonix 6CG7 vacuum tube and one new old stock (NOS) Russian-made 6N3P per channel.

We're certain anyone who purchases the comp.two will use it in every session without ever getting ever tired of how good it is. And that's probably the best compliment we could give to any compressor.

The Brave New World of Loudness Control (Part 2)

Ensuring Technical and Subjective Quality with Loudness Meters and Measurements

By
Jon Schorah
(NUGEN Audio)

With the introduction of loudness standards across the board for broadcast audio, as well as the adoption of some form of loudness normalization in many popular music formats including iTunes Radio and Spotify, now is a good time to revisit the practicalities of audio metering. This article discusses what to measure as well as how and why it should be measured.

There are two main reasons to meter audio—to ensure the audio's technical requirements and to establish the subjective quality. In general, technical requirements have gained the most attention as these are often unavoidably written into delivery specifications. Until now, few measures have addressed the more subjective side of things. However, loudness metering rather ingeniously covers both of these areas. It has a technical requirement (in true-peak) and a clear measurement of the subjective loudness based on a lot of research, which acknowledges that perceived loudness is a function of frequency and time (i.e., momentary, short-term, and program loudness).

Loudness Metering Approaches

Before we get into the specifics of loudness metering, let's review the various approaches available to audio engineers. These include the use of visual loudness meters, true-peak limiters, offline correction, and batch analysis.

Visual Loudness Meters—Clear, intuitive loudness metering enables the delivery of high-quality, loudness-compliant audio by helping engineers and editors to realize increased dynamic range and contrast within current loudness standards. The standards are based on a loudness scale designed to correspond to the human ear. Editors can use a visual meter to keep monitor the loudness profile while relying on their trained ears to make most of the decisions.

In the post-production environment, use of loudness metering software (e.g., plug-in or application) as opposed to a stand-alone real-time hardware meter can offer a significant advantage. The main target measure of integrated program loudness requires the loudness to be measured over the program's entire length. A software solution can best manage this requirement. A 30-s edit in a 50-min documentary demands that the full program be re-measured. Such an approach would cause something of a time issue if real-time meters were used. Many nonlinear editors (e.g., Avid Technology's ProTools and Media Composer) facilitate faster-than-real-time analysis, and various offline file-based tools also can speed analysis considerably to minimize downtime.

True-Peak Limiters—Loudness normalization deals with loudness jumps and consumer satisfaction issues, but engineers still need to measure peak levels to avoid distortion in the signal. Unlike sample peak limiters, a true-peak limiter can address the intersample true-peak requirement of today's standards. While maximum peak measurement is specified to prevent noticeable clipping and distortion, the way audio is sampled may not account for moments between samples when the real audio level may have gone above the maximum peak level. True-peak level is designed to measure that real level and avoid clipping.

It can be tempting to use sample peak limiters, but there are no “safe” settings for a sample peak limiter. A sample peak reading can often read as much as 6 dB higher on a true-peak meter, and it’s not possible to set a traditional brick-wall limiter (i.e., very high ratio and fast attack times) to be safe enough. A peak program meter (PPM) can be used to set artificial limits; however, this approach simply introduces another layer of complication. Despite this outcome, PPM values are still being identified in some delivery specifications, but the only way to measure correct true-peak values is to use a true-peak limiter.

Offline Correction and Batch Analysis—If the created mix is more or less loudness-compliant, then offline tools can be used to accelerate the final phase of the normalization process. Used within the editing environment, these tools can correct any true-peak overshoots and quickly bring a mix into line. Batch analysis brings valuable automated loudness processing into the post-production environment, enabling a facility to perform internal QA with files quickly assessed for compliance and then corrected or rejected as appropriate.

True-Peak Measurement in Practice

Technical Quality—The introduction of loudness standards requires a measurement of the true-peak maximum level of the audio. This replaces the older quasi peak program meter (QPPM) and sample peak measurement. It is specifically designed to avoid digital clipping by giving the peak audio’s accurate “true” measurement. A –2 or –1 dBTP maximum value is typical. This part of the standard enables transient peaks well above the average loudness level (and the legacy target of PPM6), thus introducing the possibility of a greater dynamic range and preserving audio fidelity. The true-peak maximum value ensures that distortion is not introduced into the audio downstream due to data conversion or lossy codecs such as MP3 and advanced audio coding (AAC).

It is important to note that audio is not normalized to dBTP. That value—dB relative to digital full scale, measured as true-peak—is simply the maximum allowable level permitted within the signal. As previously discussed, while the idea of using a favorite peak limiter to restrain any overshoots may be appealing, the fact that there is no direct relationship between sample peak and true-peak means that a ”safe enough” setting cannot be determined. Fortunately, a number of true-peak limiters can effectively handle the situation.

The true-peak measure adopted by Apple in the mastered for iTunes “afclip” differs very slightly from the accepted broadcast algorithm of ITU-R B.S. 1770-03, so editors working in this market should ensure their limiters can operate in both modes.

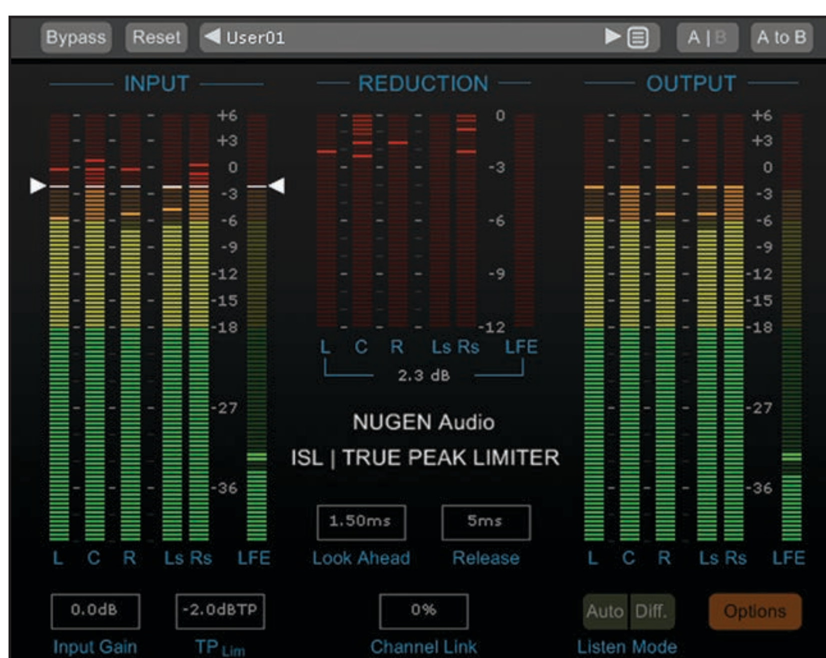
Of course, many other quality measurements can be used to ensure high-production value. For instance, mono compatibility is still an issue of paramount importance. Many of the top-selling digital audio broadcasting (DAB) radios only use one speaker. Extreme audio compression can employ mono-summing; and many TVs and MP3 player docks barely have enough speaker separation to be considered a stereo sound source. Though correlation, phase, signal-to-noise, and bit-depth issues are not covered by the loudness standards, they are commonly referred to in delivery specifications, making the tools capable of measuring these elements valuable to the audio engineer.

Subjective Quality—Audio’s subjective loudness is the area in which loudness measurements come into their own. For the first time, engineers have a set of measures that address audio in the same way the human ear does, taking frequency and duration into account. Subjective audio quality is determined by four main measures—momentary, short term, integrated program loudness, and LRA (loudness range). The most important of these measures is integrated program loudness.

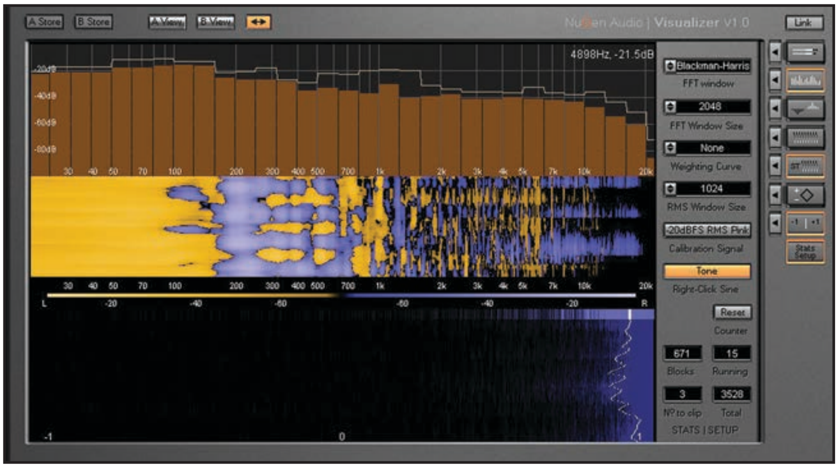
Program loudness is measured over the entire piece of audio, whether it is a 60-min documentary,

About the Author

Jon Schorah is the creative director and co-founder of NUGEN Audio, one of the world’s leading manufacturers of loudness products. Jon has a background in mastering and engineering and has considerable experience in wider aspects of the industry. A 1992 Leeds University (UK) graduate, in recent years Jon has focused on product design with a particular interest in the usability and workflow aspects of audio software.



The NUGEN Audio ISL True-Peak Limiter features B.S. 1770 and Apple afclip operation modes.



NUGEN Audio Visualizer audio analysis suite is used for quality audio production. It includes a standardized reference set of professional tools that enable users to work faster, avoid mistakes, repeat past successes, and understand others' success.

a 30-s advertisement, or a 3-min music track. The target value determined by the delivery document will almost certainly be –23 LUFS or –24 LKFS. LUFS (loudness units relative to full scale) and LKFS (loudness, K-weighted, relative to full scale) are, for all intents and purposes the same thing, but they are the result of differing European Broadcasting Union (EBU) and International Telecommunications Union (ITU) naming conventions.

So, what is the practical method for measuring an audio track's program loudness? Simply take the audio, start the meter (if it doesn't have an auto mode), and play in the entire piece. At the end, the meter will provide a single value for program loudness. However, if an edit is made, the difference between loudness normalization and peak normalization becomes critical. Previously, it was only necessary to ensure the edited section was within peak limits. Now, the entire spot must be re-measured, causing substantial workflow disruption for those only using a real-time meter!

However, in practice this situation is not as alarming as it may seem. Because the new measurements are based on how human ears respond to sound, an experienced engineer working in a calibrated room will usually find that he or she naturally mixes to ±2 loudness unit (LU), just using the meter occasionally for guidance. To facilitate the measurement of program loudness, many manufacturers also provide offline tools capable of running the measurement at many times real-time, thus yielding a result without waiting for the program's duration. Some of these tools can make corrections to bring the loudness on target with a button press. However, it is important to ensure that if these tools are employed they use an integrated true-peak limiter in case the audio is normalized upward.

NUGEN Audio LM-Correct's offline loudness analysis and correction enables users to create loudness-compliant audio.

Measurement in the Mixing Process

Various other loudness parameters are generally used to facilitate the mixing process. Momentary loudness and short-term loudness can be used to build up a loudness history profile, which provides the program content's loudness overview. This can be useful in identifying whether particular sections would benefit from further attention or in adjusting the program loudness further toward a target instead of using a global offset.

For short-form content, program loudness is often used in conjunction with either short-term or momentary loudness, in which case a target maximum value for one of these parameters will also be specified in the delivery document. In this situation, it's helpful if the meter used can also note these maxima—a capability that enables a quick check for compliance at the end of the analysis run.

The loudness range (LRA) parameter can be used as an indicator of the dynamic content of longer-form audio. (The statistical measure is not appropriate for audio that is less than a minute long.) While this parameter is not usually specified, some delivery specifications recommend a maximum LRA value for TV broadcast, which means some light compression may be required if the material is particularly dynamic.

Other subjective loudness issues (e.g., dialog clarity and background noise/music level) are not directly addressed by the standards. However, it is not difficult to envisage simple checks that ensure, for instance, that the dialog level is clearly differentiated "x" LU above background beds in a more complex bused metering environment.

Effect on the Mix

How do these loudness measurement standards and practices affect the way engineers mix? For many in broadcast, work will progress pretty much as usual. Those used to mixing with a healthy dynamic range are probably already mixing close to target. Now, there is simply a little more freedom to openly mix and increased headroom that will enable them to trust their ears. For those working in the advertising and music sectors (in which Apple loudness normalization "sound check" is the default in iTunes radio) there is an opportunity to move away from pushing into the limiter to achieve a louder result.

The commercial imperatives that drove engineers and editors to be as loud as everyone else have been eliminated. In fact, a heavily compressed mix will sound rather flat and dull compared with a dynamic mix normalized to the same loudness level. Try a few tests and the effect will be evident!

The NUGEN Audiuo VisLM loudness meter shows S_{MAX}–M_{MAX} toggles and history color bands to highlight loudness boundaries.

Source

Pro Tools audio production platform and Media Composer
Avid Technology, Inc. | www.avid.com

Weird Science Woofers

By
Mike Klasco and Steve Tatarunis
(United States)

Last month, we explored super-sized woofers, all larger than 18″ diameter. Yet true innovation is more than just scaling size, it is finding new solutions. The moving voice coil woofer has undergone few changes since its development more than 70 years ago. Aside from advances in materials and construction techniques, today's woofers still possess the same functional limitations as those from the 1950s. This month, we will discuss several unique mechanisms—not simply huge or ultra-shallow conventional speaker configurations.

Photo 1: Tom Danley, who worked with NASA hardware contractor Intersonics from 1979 to 1996, was awarded 17 patents for a variety of inventions, among them the ServoDrive subwoofer and several acoustic and electromagnetic levitation devices.

The first group of woofers we want to discuss use various types of rotary movement electric motors. The second group utilizes technology that works on a conventional voice coil/magnetic topology but with a different diaphragm configuration and form factor.

Thinking out of the box can be a fascinating engineering adventure, yet finding a different path does not always mean it is a better way. Consider the French Citroen car, which took quite a distinct approach to many engineering, ergonomic, and aesthetic aspects in the 1970s. In spite of these differences, the Citroen brought little lasting contributions to good automotive design. In this article, we'll explore "weird science" woofers, each quite different from the typical woofer designs we've discussed so far. In most cases, there is increased complexity, but each woofer provides performance characteristics that just may meet a specific application's requirement.

Regardless of the drive mechanism or diaphragm, the equipment must adhere to the laws of physics. One way or another, the speaker's backwave is going to be out of phase with the front output and they'll need to be separated or there will be some unwanted cancellation of total output. Some of these transducers are optimized for closed boxes, others are optimized for vented, transmission lines and/or horn loading.

ServoDrive

The operating principle in "ServoDrive" substitutes the traditional voice coil/magnet "motor" with an actual high-speed rotary servo-motor and belt-drive designed to move the woofer cone. This concept was created and commercialized by Thomas Danley. The servo motor is driven by an amplifier and follows the audio signal input. This approach requires very fast and agile responses as this determines the woofer's transient response and operating frequency range.

The rotary servo drive promised low harmonic a)

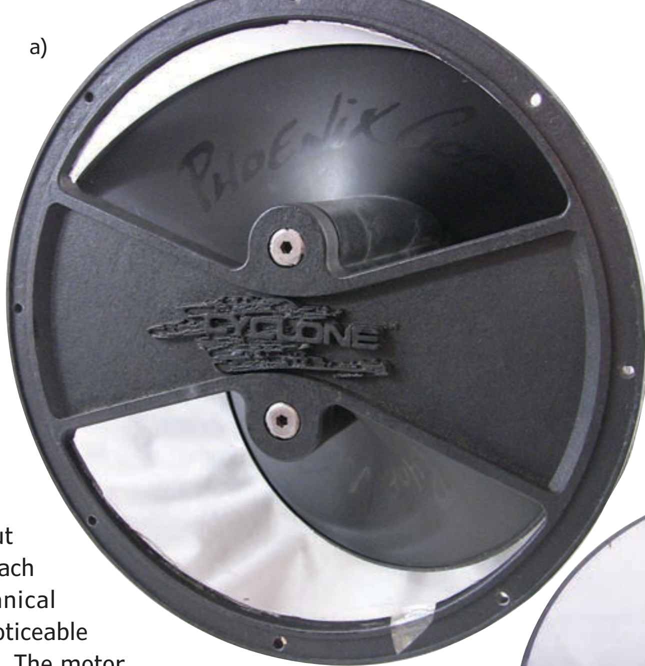


Photo 2: The Phoenix Gold Cyclone is a discontinued "rotary motion, servo motor" type subwoofer. Rather than a voice coil motor, which pushes a cone in and out in a linear motion, this driver uses a servo motor to twist a bisected panel back and forth via it's center shaft. The shaft has a back-and-forth rotating motion, hence the term "rotary." Here it is shown in a front (a) and rear view (b).

b)

distortion, minimal output loss due to power compression, deep extended frequency response, and high efficiency. The motor-driven belt design boasts linear excursion and efficiency and handles large low-frequency peaks without damage. However, this approach was also known for mechanical and motor noise that was noticeable at low acoustic output levels. The motor and mechanism-induced noise floor did not seem to be an issue for touring sound where these bass systems gained some acceptance. The system also requires belt replacements from time to time (see **Photo 1**).

Danley has developed subwoofers (most, but not all, variants of the ServoDrive) at various business entities starting with Intersonics in the 1980s, which then sold its subwoofer group to Quantum Sound (1994). Danley later founded Sound Physics, and he now designs and manufactures products under the Danley Sound Labs brand.

Phoenix Gold Cyclone

The Phoenix Gold Cyclone provides a somewhat similar yet different take on the rotary motion servo motor type subwoofer. Danley developed the design for Phoenix Gold. According to Danley, "My goal in this invention was to eliminate the problems, losses, and nonlinearity associated with the edge suspension and spider in a system where a large displacement in a small physical size was needed."

The Cyclone received a lot of press in the autosound aftermarket in 2003. Like the ServoDrive, it utilized a moving magnet rotary motor. Unlike the ServoDrive, it used a rotary radiator diaphragm instead of a traditional piston radiator. This transducer converted low-frequency signals into acoustical output without some of the physical limitations of conventional voice coils, cones, spiders, and surrounds. This innovation resulted in a subwoofer with low distortion and three times the displacement of a conventional 12" woofer. Additional refinements resulted in the Cyclone producing a lot less mechanical "self-noise" than the earlier ServoDrive designs (see **Photo 2**).

The Cyclone worked best in a sealed box from 2 to 2.5 ft^3 as the tight back volume/small enclosure size reflected the Cyclone's intended installation in a car truck. Its recommended bass complement was to be crossed over to dedicated 6.5" to 8" mid-bass drivers.

The way in which the Cyclone interacts and blends with the mid-bass drivers was considered to be the key to its sound quality. A steep (minimum 18 dB per octave) crossover no higher than 60 Hz between the mid-bass drivers and the Cyclone was suggested. The idea was to make it impossible to tell where the Cyclone was physically located. The only way to do this was to keep the frequencies that can be localized (above 50 to 60 Hz) out of the Cyclone. Sensitivity (90 dB at 1 W/1 m) was a bit low but it is typical of extended bass/limited back chamber sealed enclosures. The manufacturer claimed an 11-to-60-Hz frequency response.

The Cyclone only needed about 150 W continuous (300 peak) to achieve full excursion at 20 Hz. However, one major issue was that it sounded decent at increasing levels without a noticeable increase in distortion or other level-related distress—until it would suddenly fail. One Cyclone failure mode is the stem of the "flag pole" driving mechanism would fracture at elevated levels. Another issue with the Cyclone was that its motor housing was made from molded plastic and over time, the powerful neodymium magnets inside it caused the housing to deform, which resulted in rubbing inside the drive motor.

A beefed-up, commercial/industrial version of the Cyclone was developed for active noise cancellation (ANC) at an electrical power plant in Redondo Beach, CA. This Cyclone-on-steroids

a)

b)

Photo 3: Developed by Bruce Thigpen and available from Eminent Technology, the Thigpen Rotary Woofer Model 17 (TRW-17) breaks entirely new ground at the very bottom of the sonic spectrum. It is shown here in a front (a) and a rear view (b). (Photo courtesy of Eminent Technology)

had the same displacement as six 18″ subwoofers at 1″ peak-to-peak excursions (roughly ±350 in^3)! These were extremely rugged devices as they had to operate continuously at more or less full output. Despite the demanding specifications, the devices were a diminutive 16″ × 12″ × 8″ driven by a 10″ long × 5″ diameter motor.

Eminent Technology's Thigpen Rotary Woofer

Eminent Technology's Thigpen Rotary Woofer Model 17 (TRW-17) is another type of rotary woofer. However, the ServoDrive, Cyclone, and the TRW-17 provide different takes on the same core concept. The TRW-17's rotary fan operates on different acoustic rules than a conventional woofer. With conventional woofers, the speaker cone's displacement must increase four times for each halving of frequency to maintain the same acoustic output. This is why conventional cone woofer companies have developed "long throw" woofers adding more and more amplifier power. These types of subwoofers tend to be inefficient.

This rotary-fan subsonic woofer is designed to reproduce sounds below 20 Hz, which is below the normal human hearing range. When installed in a sealed room's wall, they can produce infrasonic frequencies down to 0 Hz, a static pressure differential, by compressing the air in the sealed room. A few years back at an Audio Engineering Society (AES) Convention, Eminent Technology demonstrated its fan subwoofer in one of the nearby hotel rooms.

A motor controller and electric motor rotate a set of blades at a constant speed. The TRW-17's pitch mechanism uses a conventional voice coil and magnet assembly. This is connected to the amplifier to pitch blades in proportion to the applied audio signal. As the blades move, a pressure wave is generated. The degree of pitch controls the pressure wave's amplitude. Air transitions through the blades. By oscillating the pitch of the blades, sound is created as they rotate (see **Photo 3**).

This same variable pitch concept is used in modern propeller planes (e.g., the De Havilland Dash 8). The purpose of varying the pitch angle with a propeller is to maintain an optimal angle of attack (maximum lift to drag ratio) on the propeller blades as aircraft speed varies. This constant-speed propeller enables the pilot to select a rotational speed for maximum engine power or maximum efficiency. A propeller governor acts as a closed-loop controller to vary propeller pitch angles as required to maintain the selected engine speed.

Like an aircraft, the TRW-17 uses a constant rotational speed motor—so just AC (or DC) power is doing the heavy lifting. The amplifier only needs to modulate the pitch of the fan blades analogous to the audio signal. Since the audio amplifier only changes the pitch of the blades, it takes much less audio amplifier power per decibel of generated acoustic sound level to drive a rotary woofer than to power a conventional subwoofer.

The blades swing both positive and negative, with respect to a zero-pitch spinning blade position. The voice coil's motion is used to change the angle of a fixed rotation speed set of fan blades to generate sound pressure waves. The pitch of the blades change according to the signal the amplifier supplies, producing a modulated sound wave due to the air moved by the spinning blades. If no signal is applied, the blades simply rotate "flat" at zero pitch, producing no sound.

Distortion increases as the input signal's frequency exceeds the fan's rotation rate, which is itself limited by the need to avoid making higher frequency noise (due to the sound of the spinning blades). Current models use an AC induction motor spinning at 800 rpm (13 Hz). The woofer is installed and carefully braced so the blades lie in a circular opening. This is so air can be moved between an external chamber (e.g., house's attic) and the main listening space. If the rotary woofer were not installed in such a "baffle" and instead placed directly in the main space, the generated sound pressure level (SPL) into the listening area would be much less.

Tymphany's Linear Array Transducer (LAT)

In 2004, Tymphany's Linear Array Transducer (LAT) was audio headline news. The LAT "motor" is conventional with a voice coil in a magnetic system. However, from the coil onward was all new. The patented LAT is designed to provide strong bass performance in a vibration-free, cylindrical form factor. The driver may be configured for use in sealed, vented bandpass, transmission-line, or open-baffle enclosures, and it provides high bass output from a very compact transducer. In typical applications, mechanical vibration is more than 90% lower than comparable loudspeakers, even at full output.

The LAT employs dual voice coils in opposing directions so the mechanical vibration cancels out. That, and the LAT's shallow form factor made it ideal for in-wall designs. It was also an ideal approach for docking stations, flat-screen TVs, and other applications in which the enclosure has lightweight construction. Unfortunately, the flat-screen market is so focused on screen size and price the only audio solutions that have made inroads are the aftermarket soundbars, which retailers often bundled with the TV. Autosound OEM is also a tough business with car manufacturers offering very little return for innovation—so the LAT had limited commercial success.

The LAT's timing to the market has also impacted its success and acceptance. It was introduced in late 2004, followed by a couple of years of manufacturing refinement and development programs with auto manufacturers. In 2008, the recession greatly impacted the auto manufacturers. The Alpine PLT-5 LAT made quite an impact, but the autosound aftermarket's decline and already began.

This "high-density" loudspeaker technology promises to deliver the bass of much larger speakers in devices occupying only one-third the space in a tubular form-factor. LAT products do this through a linear array of multiple smaller diaphragms, generating sound through multiple flow ports along the side of the loudspeaker housing. The Tymphany LAT conforms to the physics of loudspeakers, but changes the conventional shape and method of how air is moved. LAT technology is scalable down to 2" and up to 12" in diameter at any length. It can be stacked into large clusters for even more power and it is compatible with existing electroacoustic driver design tools (see **Photo 4**).

The core concept was first conceived by Oskar Heil best known for his air motion transformer. He was granted patents in 1977 (US Patent 4,039,044) and 1978 (US Patent 4,107,479). These patents expired some time ago, but an example of this design can be found in the Electro-Static Sound (ESS) Transar (see **Photo 5**). ESS is back and exhibited at the 2014 International CES in January. During the show, ESS displayed a Transar, which will be shipping this year.

Photo 4: Tymphany's LAT-700 contains the power of two 12" subwoofers in a sleek tube-shaped transducer that is only 7" wide. (Photo courtesy of Tymphany Corp.)

The Tymphany team spent years revising and refining the LAT design and added innovations for reliable, stable operation and practical production assembly. The increased speaker parts and assembly alignment complexity were challenges that were at least partially solved after a couple of years.

Today, Tymphany's Peerless conventional high performance speaker product line is the company's main focus, along with its integrated amplifier/speaker OEM and ODM business. The LAT products nevertheless represent viable technology which at one point in time made quite a splash. Perhaps we have not yet seen the end of this technology.

Next Up—Bass Shakers

Beyond subwoofer bass lays infrasonic energy. Some of the subwoofers mentioned in this article deliver the audible portion of extreme deep bass, but little of the structural bass vibration component of live sound at a concert. Next month, we will discuss bass vibration devices, aka "bass shakers."

Photo 5: The original speaker designs from Electro-Static Sound (ESS) were a hybrid of conventional woofers, passive radiators, and electrostatic tweeters in bookshelf and tower configurations. ESS was one of the only manufacturers of the patented air motion transformer (AMT) speakers designed by Oskar Heil. At the end of the 1970s, ESS and Heil introduced the ESS Transar, which used one AMT for the high frequencies and a special mid-low frequency driver unit.

Predictive Acoustics

By
Richard Honeycutt
(United States)

In ancient times, the importance of acoustics in theaters was appreciated; indeed, first-century Roman architect Marcus Vitruvius Pollio discussed acoustics in Book 5 of his "Ten Books on Architecture." In 1853, Dr. J. B. Upham of Boston, MA, undertook a careful study of the acoustics of the Boston Music Hall, culminating in an essay discussing reverberation and resonance. American physicist Joseph Henry presented papers to the American Association for the Advancement of Science in 1854 and 1856, and a new lecture room in the Smithsonian Institution was built according to the results of his experiments and theory. The results were said to be highly satisfactory.

Architect T. Roger Smith published his book *Acoustics of Public Buildings* in 1861, in which he discussed the differing acoustical requirements of speech and music. John Tyndall and Lord Rayleigh (John Strutt) investigated the control of reverberation in rooms. But the definitive work that marks the birth of scientific acoustics was that of Harvard University Physics Professor Wallace Clement Sabine. In the final years of the 19th century, Sabine was tasked with remedying the acoustics of Harvard's Fogg Lecture Hall. He began with a quantitative study of the nature of reverberation and the physical factors controlling it. His equation for predicting the reverberation time of a room is still widely used:

$$RT = \frac{CV}{S\alpha}$$

where RT is the reverberation time of the room, C is a constant whose value depends upon the system of units, V is the volume of the room in cubic feet or cubic meters, S is the surface area of the room boundaries, and α is the average acoustical absorption of the boundary surfaces.

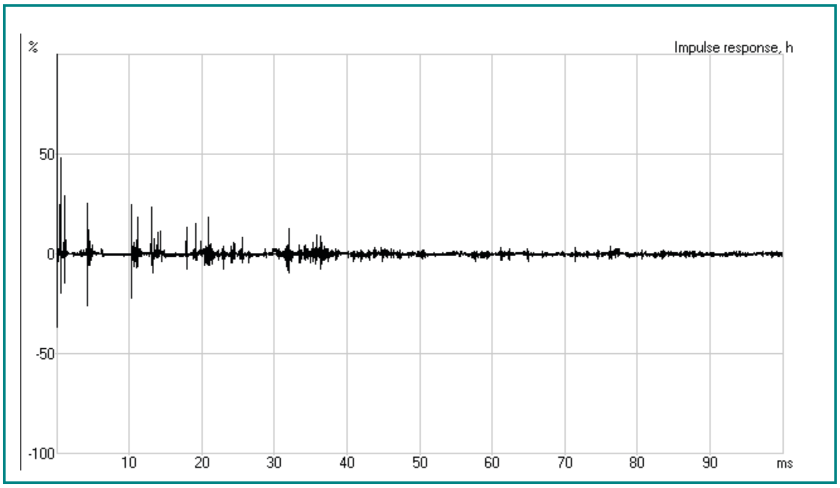


Figure 1: This impulse response shows how sound decays with time.

Reverberation Time

Sabine's equation is based on a statistical approach involving the mean free path of a sound wave between reflections and the average acoustical absorption of the room boundaries. It works well for calculating RT in rooms that meet the necessary conditions for statistical analysis: not too high an average absorption (less than 10%), a diffuse sound field, and more-or-less uniform distribution of absorption across all surfaces. Pencil-and-paper predictions using the Sabine equation were considered state of the art for many years.

With the introduction of digital computers, large institutions could greatly decrease the time it took to conduct the many calculations needed to predict RT in a real venue. PCs and affordable spreadsheet programs brought this capability to acoustical consultants and architects. However, many rooms do not meet the criteria for successful prediction using a statistical approach.

In an occupied auditorium, most of the absorption is provided by the occupants, and thus, is located on the floor, with substantially less absorption on the walls and ceiling. Deep balconies and insufficient diffusing features prevent a room from having a truly diffuse sound field.

For some very high-budget projects, scale models were built and frequency-scaled acoustic inputs were applied using miniature microphones to measure the results. However, the scale-model technique has its limitations. While wave characteristics of sound can be scaled to match the scale-model room, absorption by surfaces and air is not easily scaled. It is a tribute to acousticians of the time that so many good venues were designed in spite of design tool limitations.

Impulse Response

In 1968, Asbjørn Krokstad, Svein Strøm, and Svein Sørsdal published a paper entitled, "Calculating Room Acoustical Response by Use of a Ray Tracing Technique." This paper described a computerized method for calculating a room's impulse response (IR) using rays emitted in random directions from a source. The paths of the rays were traced mathematically using the Law of Reflection, with the strength of each reflection modified by the absorption coefficient of the surface from which the ray was reflected. From the IR, not only the RT, but many other acoustical factors could be determined.

A room's IR could be described as the instantaneous sound pressure in the room, plotted as a function of time, resulting from a very brief burst of sound (the impulse). Creating a theoretically perfect IR would require an impulse whose duration is close to zero. However, an impulse that is very short compared to the RT of the room gives a satisfactory approximation. **Figure 1** shows the first 100 ms of the impulse response of a medium-sized church sanctuary.

Once a room's IR is found, its RT can be predicted. Other acoustical characteristics of the room can also be predicted from the IR, including early decay time (the RT that would result if sound decayed at the same rate as it does in the first 10 ms): early-to-late energy ratios such as D50 (definition) and C80 (clarity); acoustical strength G; bass ratio (ratio of RT at low frequencies to RT at mid frequencies); treble ratio (ratio of RT at high frequencies to RT at mid frequencies); interaural correlation coefficient (IACC), which relates to perceived spaciousness; lateral energy fraction (ratio of early energy coming from a listener's sides to total energy, which also relates to spaciousness); and stage support (ratio of energy reflected from a stage enclosure to total energy). If sound in real rooms behaved according to the Sabine statistical concepts, the IR would be the same everywhere in the room. However, in actual cases, the IR varies from place to place. Using computer acoustical modeling software to predict the IR in specific places in a room enables us to see the values of RT and all these other characteristics at any desired place in the room.

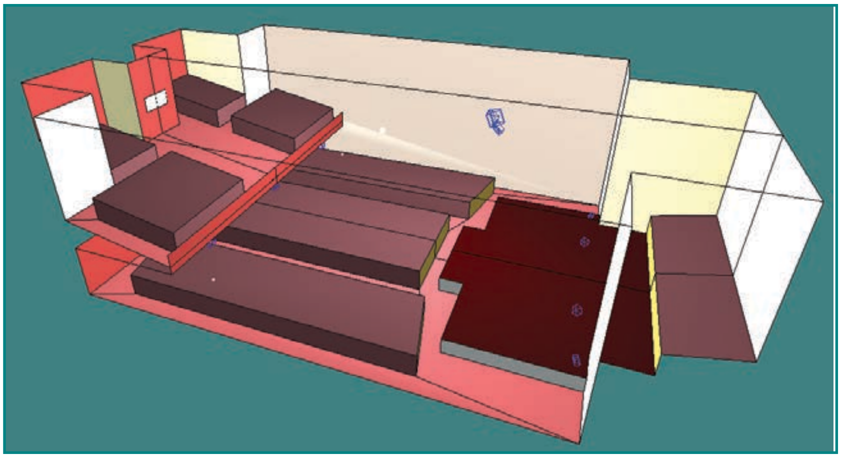


Figure 2: This rendering of a church sanctuary was created by a well-known acoustical modeling program.

Ray Tracing

In 1980, Charles Hurst of Virginia Tech presented an early paper on acoustical ray tracing to the North Carolina Regional Acoustical Society of America (ASA) chapter. This paper discussed successful predictions of a room's acoustical properties using Fortran software called RAYTR, written at Virginia Tech. Hurst and Bruce Held, a master's degree student, did most of the work. A complete prediction, including

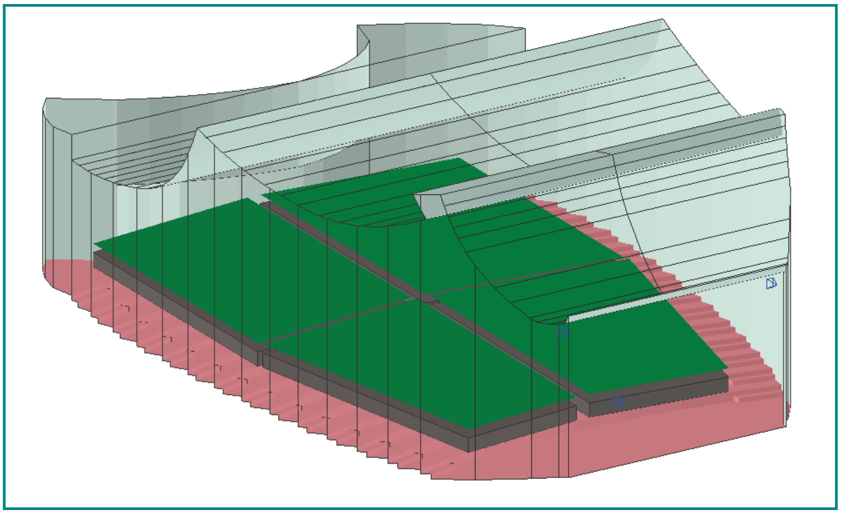
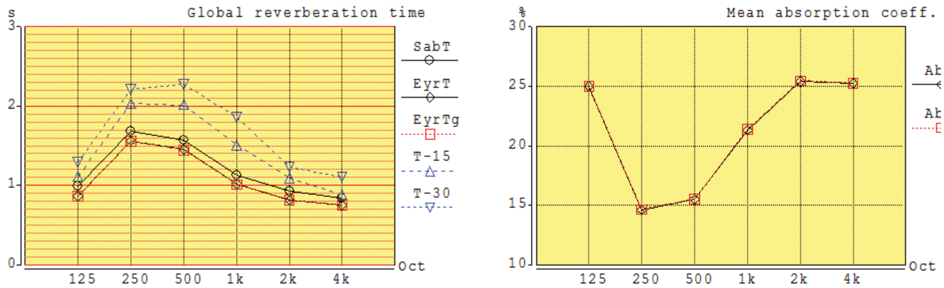


Figure 3: Another well-known modeling program produced this rendering of an auditorium.



	125	250	500	1k	2k	4k	
EyrT	0.87	1.56	1.45	1.01	0.81	0.75	s
EyrTg	0.87	1.56	1.45	1.01	0.81	0.75	s
SabT	1.00	1.68	1.57	1.13	0.93	0.85	s
T-15	1.11	2.03	2.02	1.50	1.09	0.88	s
T-30	1.30	2.21	2.27	1.86	1.23	1.10	s
AbsC	25.01	14.63	15.48	21.33	25.42	25.24	%
AbsCg	24.99	14.64	15.54	21.41	25.47	25.28	%
MFP	6.22	6.21	6.21	6.21	6.21	6.21	m
Diffs	12.16	14.06	19.04	23.96	28.97	32.05	%

Trunc	3000.0	ms
Rays	39896	(used/oct)
	0	(lost/oct)
	0	(absorbed/oct)
Angle	1.02	degrees

Figure 4: The reverberation time (RT) panel displays many different types of information.

up to third-order reflections, required several minutes to run on an IBM mainframe computer.

Since that time, personal computer speed and capacity have improved by huge margins, and a number of commercial architectural acoustical modeling programs have been released: notably, Modeler, EASE, CATT-Acoustic, Odeon, and Ulysses. All of these programs use ray-tracing algorithms to provide a plethora of acoustical predictions for the space being modeled. In addition to all the characteristics already mentioned, the algorithms can predict if and where echoes will be a problem. They also provide speech intelligibility using several different systems, and direct and total sound coverage for specific models, locations, and aiming of speakers.

Auralization

In the last two decades, the calculated IR has been put to another use: auralization. By convolving (mathematically modifying) an anechoically recorded wave file with a room's IR, one can create a very good simulation of that room's sound. Using binaural processing and good headphones, directional cues can be incorporated as well, making for a very lifelike representation of how the original sound source would sound in the specific listener location for which the IR was created. Auralization is very useful in acoustical design of listening spaces. It is also beneficial in virtual reality programs used for gaming and military training.

Modeling Programs Differ

Each acoustical modeling program has advantages and disadvantages, as well as characteristics and quirks that cause a given user to prefer one over the other. These differences include various ways of inputting the dimensions and physical characteristics of the venue, variations in the manner that the output data is presented, and special functions offered in some programs, such as walk-through auralization, which presents a visual rendering of the venue as though you were walking through it, coordinated with auralized sound corresponding to the part of the venue where the visualization shows you to be located. Although the basic ray-tracing algorithms of various programs are somewhat similar, there are details—especially those affecting auralization—that differ. **Figure 2** and **Figure 3** show examples of projects rendered by two well-known acoustical modeling programs.

Obtaining reliable results from a modeling program requires an understanding of acoustics, experience with the specific software, and an appreciation of certain basics of modeling. For example, to the uninitiated it may appear that the more detail one can include in the model, the more accurate the results. According to this line of thought, modeling stair steps shown in **Figure 3** would be better than replacing them with an inclined plane, as shown **Figure 2**.

The problem with this thinking is that sound diffracts around objects that are small compared to a wavelength and ray tracing does not capture this behavior. Thus, a sound wave's actual behavior is better simulated by the inclined plane than by stair steps that will initiate a lot of ray-scattering in the simulation, which may not actually occur in the real venue.

Figure 4 and **Figure 5** show different ways of presenting RT variation with frequency in two different modeling programs. The panel shown in **Figure 4** displays statistically calculated RT using the Sabine and the Eyring equations, with RT determined by ray tracing using the T-15 and T-30 methods, average surface absorption vs frequency, and mean free path as determined by ray tracing. The panel in **Figure 5** shows the coordinates of the location for which the RT was determined, as well as overall maximum, average, and minimum RT values. Other data is available via tabs. Statistically calculated RT values are shown in this program on a separate page. These figures are only included to illustrate some of the differences among acoustical modeling programs—differences that may affect different users in different ways.

Wave Phenomena

Sound is actually not a ray phenomenon. It is a wave phenomenon. This means that sound waves do not travel in simple straight lines, as approximated by ray-tracing algorithms. Three important wave phenomena that are not accounted for by ray tracing are refraction, diffraction, and seat dip effect.

Refraction is the bending of waves due to a change in the air's acoustical properties, as can result from temperature gradients in a large gymnasium or outdoor venue. It can cause unexpected variations in the direct sound coverage from speakers: hot spots and dead spots in the audience.

Diffraction is bending of waves around obstacles. An example is the way that sound shadows do not occur behind small pillars, at least for low and mid frequencies. A pure ray-trace would show such shadows in the coverage pattern, but you will not be able to detect the predicted dead spots by listening or by a sound-level meter. Another example is the way that sound hugs a curved sidewall rather than skimming past it or ricocheting off it.

Seat dip effect is attenuation in the audience area (occupied or unoccupied) resulting in a dip in sound pressure level between 100 and 300 Hz, beginning at the first row. It is caused by constructive and destructive interference between the direct sound and the sound reflected from the floor and seating between rows of seat backs. The severity of the dip depends on the angle between the plane containing the seat tops and the direct sound. The effect is worst at small grazing angles and is reduced as the direct-sound angle becomes steeper.

All three of these wave phenomena can be important in specific cases, and in these cases ray-tracing modeling programs may not give accurate results. The developers of these programs are aware of these ray tracing shortcomings. An improvement in these programs that has become almost ubiquitous in the last 10 to 15 years is the ability to apply scattering data.

A basic ray-tracing program assumes that when a sound wave strikes an object, all the energy is either reflected in a predictable specular way (like light), or absorbed. Depending on the surface roughness and the scale or size of the surface features compared to the wavelength of sound for which the space is being analyzed, a substantial proportion of the sound may be scattered; that is to say, specularly reflected in a way that is not practically predictable. The scattered

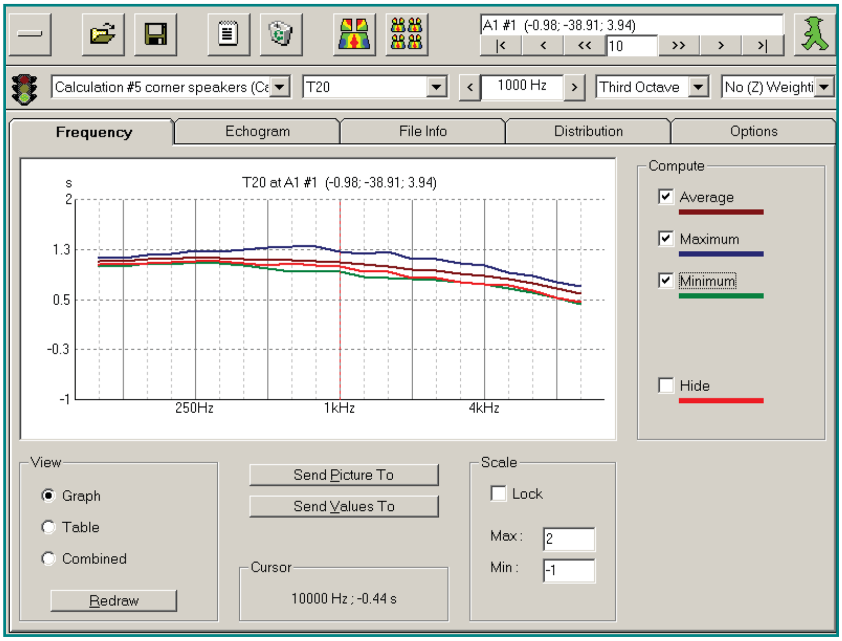


Figure 5: This RT panel shows information in a different format, which is accessible via tabs.

sound will contribute to the reverberant field but will not create echoes. Newer versions of modeling programs that apply scattering data can incorporate these features in their predictions.

Computer Modeling

An even more recent advancement in ray-tracing modeling programs is that the ability to calculate diffraction effects is now available in some of the programs. Coupled with scattering calculations, diffraction handling brings ray tracing a step closer to the exact results that would be available from a program that would rigorously solve the wave equation in the space. Still, the computational efficiency and ease of data input provided by ray-tracing programs is retained.

An acoustician who is new to computer modeling and wants to purchase modeling software can be a little overwhelmed unless (s)he has some understanding of the differences among different models. With modeling programs, the main differences are in the method of data input, the ability to incorporate scattering and diffraction, the method of presenting data output, acoustical parameters calculated by the program, the availability of manufacturers' speaker data in a form acceptable to the software, the inclusion of walk-through auralizations, and the accuracy of auralizations.

One other aspect is the expectations of the acoustician's professional partners: Is there a certain format required for models so that they can be passed back and forth among the acoustician, sound contractor, and perhaps the architect? Carefully studying the manuals of the various programs and trying out demonstration versions helps you make an informed decision.

Engineer Takes a Chance on Start-Up Audio Venture

By
Shannon Becker
(United States)

An Interview with Morten Sissener of Tortuga Audio

Morten Sissener used his engineering knowledge to open his own audio design boutique, which he named Tortuga Audio.

SHANNON BECKER: Tell us about your company Tortuga Audio. Can you also share the story behind your sea turtle mascot?

MORTEN SISSENER: Tortuga Audio is a boutique audio design, manufacturing, and marketing company located in South Florida. At present we only sell through the Internet via our website although we expect to add channel partners in the future probably starting in Europe. We're a little over a year old in terms of coming out with our first products—a line of passive preamplifiers built around light-dependent resistors (LDRs). We actually manufacture here in the US. While it may not hold much sway with customers, there's something satisfying in being able to say "Made in the USA." That's the short and mostly dry of it.

Our mascot is the sea turtle wearing a set of headphones. People who know me wouldn't describe me as being particularly religious or spiritual, but I've come to accept the sea turtle as my totem. A totem is an object or symbol representing an animal or plant that serves as an emblem of an individual, family, or tribe. You don't pick your totem. Your totem picks you.

I've always had a fascination with a place called The Dry Tortugas. It's a small cluster of islands about 70 miles west of Key West, FL. I first heard of this place in an adventure book I read when I was a young boy. I fell in love with reading, warm blue water, and The Dry Tortugas.

Years later, I bought a boat in South Florida. The first time I took that boat offshore I went out 20 miles, stopped, and turned off the engines. I was out of sight of land. The water was glassy calm. Thirty feet off the starboard beam a sea turtle surfaced and stared at me. I stared back. This lasted for over a minute. Then the turtle dove away. The name I'd already put across the transom of my boat was The "Tortuga Dreamer." So when it came time to name my audio company you could say it named itself—Tortuga Audio.

SHANNON: What prompted you to start a company that designs and manufactures audio equipment, particularly in 2010 when the economy was so uncertain?

MORTEN: Starting an audio company that caters to a shrinking niche of audio fanatics as the masses continued shifting to low-resolution MP3 audio and inexpensive ear buds was arguably a questionable business decision. However, it was definitely a passionate business decision. Approaching 60, I figured it was time to pursue my passion rather than my resume. And I've always been passionate about technology, audio, and music. You could say the rational engineer decided to follow his heart.

a)

b)

The Tortuga LDR6 passive preamplifier's front (a) and back view (b) are shown. As with all Tortuga Audio's LDR passive preamplifier (LDR*x*) products, the LDR6 has unity gain passive (no active amplification) volume controllers that employ digitally controlled audio grade light-dependent analog resistors to provide neutral and transparent attenuation.

SHANNON: What kind of audio products do you build? Can you share some of your design challenges?

MORTEN: While the business rationale behind the founding of Tortuga Audio may not win first prize in any business plan contest, the decision to pursue our flagship product was, and remains, highly rational and compelling.

Back in 2009, I was building a tube preamplifier mostly as an exercise to see if tubes could really offer more than solid state. I was less than impressed with the results even though I'd used a well recognized kit/design and top-of-the-line components.

At the time I was using a motorized Alps Blue Velvet potentiometer for volume control. Thinking that perhaps I could improve the sound by going to a stepped attenuator, I stumbled across LDRs. I cobbled together a very basic LDR volume control based on bits and snippets of information on the Internet, pulled out the Alps potentiometer and installed the LDR. The result was nothing less than a revelation!

The fact that this tube preamplifier went from disappointing to awesome simply by changing the attenuator made quite an impression on me and frankly I couldn't stop thinking about it. It's what engineers do. Especially this engineer. I can't stop thinking of ways to do things differently or better. I've always hated that old saying, "if it ain't broke, don't fix it." Can you imagine Steve Jobs saying that?

This led me to ask a simple question. Why do I need a preamplifier? Why preamplify and then amplify? Do I really need the additional gain? In most instances, the answer is no.

I looked at all the complexity of that tube preamplifier and decided to pull out the LDR attenuator, set the preamplifier aside, and use the LDR as a purely passive volume controller. The result? It sounded even better without the tube preamplifier. And not just a little bit better, a lot better! I was so impressed with this LDR attenuator that I couldn't leave it alone. From that point forward, I dove into the deep end of LDR volume control.

SHANNON: What makes your audio equipment unique?

MORTEN: LDRs are challenging to work with because they are both nonlinear and variable. Nonlinear means their relationship between control signal applied to an LDR and the resulting resistance level is not a simple fixed ratio. Variable means that this nonlinear relationship can vary from one LDR to the next even with LDRs of the same model, from the same manufacturer, and from the same production run. That's a lot of variable nonlinear stuff and that makes it very hard to get consistent predictable behavior when using LDRs for volume control. No designer likes to work with audio components that behave like LDRs.

Part of our solution to taming the wild LDR was to design a programmable digital control unit that enables us to control the analog LDR with proprietary software algorithms. We combine digital control with a two-step testing protocol such that each LDR preamplifier we build has a custom set of software-based correction curves that ensures predictable performance. This is neither simple nor easy, but we've put an enormous amount of time and effort into developing the software and hardware tools to do this cost effectively.

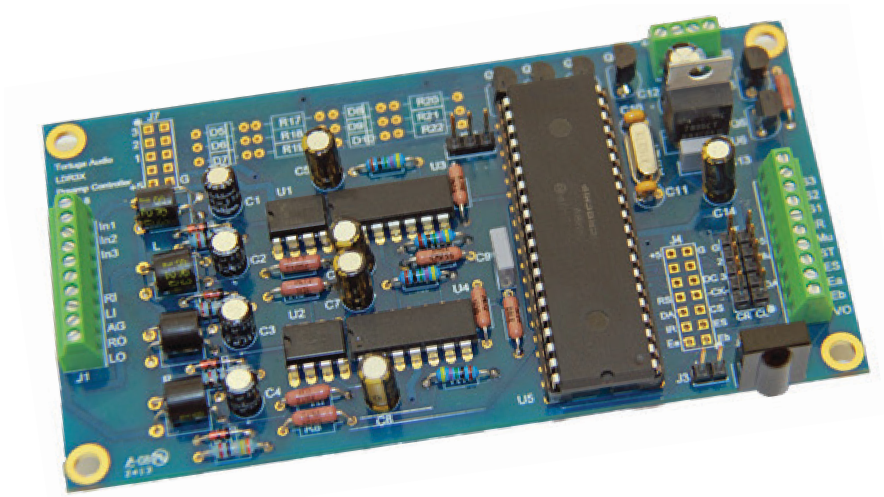
The result is a unique and game-changing LDR-based passive preamplifier (volume control) design that we believe rivals not only all other passive preamplifiers out there but also meets or beats even the best high-end active preamplifiers. While I happen to believe this personally, feedback from our customers and reviewers continues to reinforce this view.

SHANNON: Are you currently planning or working on any new product designs?

MORTEN: Our core focus continues to be advancing the development of our LDR-based volume controller products. In the third quarter of 2012, we came out with our LDR1 and LDR6 passive preamplifiers, which are finished preamplifier products. In the third quarter of 2013, we introduced the LDR3*x* passive preamplifier controller board (the LDR3*x*), which we marketed to the DIY audio community. We plan on continuing to serve the high-end audiophile consumers with finished products and provide DIY products to audio enthusiasts who'd rather build it themselves.

In November of 2013, we introduced the HiZ upgrade to our LDR-based preamplifiers. The HiZ algorithm enabled us to raise the input impedance of our LDR*x* products resulting in a remarkable improvement to an already fantastic-sounding preamplifier/volume controller. As far as we know, nobody else has done anything like this.

In terms of what's next, we are working hard on coming out with our new line of LDR*x* passive preamplifiers including our new LDR3B, which I believe may be the first-ever LDR-based preamplifier for balanced audio. We hope to release the LDR3B before the end of March. Since we are a relatively low-volume business and want to offer distinctive products that are not priced out of reach to most audiophiles, we've decided to manufacture our own enclosures in-house going forward. This will keep our costs down while enabling us to offer high-quality products and still retain the flexibility of small-batch production, quick design changes, and the ability to offer custom solutions.



The Tortuga LDR3x is a preamplifier controller board designed around LDRs that enables DIYers to build a passive or active preamplifier including remote control.

Beyond our next generation line of LDR preamplifiers, we plan to introduce a buffer companion product to our passive preamplifiers that will expand the application of our preamplifier/volume controller to include sources and amplifiers where a pure passive may not be the best fit. We are also considering the introduction of an integrated amplifier product that will allow us to target a broader market. These will be second half of 2014 products.

Longer term, we are quite excited about the prospects for an OEM version of our LDR preamplifier controller product. Every active preamplifier or integrated amplifier sold and marketed to the audiophile community that currently uses a potentiometer for volume control would sound better with a Tortuga Audio LDR volume controller. And along with being the best-sounding attenuator available, it also includes input switching, IR remote control, and a built-in encoder control.

SHANNON: How did you become interested in audio electronics?

MORTEN: As a newly minted mechanical engineer, I started my professional career in the aerospace sector working with complex electromechanical systems. This segued into energy when the company I was with in California became interested in alternative methods of power production. This eventually led me into industrial construction, large capital project development, project finance, software, sales and marketing, wind, solar and biofuels, as well as several start-ups along the way. An interesting ride but all along I was remained very interested in technology, software development, audio, and music. I decided it was time for my true interests to rule the day rather than the inertia of my resume. Plus I'm just an unapologetic techno-geek with a big creative itch that needed scratching. I also like to run my own show.

SHANNON: Where do you see the audio industry in 10 years?

MORTEN: I believe the high-end audiophile market with many components costing $10,00 or more is going to continue to decline into obscurity. Many have argued that the high-end market may already be in a terminal death spiral of rising prices and shrinking volume. I tend to agree. If true, that's not a sustainable scenario for high-end audio.

The audio listening paradigm of a big-rig stereo in the living room that the aging baby boomer audiophiles were introduced to in the 1970s is not the central paradigm of contemporary audio. Where only a few years ago you could go into a big box store and see racks of receivers, rows of speakers and even a "high end" listening room, today, most of that is simply gone.

Ironically, we are collectively listening to more music from more sources more of the time than ever before. Access to music is wide if not deep. The Internet has become the new radio. Online streaming is becoming the norm for most consumers while buying and owning music is slowly retreating, This is especially true for physical media such as CDs.

Despite this bounty of access, we've also experienced the concurrent dumbing down of audio quality (e.g., low-resolution MP3 files) and listening through lo-fi hardware, most of which has gone mobile. So it's an interesting mixed bag of good and bad news for us audio nutters.

While the road ahead may be unclear, I believe that a significant percentage of all those 20-to-30-something Millenials and Xers are eventually going to raise the bar on their audio game as they grow older and their incomes rise. But you can forget living rooms filled with big, heavy, and expensive gear as the norm. "Personal audio" will continue to grow and evolve and that means computer centric audio.

For most, that will mean DAC->preamplifier (volume controller)->amplifier->speaker configuration in which the DAC/preamplifier/amplifier separates will trend toward being a single integrated component. Speakers will be smaller, but higher quality near-field units usually located on desks or bookshelves near where people sit and work with their computers. And yes, no doubt a subset of these folks will eventually go with some bigger gear as well. But I believe we're talking a few thousand dollars of audio gear and not tens of thousands of dollars.

SHANNON: Do you have any advice for *audioXpress* readers who want to build their own sound systems?

MORTEN: I tend to be a minimalist and a skeptic and try to not get distracted by bright shiny objects. What I recommend is forget cables, power conditioners, cryogenics, and ceramic outlet face plates. Focus first on what matters the most.

Nothing will affect your audio enjoyment as much as speakers. Poor-quality speakers can make a great rig sound awful. Great speakers can make a low-quality rig sound remarkably good but not great. My personal favorite these days are full-range speakers with alnico magnets. Full-range speakers are point sources with no crossovers or phase-shifting. They offer amazing clarity, articulation, and bass.

Second (self-serving statement alert!) get the best preamplifier/volume control you can. Every note gets squeezed through the bottleneck of your volume control and this is where the most irreparable harm happens to your audio signal, even if everything else you have is really good.

Third, choose the best DAC you can. DACs are evolving rapidly, which is fantastic news.

The last thing I would worry about in terms of main components is your amplifier. Not that amplifiers don't matter, they just don't matter that much compared to everything else. And the good news is there's a huge selection of great amplifiers out there.

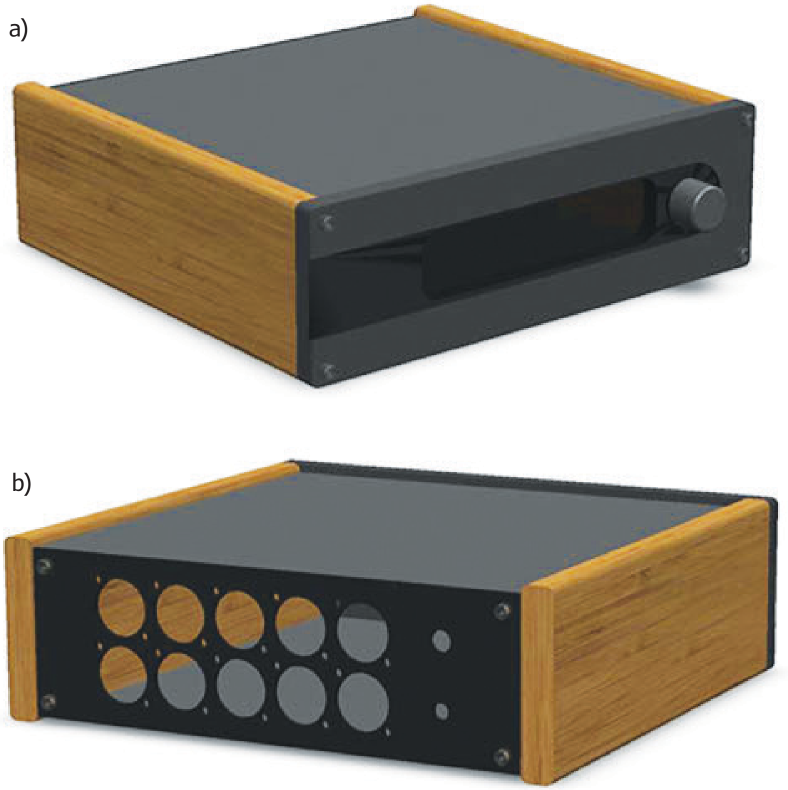
To summarize, if you're deciding on how to prioritize your money, make it speaker->preamplifier (volume control)->DAC->amplifier. Of course, if you're into vinyl then a good turntable and cartridge is critical, but don't forget the phono preamplifier. This can get expensive fast. I'd expect to spend a few thousand dollars to get into the land of great vinyl audio. It will cost more for fantastic.

Once you've got a decent system you really enjoy listening to, you can begin the madness of tweaking this and that, trying various cables, power conditioners, and so forth. But remember that the purveyors of audio equipment will tell and sell you practically anything you can imagine to get that extra ounce of goodness out of your rig.

Despite all the changes happening in technology and the audio industry, music remains a wonderful art form and audio is still a great hobby. Enjoy!

a)

b)



This 3-D CAD rendering shows the front (a) and the back (b) view of a prototype enclosure design for Tortuga's new LDR3B balanced passive preamplifier, a new product line that will be coming out in March.

The MC100

A High-Quality Moving Coil RIAA Preamplifier

By
George Ntanavaras
(Greece)

Moving coil cartridges are frequently preferred by audiophiles due to a subjectively better performance. But designing a high-quality preamplifier for this type of cartridge is challenging. Their output voltage is usually more than an order lower than the moving magnet cartridges and their source impedance is much lower and completely different.

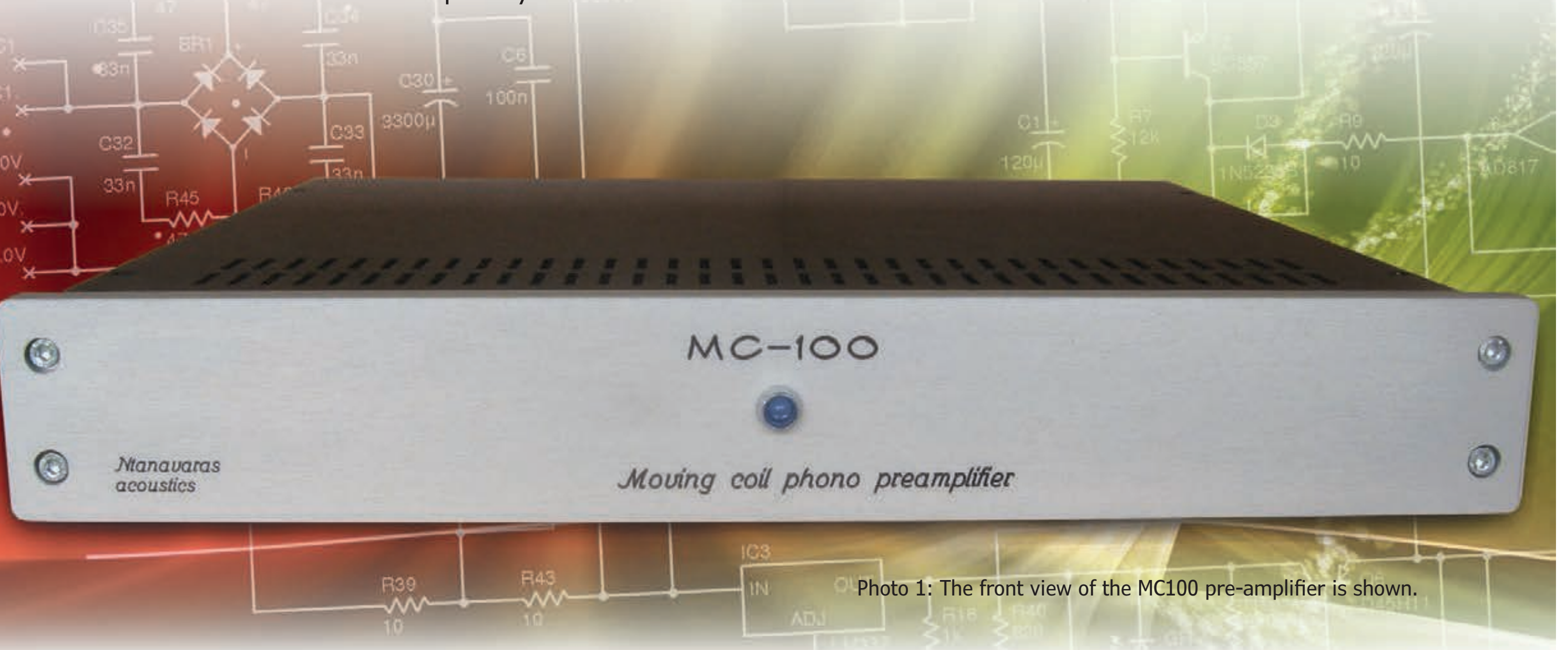


Photo 1: The front view of the MC100 pre-amplifier is shown.

About the Author

George Ntanavaras graduated from the National Technical University, Athens, Greece, in 1986 with a degree in Electronic Engineering. He currently works in the Development Department for a Greek electronics company. He is interested in the design of preamplifiers, active crossovers, power amplifiers, and most loudspeakers. He also enjoys listening to classical music.

In this article, I will describe the design and construction of a high-quality RIAA preamplifier for moving coil cartridges (see **Photo 1**). In designing the preamplifier, I determined my main objectives. First, I wanted the amplifier to be based on high-quality op-amps designed for low-noise audio applications and driven from very low source impedance.

A high-quality low-noise preamplifier requires a high-quality power supply, so I used Walter Jung's regulated power supply as described in his article "Improved Positive/Negative Regulators" (*Audio Electronics,* 2000).

I wanted all the components of the preamplifier and its power supply (except the transformer) to be placed on a single PCB for easiness of the construction. The preamplifier would include an adjustable gain to compensate for the differences a cartridge has between the two channels and optionally a volume control so it could be used to directly drive a power amplifier.

Preamplifier Design

The most important decision for the preamplifier's design was my op-amp selection. After some investigation, I selected Analog Device's AD797, which is an expensive single op-amp with bipolar input transistors biased at high collector currents. Considering its ultra low voltage noise (0.9 nV/√Hz) and low total harmonic distortion (–120 dB) in audio bandwidth, the AD797 is an excellent choice for a moving coil preamplifier. This preamplifier is especially useful for DIY projects in which quality is the first priority and cost is secondary. Due to

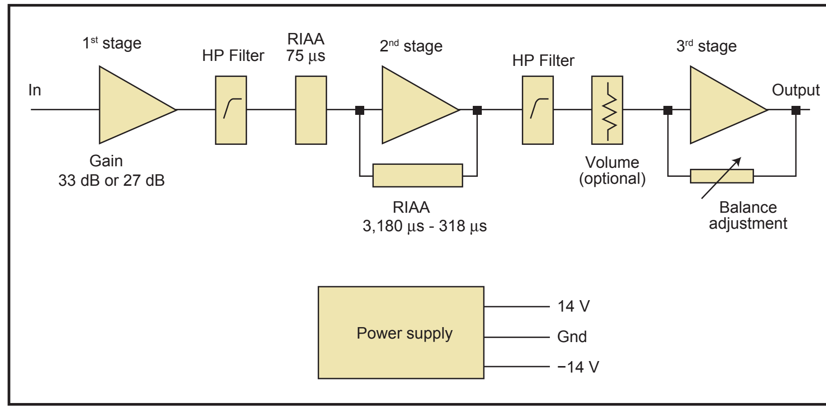


Figure 1: The MC100 preamplifier block diagram includes the preamplifier circuit and the power supply circuit.

its high quality, I decided to use this op-amp in all stages of the preamplifier.

The Preamplifier Circuit

The preamplifier's simplified block diagram includes the preamplifier circuit and the power supply circuit, which is based on the Jung regulator (see **Figure 1**).

The preamplifier circuit consists of three stages. The first stage buffers the cartridge and provides two different gains depending on the cartridge's nominal output voltage. The first high-pass filter comes next and the RIAA equalization time constant at 75 µs follows. These are buffered by the second stage, which also provides the 318-µs and 3,180-µs time constants of the RIAA de-emphasis curve. The third stage buffers the second high-pass filer and the optional volume control, which provides an additional gain between 5 and 7 dB. This can be adjusted with a trimmer to compensate for any possible balance errors between the left and right channels of the moving coil cartridge output. These balance errors are typical for all the cartridges, and in my opinion, should be accurately adjusted.

Figure 2 shows the amplifier's complete electronic schematic. The schematic only shows

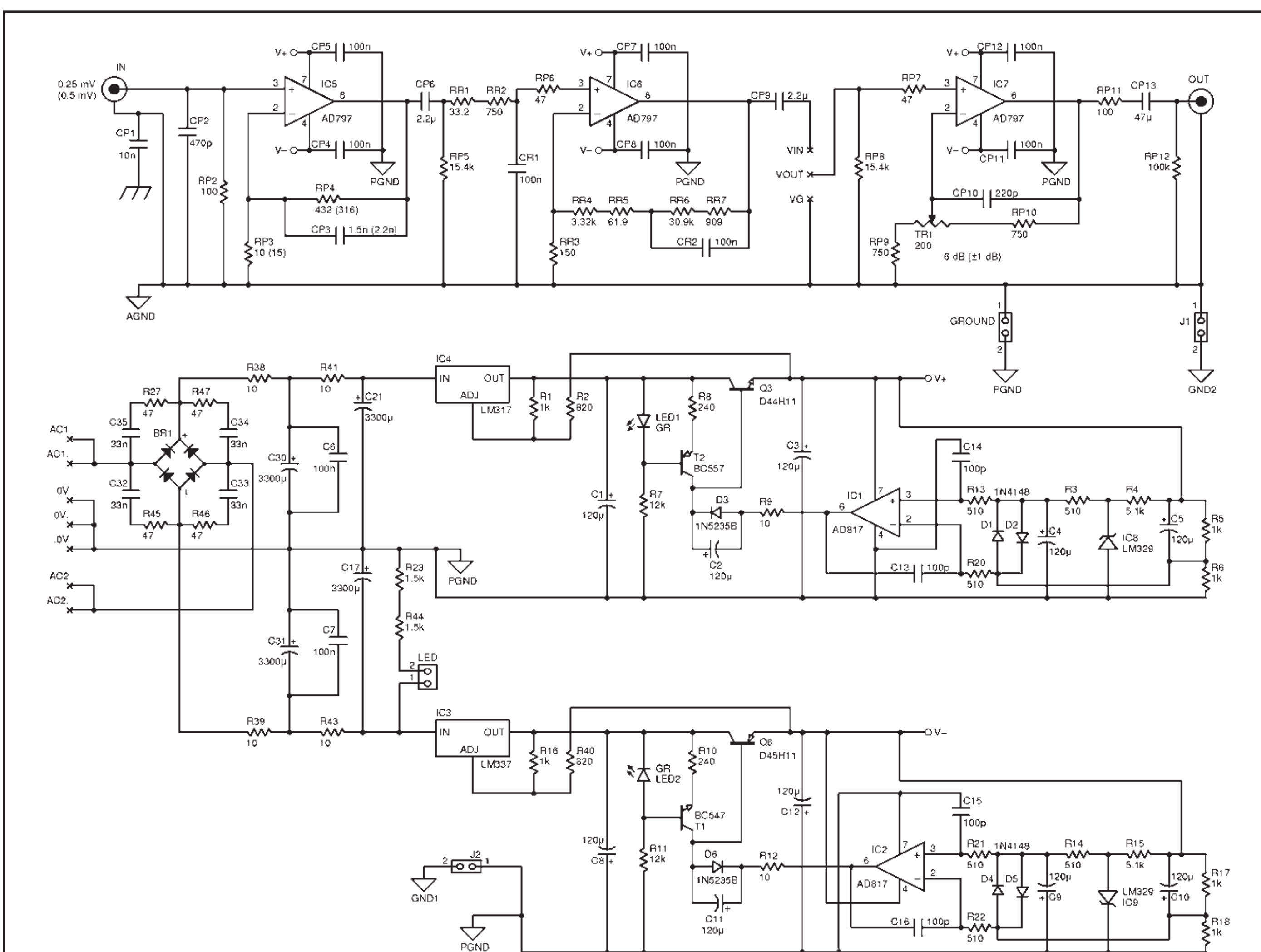


Figure 2: This is the MC100 preamplifier electronic diagram.

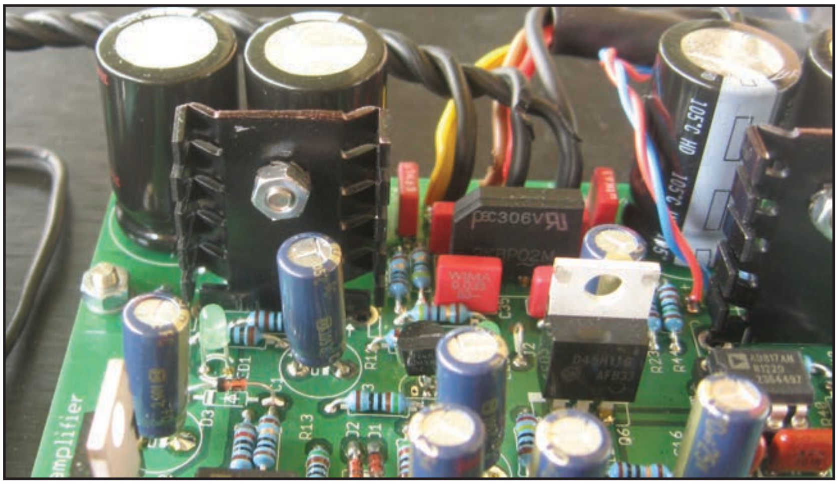


Photo 2: I used a small Fischer Elektronick heatsink on each regulator.

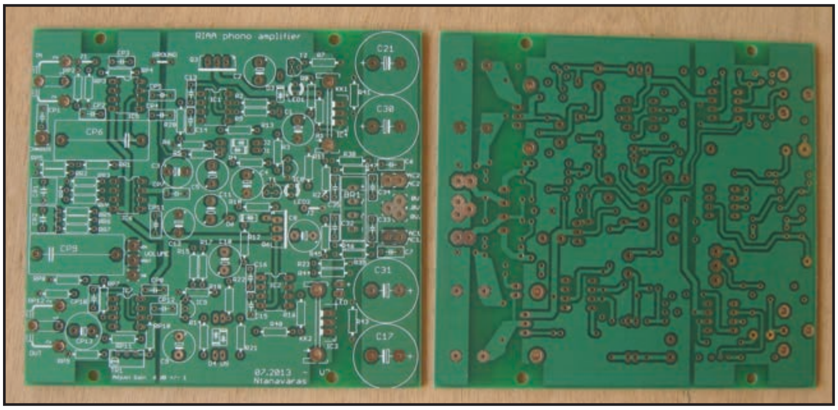


Photo 3: The design shows two PCBs for the preamplifier.

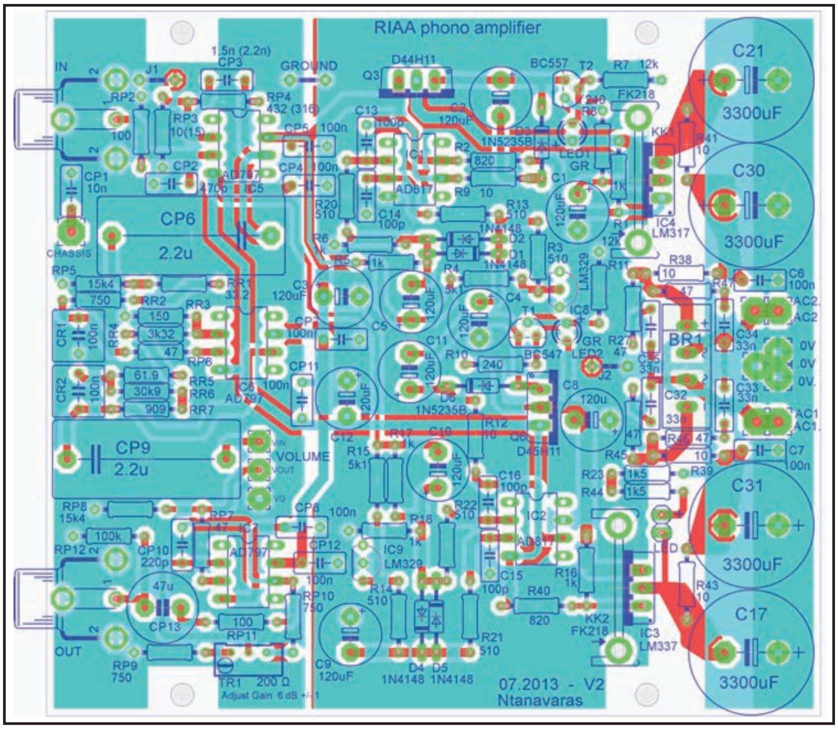


Figure 3: The MC100 preamp PCB layout is shown.

one channel; however, the other channel is identical. Capacitor CP1 is used to bypass the input ground to chassis. This is needed to suppress any radio frequency (RF) from entering the preamplifier through the input cable's shielding. Capacitor CP1 should have a good response in the RF spectrum, so I used a 10-nF, 50-V ceramic COG capacitor.

Resistor RP2 and capacitor CP2 adjust the load impedance the preamplifier presents to the moving coil cartridge. They should have the cartridge manufacturer's recommended values. In the schematic, I used 100 Ω, which is the typical value.

The first stage is a non-inverting amplifier based on the AD797. This stage's gain is defined by the values of the resistors RP3 ad RP4. I tried to keep the resistor RP3's value as small as possible for minimum noise. In the schematic, I indicated two possible values for the resistors RP3 and RP4.

For a cartridge with a 0.25-mV nominal output, the higher gain should be used with the resistor values RP4 = 432 Ω and RP3 = 10 Ω. These values will give a gain of times 44.2 (32.9 dB).

If the cartridge has a 0.5-mV nominal output, then the resistors should be RP4 = 316 Ω and RP3 = 15 Ω. This will provide a lower gain of times 22 (26.9 dB). Capacitor CP3's value is also adjusted according to RP4's value so the high-frequency response's rolloff will be constant at about 240 kHz for both gain values.

CP6 and RP5 form the preamplifier's first high-pass filter at about 5 Hz. This high-pass filter is required for two reasons. First, you want to prevent any possible offset of the first stage entering to the next stage, which has a very large DC gain. Second, you need to reject the very low frequencies from the turntable, which may overload the woofer.

Resistors RR1 and RR2 and capacitor CR1 form the RIAA equalization first-time constant at 75 μs (2,122 Hz). I used high-accuracy resistors with 0.1% tolerance and a polypropylene capacitor with a 1% tolerance to minimize the circuit's deviations from the RIAA curve.

The total value of the resistors RR1 and RR2 is somehow larger than the nominal value of 750 Ω because in the PSpice simulations I performed, the high-pass filter of the CP6 and RP5 affects the equalization's time-constant value and the resistor RR1 has to be added to compensate for this.

The next op-amp stage is also based on the AD797 and forms the RIAA equalization's time constants at 3,180 and 318 μs with the resistors RR3, RR4, RR5, RR6, and RR7 and the capacitor CR2. Here, I also used high-accuracy resistors with 0.1% tolerance and a polypropylene capacitor with a 1% tolerance to minimize the RIAA curve's deviations as much as possible.

Capacitor CP9 and resistor RP8 form the preamplifier's second high-pass filter, which is required for the same reasons as the first high-pass filter. Then, I included three pins to connect the optional volume control potentiometer. Any high-quality potentiometer can be used here. However, the total resistor of the potentiometer value in parallel with resistor RP8 should be 15.4-kΩ. If not, the capacitor CP9 should be modified accordingly to keep the low-pass filter around 5 Hz. If volume control is not required, a short circuit between pins VIN and VOUT should be used.

The final stage is again based on the AD797 and provides an additional 6-dB nominal gain, which can be adjusted by ±1 dB with the 200-Ω trimmers. It is absolutely necessary to adjust the gain of each preamplifier's channel because most of the cartridges I have used do not have the same output signal on both channels. Usually, they differ by as much as 1.5 dB. Use a test record playing the same signal on both channels to adjust each channel's gain until they have exactly the same output.

Resistor RP11 buffers the amplifier's output if long cables are connected to the preamplifier. Capacitor CP13 is used to protect the next stage from any possible DC offset that may exist at the preamplifier's output. This should be a large value capacitor to prevent low frequencies loss so I used a bipolar capacitor.

On my prototype, I used a no-output capacitor since the output offset was less than 1.5 mVDC on the right channel and about 0.2 mVDC on the left channel and the following equipment, that was driven by the preamplifier, was DC protected with an input capacitor. But, be careful! You should use the output capacitor if the line preamplifier or the power amplifier that is going to be driven by this preamplifier is not DC protected with an input capacitor. Otherwise, problems may occur.

The Power Supply

A toroidal power transformer provides the required 2 × 20 VRMS for the preamplifier's power supply. I used a low-noise 15-VA toroidal transformer with a nominal output voltage of 2 × 18 VRMS with a 230-VRMS input. This transformer actually gave 2 × 20.5 VRMS when it was connected to the preamplifier's power supply because the required power was much lower.

A 2-A/100-V bridge rectifies the AC voltage to DC voltage and charges the two 3,300-µF/35-V capacitors. Resistors R45, R46, R47, and R27, in series with capacitors C32, C33, C34, and C35 are in parallel with the rectifier diodes to reduce their noise.

Part	Value	Tolerance	Quantity
C1, C2, C3, C4, C5, C8, C9, C10, C11, C12	120 µF/25 V, electrolytic		10
C6, C7, CP4, CP5, CP7, CP8, CP11, CP12	100 nF, 63 V, polyester	10%	8
C13, C14, C15, C16	100 pF, polyester	10%	4
C17, C21, C30, C31	3,300 µF/35 V, electrolytic		4
C32, C33, C34, C35	33 nF, polyester	10%	4
CP1 (for RF protection)	10 nF, GOG, NPO	5%	1
CP2, Cartridge loading	470 pF, polypropylene	5%	1
CP3 (for 0.5-mV sensitivity)	2.2 nF, polypropylene	5%	1
CP3 (for 0.25-mV sensitivity)	1.5 nF, polypropylene	5%	1
CP6, CP9	2.2 µF, MKP	10%	2
CP10	220 pF, polyester	10%	1
CR1, CR2	100 nF, MKP	1%	2
CP13 (optional)	47 µF, bipolar		1
R1, R5, R6, R16, R17, R18	1 kΩ	1%	6
R2, R40	820 Ω	1%	2
R3, R13, R14, R20, R21, R22	510 Ω	1%	6
R4, R15	5.1 kΩ	1%	2
R7, R11	12 kΩ	1%	2
R8, R10	240 Ω	1%	2
R23, R44	1.5 kΩ	1%	2
R27, R45, R46, R47	47 Ω	1%	4
R9, R12, R38, R39, R41, R43	10 Ω	1%	6
RP2, Cartridge loading	100 Ω	1%	1
RP3 (for 0.5-mV sensitivity)	15 Ω	1%	1
RP3 (for 0.25-mV sensitivity)	10 Ω	1%	1
RP4 (for 0.5-mV sensitivity)	316 Ω	1%	1
RP4 (for 0.25-mV sensitivity)	432 Ω	1%	1
RP5, RP8	15.4 kΩ	1%	2
RP6, RP7	47.5 Ω	1%	2
RP9, RP10	750 Ω	1%	2
RP12	100 kΩ	1%	1
RR1	33.2 Ω	0.1%	1
RR2	750 Ω	0.1%	1
RR3	150 Ω	0.1%	1
RR4	3.32 kΩ	0.1%	1
RR5	61.9 Ω	0.1%	1
RR6	30.9 kΩ	0.1%	1
RR7	909 Ω	0.1%	1
D1, D2, D4, D5	1N4148		4
D3, D6	1N5235B	Zener	2
IC1, IC2	AD817		2
IC3	LM337T		1
IC4	LM317T		1
IC5, IC6, IC7	AD797N		3
IN, OUT, PCB-mounted RCA connector	TOBU3, phono socket		2
KK1, KK2, Heatsink	FK218		2
LED1, LED2	Green LED 3 mm		2
Q3	D44H11		1
Q6	D45H11		1
T1, Transistor	BC547B		1
T2, Transistor	BC557B		1
TR, Trimmer	200 Ω		1
BR1, bridge rectifier	2KBP		1
UREF1, UREF2	LM329B		2
Transformer	2 × 18 VAC/15 VA		1
IEC Entry with EMI filter and 0.5-A fuse			1
Metal enclosure	330 mm × 280 mm × 40 mm		1

Table 1: The MC100 preamplifier's parts list (quantity for one channel) is shown.

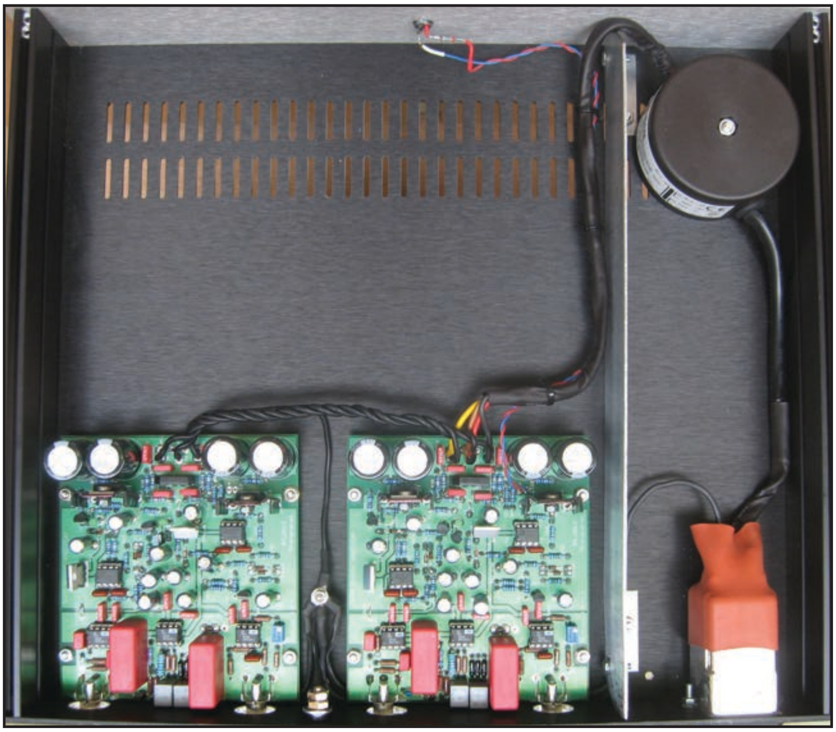


Photo 4: The MC100 preamplifier's internal view shows the power transformer on the same box as the gain circuits.

Photo 5: The preamplifier's back panel is clearly labeled.

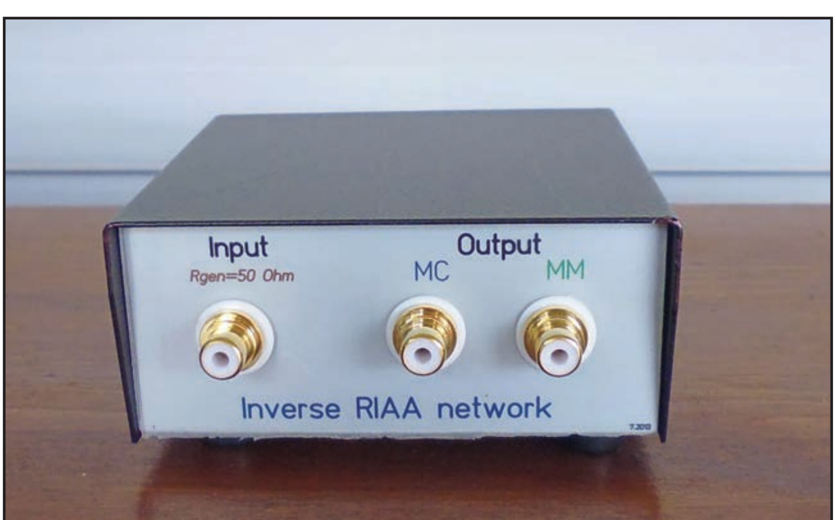


Photo 6: This enclosure contains the inverse RIAA network that I used for the measurements.

Resistors R38, R39, R41, and R43 with the reservoir capacitors C30, C31, C17, and C21 form low-pass filters to reduce the ripple voltages of the capacitors.

The two regulators are identical to the Jung regulators. Refer the Jung's original article for further details (see Resources).

A DC voltage above ±19 V should be always kept at the input of the LM317 and the LM337 regulators. This is very important because if the voltage falls below these levels, the regulators will not properly work or they may not work at all.

The transformer I used measured a ±24.2-V voltage at the input of the regulators when the input voltage was at the 230-VRMS nominal value. Keeping in mind the tolerance of the 230-VRMS mains voltage is ±10%, the mains voltage may change from 207 to 253 VRMS. With the mains voltage at 200 VRMS, I measured ±20.6 VDC at the input of the regulators. When the mains voltage went up to 250 VRMS, I measured ±27 VDC.

The input voltage at the regulators IC3 and IC4 inputs will vary from ± 20.6 to ±27 VDC, which means they will need some heatsinking to avoid high temperatures on their cases. I used two Fischer Elektronick FK218SA-32 heatsinks, which have a 21°C/W thermal resistance on each regulator (see **Photo 2**). This was enough to keep them cool even when the line voltage was set continuously to 250 VRMS. I set the nominal output voltages to ±14 V and measured 14.19 and –14.18 V on my prototype.

Amplifier Construction

One of the main objectives of this amplifier's design was to include all its components on a single PCB so the construction would be easy. I used CadSoft's EAGLE software to design the PCB (see Resources). The one drawback is the demonstration program limits the PCB's maximum dimensions. The outcome was a 90 mm × 115 mm double-sided PCB with ground planes on the top and bottom layers. I used a separate ground plane for the power supply area.

Due to the limited dimensions, each PCB has components for only one channel. Two of these PCBs are needed for the construction of a stereo preamplifier. **Photo 3** shows the PCBs, consist of 1.6-mm thickness FR4 board with 35-µm copper. Also they have plated through holes, solder resistant on both sides of the board, and silkscreen on the top side.

Figure 3 shows the PCB's complete guide assembly. **Table 1** shows the parts list for one channel. I used high-quality parts for the preamplifier's construction. I separated the signal

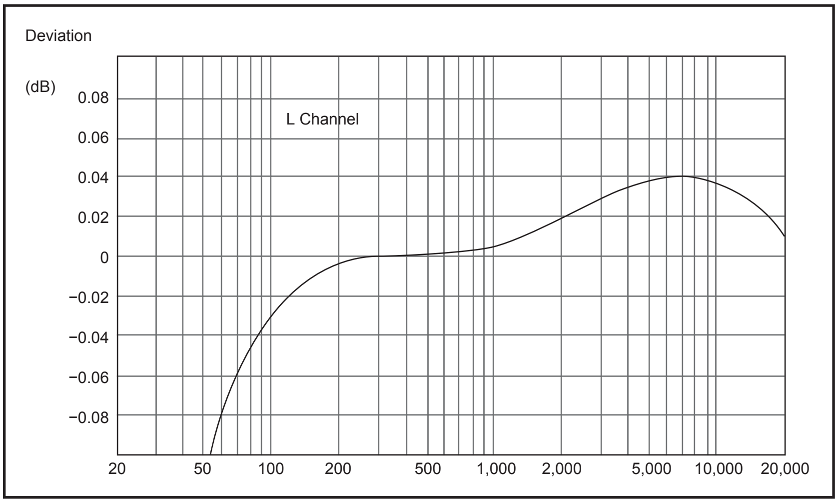


Figure 4: The preamplifier's RIAA accuracy is less than ±0.04 dB from 80 Hz to 20 kHz.

ground and the power supply ground plane to minimize the noise. They are connected to a single point.

I housed the preamplifier's components in a metal enclosure with a 10-mm aluminum front panel. The enclosure's external dimensions measured 330 mm wide × 280 mm deep × 50 mm high.

Photo 1 shows the amplifier's front panel. A single blue LED in the center of the panel indicates the preamplifier's operation. I used Front Panel Express's Front Panel Designer program to create the front plate text. Then, I used my inkjet printer to transfer the file to a transparent self-adhesive sheet.

Photo 4 shows an internal view of the preamplifier. Although my first thought was to use a separate enclosure for the transformer, I decided to mount the power transformer on the same box as the gain circuits. I used a quality toroidal power transformer, which, according to the manufacturer, has an extremely low radiated magnetic field and is suitable for sensitive electronics. I mounted it as far as possible from the amplifier's electronics parts to avoid any possible interference due to the high voltage gains of the preamplifier circuits, especially at the low frequencies. Additionally, I used a metal sheet for shielding between the power transformer and the PCBs to eliminate any possible interference.

The two PCBs were mounted to the back of the enclosure because of the on-board RCA input and output connectors. This means cabling is not required for the preamplifier's input and output connections. The star point ground is placed between the amplifier's two PCBs. Here, all the grounds are connected together to the chassis.

Photo 5 shows the amplifier's back panel. The IEC socket for the 230-V mains is on the right side with an integrated EMI filter. The fuse holder is next to it. On the left side are four large holes for the RCA input and output sockets, which are mounted directly on the PCB.

For all the amplifier's op-amps, I used high-quality IC sockets with gold-plated pins. This facilitates amplifier testing and any repairs that may be necessary.

Before inserting any of the expensive AD797 op-amps, check the power supply voltages on pins 4 and 7. They should be ±14 V.

Preamplifier Measurements

After preamplifier construction, I performed the usual measurements to check its performance. First, I measured the preamplifier's gain at 1 kHz. This was 65 dB with the trimmer 200 Ω turned fully left and 67 dB with the trimmer 200 Ω fully right. The 2-dB margin is more than enough to adjust any balance errors on the output of every moving coil cartridge.

Next, I measured the accuracy of the preamplifier's RIAA curve. I have an accurate inverse RIAA network, which was constructed with high-accuracy components (0.1% resistors and 1% capacitors), using the ideal values of the schematic diagram (see Resources). **Photo 6** shows the network's enclosure.

Figure 4 shows the preamplifier's accuracy with this inverse RIAA network connected at its input. The deviation from the ideal RIAA curve is less than ±0.04 dB from 80 Hz to 20 kHz. Of course, this includes the error of the inverse RIAA network and the MC100 preamplifier.

The increased deviation below 80 Hz is intentional and is due to the two high-pass filters used in the circuit to reduce the infrasonic frequencies. The attenuation is −0.5 dB at 20 Hz and −1.8 dB at 10 Hz. The −3-dB point is at 7.8 Hz.

Then, I measured the maximum output voltage before clipping at the preamplifier's output. This was 9 VRMS (12.7 V peak) as expected from an op-amp with a ±14-V voltage supply. **Table 2** shows the corresponding maximum permitted voltages in several different frequencies at the preamplifier's

Frequency (Hz)	Input level (mV)
20	0.5
100	1
1,000	4.4
2,000	6
10,000	21.5
20,000	42

Table 2: The numbers indicate the MC100 preamplifier's maximum input voltage when the gain at 1 kHz was set at 66 dB.

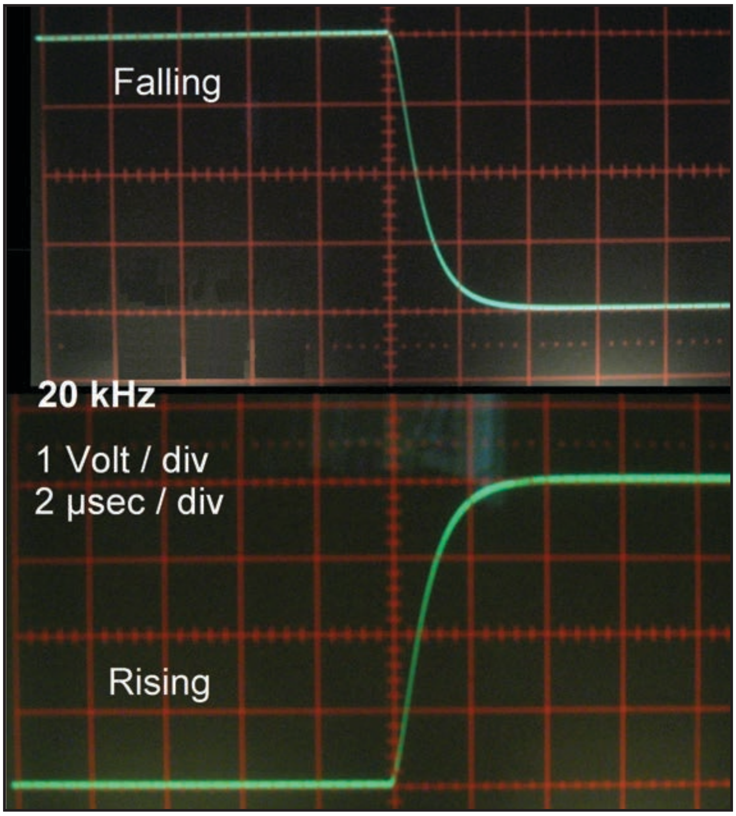


Photo 7: The response of Inverse RIAA network and the preamplifier on a 20-kHz square wave is shown.

input, which I measured for confirmation.

Finally I measured the inverse RIAA network's response and the preamplifier to a 20-kHz square wave. I connected the generator to the input of the inverse RIAA network and the output of the inverse RIAA network to the input of the preamplifier. **Photo 7** shows the excellent response of the preamplifier for the rising and the falling part of the 20 kHz square wave.

Overall Impression

This is an easy to construct high-quality RIAA moving coil preamplifier. It offers an excellent transparency and has a very low noise that is almost undetectable even when my system is set to its maximum volume. If you need such a preamplifier, I think you will enjoy this one. I rediscovered several of my old LPs, and I really appreciate my system's sound on the quality recordings.

Author's Note: I have a small quantity of PCBs (manufactured as shown in Photo 4) for the construction of this preamplifier. If you are interested, send me an e-mail at gntanavaras@gmail.com

Resources

W. Jung, "Improved Positive/Negative Regulators," *Audio Electronics*, Issue 4, 2000.

W. Jung and S. Lipshitz, "A High-Accuracy Inverse RIAA Network," *The Audio Amateur*, January 1980.

Sources

AD797 Op-amp
Analog Devices, Inc. | www.analog.com

EAGLE PCB software
CadSoft | www.cadsoftusa.com

FK218SA-32 Heatsinks
Fischer Elektronick GmbH | www.fischerelektronic.de

Front Panel Designs
Front Panel Express, LLC | www.frontpanelexpress.com

Testing a Tripath Power Amplifier

By
Ron Tipton
(United States)

A Tripath or Class-T power amplifier is a proprietary version of the Class-D switching amplifier developed by Tripath Technology. Tripath claims its architecture uses "adaptive" processing to improve general switching amplifier design. This article examines two amplifiers that use Tripath ICs: the Tripath TA2024, which is rated at 15 W per channel, and the Lepai TA2020A+, which is rated at 20 W per channel. The amplifiers sound good, but testing them is problematic. This article explains their testing challenges.

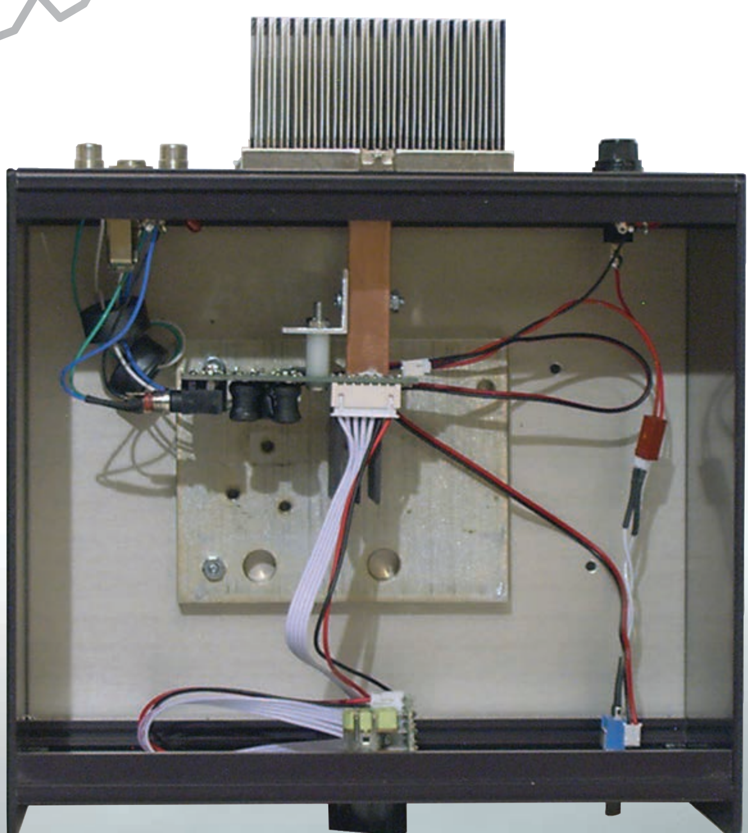


Photo 1: The Tripath TA2024 amplifier module is shown with its heat removal system. The underside of the circuit board rests on one end of the copper bar that is soldered to the finned heatsink. A portion of the finned heatsink expoxied to the top surface of the IC can be seen underneath the light-colored ribbon cable.

As I mentioned, the Tripath works very well reproducing music but distortion testing must be done at a single frequency. So, this architecture does not deliver much output power at a single frequency because the IC heats up and its internal thermal sensor shuts it down.

Getting the Heat Out

I increased the cooling for a Tripath Technology TA2024 module by adding heatsinks (see **Photo 1**). The IC's underside rests on the end of a 0.5" × 0.5" copper bar, which is soldered to the copper bottom of the finned radiator fastened to the back of the enclosure. I used some silver bearing thermal grease between the IC and the copper bar to maximize the heat conduction. I also used conductive epoxy to cement a finned black aluminum heatsink to the top of the IC. The modifications are very effective.

The TA2020 IC in the Lepai TA2020A+ is physically larger and it is spring clipped to a finned aluminum heatsink inside the enclosure. I increased the cooling by adding a 12-VDC fan (see **Photo 2**). I also drilled six 0.3" diameter holes in each side of the enclosure for increased air flow. The Lepai unit is lightweight and the cables drag it around so I added a walnut base and a switch on the left rear corner to turn the fan off for normal listening. Again, the increased cooling is very effective.

Floating Speaker Connectors

Most, if not all, analog power amplifiers do not object if the "low" sides of the speaker outputs are connected to a signal common. But the Tripath architecture does object and the internal fault detection circuit shuts down the amplifier. The speaker connectors are floating so one side of the dummy load can't be connected to circuit common. This calls for an output isolation transformer. I used a pair of 70-V line to 8-Ω units connected "line to line" for 8 Ω in and 8 Ω out. This isolation works well and was used for testing both amplifiers.

Output Filter

Switching amplifiers generate a lot of high-frequency "noise" in their output signals. Loudspeakers ignore the noise so it isn't a listening problem; however, it does interfere with distortion testing. I built an active output filter with an eight-pole Bessel low-pass response and a 30-kHz cutoff frequency. **Photo 3** shows the filter. **Figure 1** shows the circuit diagram. The output isolation transformer attenuates some of the high-frequency noise but there seems to be enough coupling for the output filter to be useful.

Testing

Now that I've taken care of the housekeeping issues, it's time to test the unit. The total harmonic distortion (THD) test is straightforward (see **Figure 2**). My signal source is a 1,000-Hz Wein-bridge oscillator with a 0.0095% measured THD that I built for another project. **Figure 3** shows its output spectrum as measured by TrueRTA.

The TA2020 module's THD measures 0.13% at 2.5 W RMS into 8 Ω. **Figure 4** shows the TrueRTA spectrum. At 5 W RMS into an 8-Ω output the THD increases to 0.19% with the second and third harmonics still predominate. At higher outputs into 8 Ω, the THD increases rapidly. For example, at 10 W RMS the sine wave is severely clipped and the THD is 38%. The 12-VDC power supply is insufficient to deliver 10 W RMS into an 8-Ω load.

The Lepai module measures 0.16% at 2.5 W RMS into 8 Ω and 0.17% at 5 W RMS. The output spectra are very similar to **Figure 4**. These numbers exceed the published THD specifications for power into 8 Ω but not by a significant amount.

Intermodulation Distortion

Rather than making static intermodulation distortion (IMD) measurements, I chose to use a transient intermodulation distortion (TIM) test. This test is not as well known but I fully described a method for it in my article "TIM Revisited" (*Multimedia Manufacturer,* 2008).

Physically, TIM is produced in analog power amplifiers by global negative feedback. This occurs when the open-loop cutoff frequency of the output stages is less than the open-loop cutoff frequency of the input stages. Many modern analog power amplifiers have low TIM, but I was uncertain if this was true with a Tripath amplifier.

My measurement method uses a signal source of a microprocessor-generated 3,100-Hz square wave added to a sine wave at 14,368 Hz that has an amplitude 12 dB lower than the square wave. (This composite signal is band limited 2 to 30 kHz by an eight-pole Butterworth low-pass filter.) With 3,100 Hz

About the Author

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

Photo 2: The Lepai TA2020A+ amplifier is shown with its cooling fan and the walnut base added for increased weight. The rear, left toggle switch turns the fan on or off.

Photo 3: A front view of the active low-pass filter used in the Class-T amplifier testing to remove the output high frequency switching noise is shown. This filter was not used for the transient response test.

being f1 and 14,368 Hz being f2, the measurement circuit looks for a product at 1,132 Hz (i.e., 5f1 – f2). **Figure 5** shows the TIM test waveform. **Figure 6** shows the spectrum. Note that **Figure 6** shows no product at 1,132 Hz above –117 dBu so any 1,132 Hz in the output is due to amplifier TIM. The idea is the sharp rise and fall times of the square wave stress any slew-rate limited portions of the amplifier. This makes the intermodulation products between the square wave harmonics and the sine wave show up as TIM. **Figure 7** shows a block diagram of the test setup. The measurement is straightforward, however, building the equipment wasn't!

In my referenced article, I measured eight different analog amplifiers. One used an integrated circuit (National LM3876), six were all solid-state and one used a vacuum-tube input with a MOSFET output. The IC design and the Leach "Low TIM" design showed no TIM that I could measure at either 1 or 10 W RMS output, while the others did.

Figure 8 shows the TIM test result for the TA2024 module at 2.5 W RMS into 8 Ω. It shows a TIM product of about 10 dB. The Lepai results are similar with 15 dB at 2.5 W RMS and 21 dB at 5 W RMS. These numbers mean the Tripath designs are similar to what I measured on some of the analog designs. Although measurable TIM is present, it is probably not audible. I measured TIM as high as 30 dB on one of the analog amplifiers, yet it sounded fine in listening tests. TIM is audibly similar to crossover distortion so it would be heard if it was present at large enough amplitude.

Transient Response

Transient response is a function of an amplifier's bandwidth. The higher the bandwidth, the better it is at reproducing percussive sounds with fast rise or fall times. Although these transient sounds are often above the range of human hearing, it has been amply demonstrated that they are perceived when they are cleanly reproduced. Music with percussive sounds (e.g., cymbals, harps, harpsichords, etc.) sound more "real" with a wider bandwidth amplifier given the same audio system.

The TIM signal generator's 3,100-Hz square wave is produced by a microprocessor running at a 18 -MHz clock rate so the square wave has rise and fall times of 220 nS (four times the clock period). I used this signal to drive two amplifiers to 2.5 W RMS into 8 Ω.

Figure 9 shows the result for the Lepai amplifier. The top trace is the input and the lower trace is the

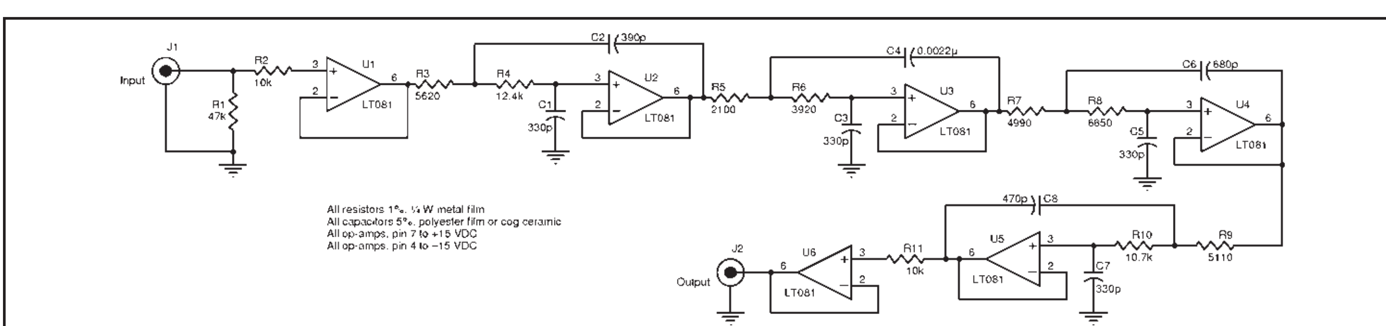


Figure 1: The Tripath amplifier output is tested with a low-pass filter with an eight-pole Bessel response and 30-kHz cutoff frequency. Low noise op-amps are not needed for this application.

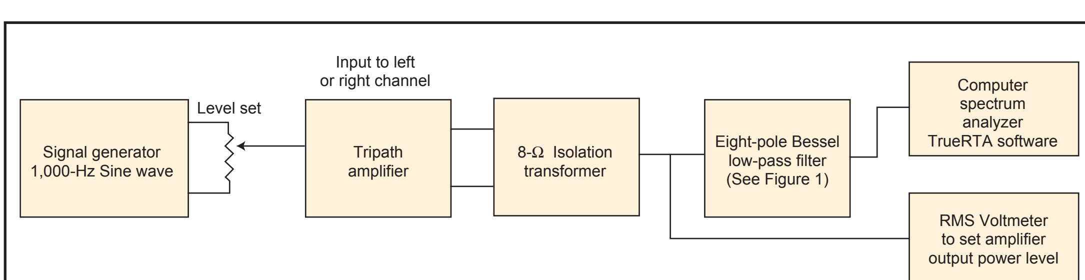


Figure 2: The block diagram of the harmonic distortion test equipment setup is shown. Any low-distortion signal generator could be used.

The Tripath Story

Photo 1: Tripath's headquarters were built in Santa Clara, CA.

Photo 2: A Tripath TA2041 four-channel amplifier was rated at a maximum output of 280 W (70 W per channel).

In 1995, Dr. Adya S. Tripathi had some ideas for improving the performance of Class-D audio power amplifiers so he started Tripath, Inc. to pursue these goals. Apparently he found adequate venture capital as evidenced by the headquarters building in Santa Clara, CA, with some 200 employees (see **Photo 1**).

Class-D designs were enjoying some popularity, especially for woofer power, because of their high efficiency compared to Class-A and Class-AB. However, their total harmonic distortion (THD) was fairly high at higher audio frequencies. This was one area in which Tripath achieved significant improvement. Just how this was accomplished can be surmised from online information.

A starting point is an interview Crutchfield Company, a consumer audio and video distributor, conducted with Shawn Scarlett, Tripath's Senior Product Manager. In answer to the question "how does a Class-T amplifier work?" Scarlett replied, "Class-T (a registered trademark) is the name for our proprietary architecture that improves on general switching amps. It uses a combination of 'predictive' and 'adaptive' processing. On top of that, we use a very high switching frequency."

The switching frequency is variously reported in different sources. A web post by Jon Iverson (*Stereophile* magazine, November 1998) refers to the switching frequency as "up to 1.5 MHz." However, Brian Santos's article, "25 Microchips That Shook the World," (*IEEE Spectrum*, May 2009) refers to a 50-MHz switching frequency. Datasheets (up to 2002) agree with up to 1.5 MHz, but perhaps the frequency was increased as technology permitted.

Scarlett continued, "The basic idea is that we look at the incoming signal to determine the best way to encode it, making sure to minimize interference or mistakes. We then use feedback, or 'adaptive' processing to analyze the output and keep the system stable. The high switching frequency allows it to correct any issues quickly before they become audible. Because of the robustness of the system, we can maintain our fidelity even with mismatches in the output field-effect transistors (FETs), power supply 'ripple,' and other issues."

Scarlett presented us with three basic ideas: a high switching frequency, output-to-input feedback, and "processing" of some kind to overcome any mismatch between the output switching transistors. During its 12-year operating life, Tripath was granted some 100 patents. The most significant one from the point of view of this story seems to be US Patent #5,777,512, a 20-page document with many claims and quite a bit of detail. It reinforces the three basic ideas I mentioned. From the information, I drew **Figure 1.** It is my "best guess" at a commercial Tripath design. **Photo 2** shows the TA2041, a four-channel audio amplifier. This is a four-channel amplifier, with 50-W per channel at 10% THD with an efficiency of 85% at 50 W into 4 Ω. Note the relatively small size of the heatsink to dissipate the other 15% of input power.

I have spoken with a trademark-patent attorney and all a patent actually does is enable the holder to sue for possible infringement. When granted, the patent document is public information so anyone can use the ideas as long as the embodiment is not copied. It is apparent from the improvements that appeared in Class-D designs that other manufacturers were vigorously applying these ideas. The competition proved too great a burden so Tripath filed for Chapter 11 (reorganization) bankruptcy in February 2007. The company was purchased by Cirrus Logic latter that year but they soon discontinued the Tripath operation.

A web search will show there is no shortage of either Tripath amplifiers or chips so excellent innovation lives on even though Tripath probably did not profit from it as much as it had anticipated.

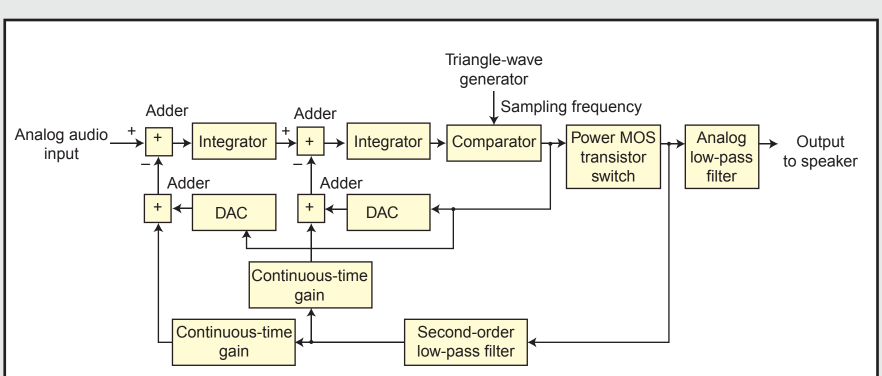


Figure 1: I derived this block diagram of the Tripath power amplifier design based on the available online information.

output. The output square wave rise time is 10 µs so the bandwidth is 35 kHz (i.e., bandwidth in Hz = 0.35/ rise time in seconds). As a sanity check, I also made this test with an input isolation transformer and 8-Ω load instead of the output isolation transformer. The result was the same. A 35-kHz bandwidth would provide a fair transient response. Whether it would be audibly noticeable depends on the type of music, the loudspeaker system, and the room acoustics.

Contrast **Figure 9** with **Figure 10** to get the

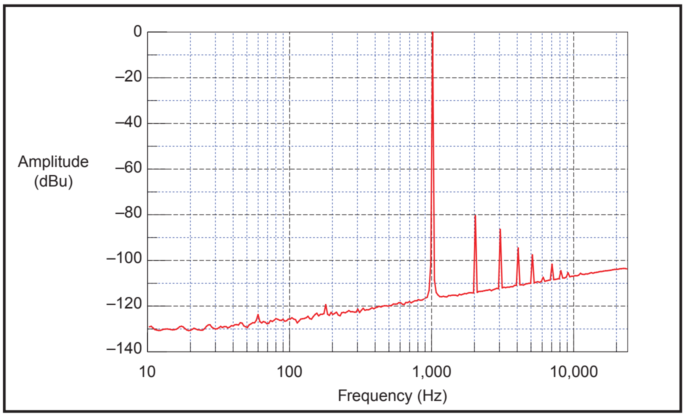


Figure 3: The output frequency spectrum of the 1,000-Hz Wein-bridge test signal generator provides a 0.0095% measured THD.

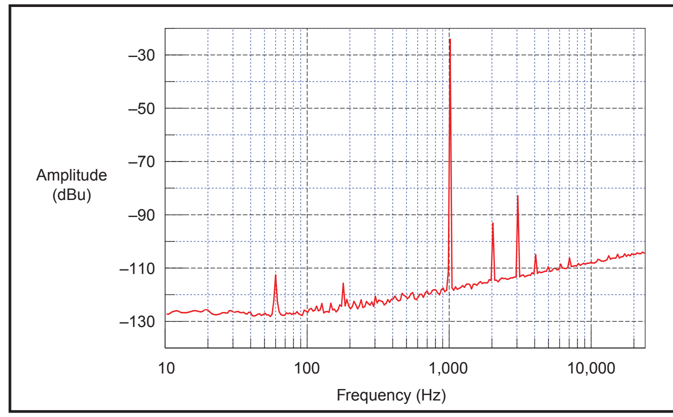


Figure 4: The TA2024 Tripath amplifier's output frequency spectrum is shown at 2.5 W output into 8 Ω. The amplifier volume control set to mid position. An attenuator was placed between the amp output and the sound-card input to prevent over driving. The measured THD is 0.13%.

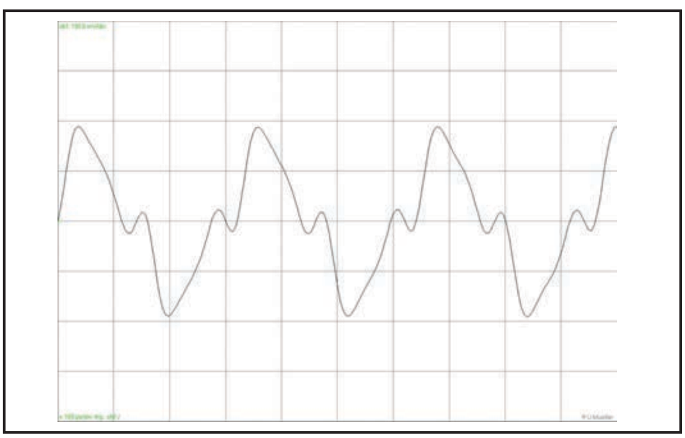


Figure 5: The TIM test signal consists of a 3,100-Hz square wave added to a 14,368-Hz sine wave with an amplitude 12 dB lower than the square wave. The horizontal axis is linear time, 100 µs per division. The vertical axis is amplitude, 100 mV per division.

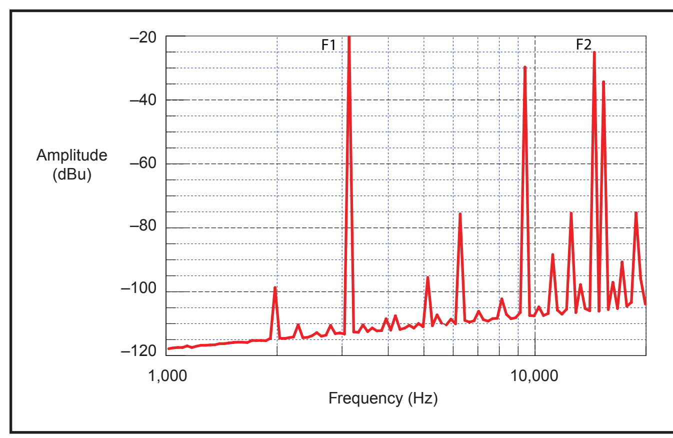


Figure 6: TIM test signal is shown at the output of the TIMFIL-1 eight-pole Butterworth band-pass filter. The lower band edge is 2,000 Hz and the upper band edge is 30 kHz. This is an active filter and it is housed in a cast aluminum enclosure to minimize noise pickup.

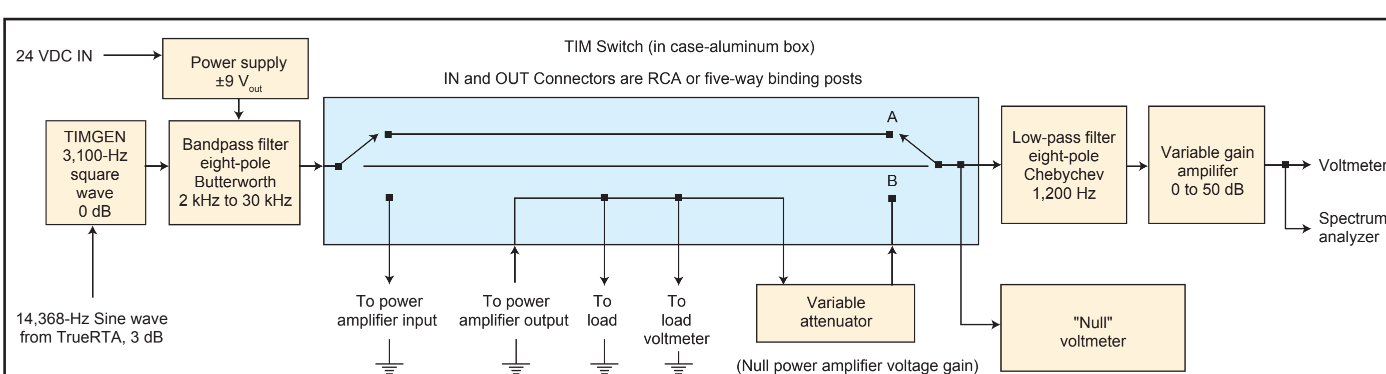


Figure 7: TIM measurement block diagram is shown here.

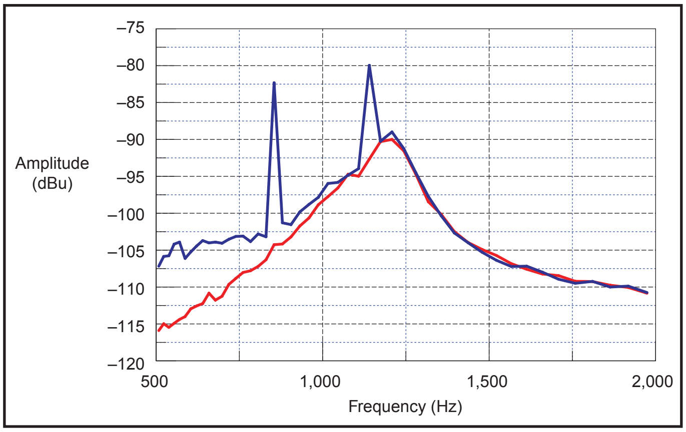


Figure 8: A TIM test result for the Tripath TA2024 is shown at 2.5 W RMS into 8 Ω. The lower graphed line (black) is the calibration baseline. The upper line (blue) is the TIM measurement with a 10 dB peak at 1,132 Hz.

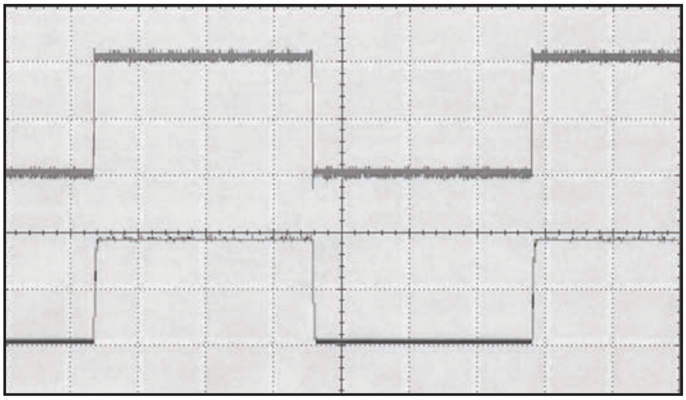


Figure 10: The transient test result for the Leach Low-TIM amplifier (version 4.5) shows 2.5 W RMS into 8 Ω. The upper trace is the input 3,100-Hz square wave. The lower trace is the amplifier output showing rise and fall times of about 1 µs with no overshoot or ringing. The horizontal scale is linear time, 0 to 500 µs.

result for the Leach Low-TIM amplifier. The rise time is less than a microsecond and there is no overshoot or ringing. The Tripath ringing shown in **Figure 9** is probably the result of the Tripath's lower bandwidth and perhaps its overall architecture. The Leach design is an excellent amplifier and you can get all the details, including a copy of Leach's original *Audio* magazine article by downloading the zip file from my website.

Resources

TDL Technology, Inc., www.tdl-tech.com/tim-mmm.zip.

R. Tipton, "TIM Revisited," *Multimedia Manufacturer*, November–December, 2008.

Sources

Tripath TA202A+ Amplifier
Lepai | www.lepai.us

LM3876 Amplifier
Texas Instruments, Inc. | www.ti.com

Tripath TA2024 Module and isolation transformers
Parts Express | www.parts-express.com

TrueRTA audio spectrum analyzer software
TrueAudio | www.trueaudio.com

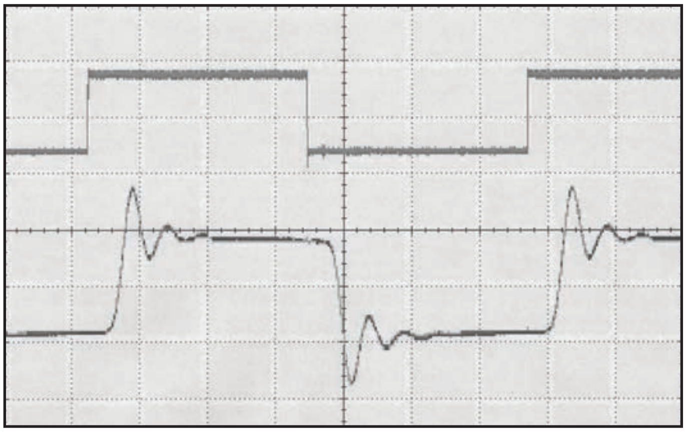


Figure 9: A transient test result for the Lepai amplifier is shown at 2.5 W RMS into 8 Ω. The top trace is the input 3,100-Hz square wave. The lower trace is the amplifier output showing rise and fall times of 10 µs. The horizontal scale is linear time, 0 to 500 µs.

Final Thoughts

The Tripath amplifiers are inexpensive so in a way it's unfair to compare them to amplifiers that cost hundreds of dollars. Nevertheless, the comparison is useful because you do "get what you pay for." Both the Tripath models I tested sounded excellent in listening tests until I compared them using music that stressed the transient response, then I could hear some differences. Still, there are many applications where these small, inexpensive, and power-efficient units will do nicely.

Reamping

Creativity with Control

By
Thomas Perazella
(United States)


Les Paul was an early adopter of reamping.

Reamping is a two-stage process whereby we first record a dry or clean track and then re-record the track by sending the clean track back through amplifiers and effects. This enables us to capture the musician's best performance. Later we can spend the time moving microphones around the room, changing amps, or adding effects as needed. We can also go back and change the recorded track's sound as the production advances.

The process has been used since the early days of sound recording. Pioneered by Les Paul and Phil Spector, the process was part of the Motown studio routine in the late 1960s and it was used by Roger Nichols in the 1970s when recording Steely Dan, Crosby Stills and Nash, and Frank Zappa. Roger Nichols and John Cuniberti designed application-specific "reamping" devices, which basically can be active or passive circuits, converting low-impedance balanced lines to guitar levels.

When I was working as the IT Director for an international company, I had a few key guiding precepts that drove our activities. Topping the list was the requirement that whatever else happened we would not mess up the data. Almost everything else was negotiable. What does this have to do with audio? In audio, as with so many other endeavors, the greatest amount of time and energy is spent on creating original "data." This data in many cases is also irretrievable if lost.

In music, loosely speaking, the data can be thought of as the notes produced by the artist. However, to be really useful, the data must be converted into information. Simply playing the notes at the same tempo with the same intensity does little to convey the artist's emotions. To do that, the artist relies on major and subtle changes in the musical elements. The artist may also use other auxiliary devices to expand the realm of possibilities.

On the surface it may seem this creative expression is a straightforward exercise. You decide what you want and then do it. The reality is there are so many factors to contend with, it is extremely rare when they all combine at the same time to make a one-shot performance work. Once a take is finished and auditioned, the artist often envisions other possibilities to create an even more compelling performance. It may require several attempts for the final outcome or "information" to happen. Often, the artist may have one take that has the perfect timing but the level is wrong. The next take may fix that problem but a special effect may be too prominent.

Enter Reamping

Even in the early days of music, recording people realized creative elements could be added to the basic performance to achieve better results. One of the critical elements was to record the base performance and then use other mechanisms to add to that recorded performance. By keeping the original performance intact and later adding effects, many combinations could be tried while the original work remained intact.

A fascinating history of the process's evolution is provided through a link to the Radial Engineering's history page (see Resources). As early as 1913 performances were recorded on Berliner discs, played back through "noise machines," and then re-recorded. Once electronic recording became the norm, it was discovered that a performance could be recorded on tape and then passed back through other electronic devices (e.g., guitar amplifiers) to modify the original sound. Thus, the name "reamping."

Paul and Spector were early pioneers in this technique's use. You may also be interested to know that almost every Steely Dan album was produced with reamping so problems with individual amps and speakers could be eliminated. Having a direct clean recording enabled them to substitute equipment as needed during the production phase. This is the classical "physical" version of reamping. In recent years, the use of software plug-ins in computer audio applications enabled a "virtual" form of reamping to develop.

How Reamping Works

The first step in the reamping process is to collect the "data," (i.e., the recording usually done directly from the guitar or keyboard output) before any amplification or reproduction occurs through speakers. Recently other sources have also been used including voice and drum tracks. This is known as a clean or "dry" track, which is free from most of the modifying effects of the amplifier, speaker, room acoustics, microphones, or microphone placement, as well as contributions from other artists and their instruments.

The initial "dry" track contains none of the sonic signatures that make the performance ultimately unique and satisfying to the listeners. In many cases, the same feed will be sent through an amplifier and speakers during the recording to give the artist a reference. In this way, the artist can make one or multiple passes to correct the performance.

"Physical" Reamping

With physical reamping, an engineer can take the recorded track's electrical output and run it through whatever devices are desired (e.g., various amps and effects boxes). The engineer can also change the microphones and the microphone placements until the desired effects are achieved. Each combination can be compared to the original performance so the performer does not need to perform repeated trials. Reamping also affords many other advantages. Dan Alexander Audio's website explains more advantages of this technique (see Resources).

What's the Catch?

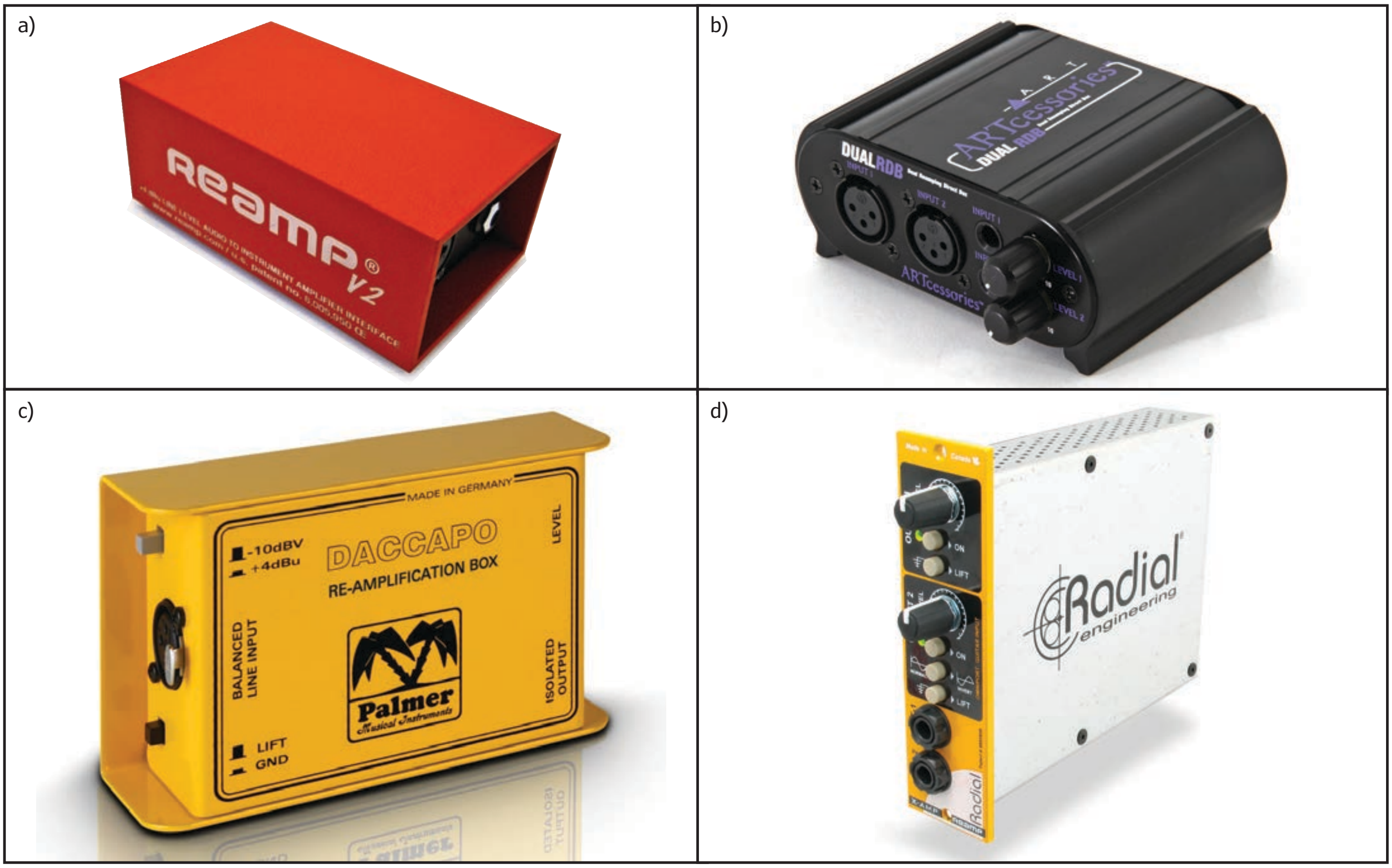
The simple idea of passing the signals from devices (e.g., guitars) to recorders and then back to amplifiers is not really so simple. Several electrical parameters vary between devices. At a minimum, these electrical differences must be corrected. In addition, those corrections must be done with minimum fidelity loss and noise pickup.

For example, guitar feeds are typically high-impedance low-level unbalanced signals. Those generally do not play well with the lower-impedance balanced inputs of the mixers that ultimately feed the recorders. To handle this problem, direct injection (DI) boxes were developed. The outputs of the recorders are commonly low-impedance, balanced, and much higher in level (4 dB) than what is expected by the amplifiers' unbalanced high-impedance inputs. To achieve this, reamping devices were developed.

Steely Dan made extensive use of reamping to help with amplifier and speaker problems. This image appeared on the back cover of the group's second album, *Countdown to Ectasy*.

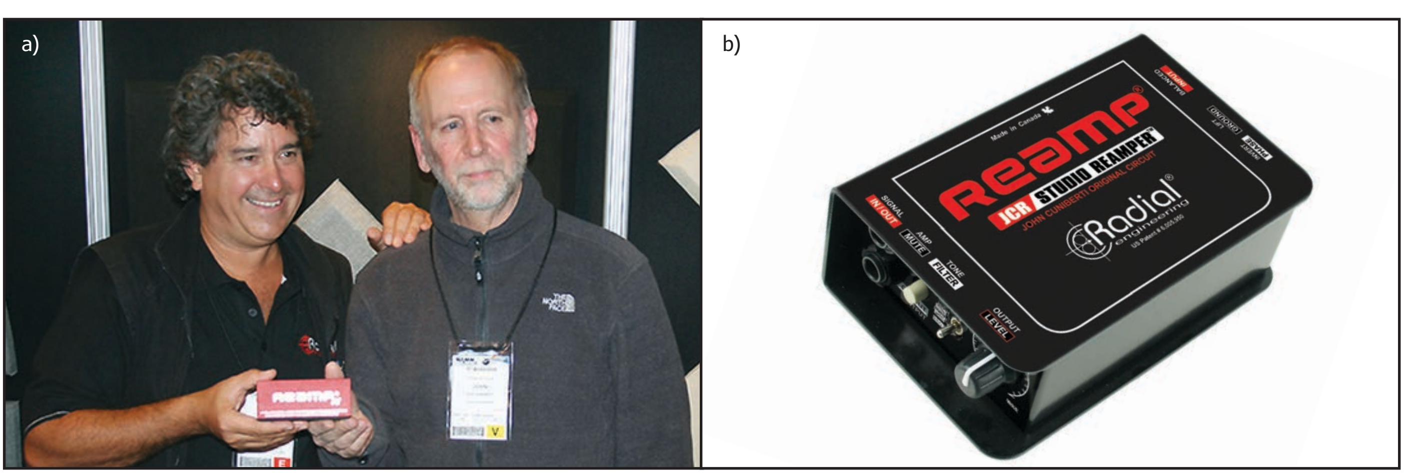
About the Author

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



Active and passive reamping devices are available. Some were adapted from direct injections (DIs), and others were optimized for different signal sources in a studio. This gallery shows the Reamp V2 as originally designed by John Cuniberti (a); the ART Dual Reamping Direct Box (b); the Palmer Daccapo Re-Amplification Box (c); and a Radial X-Amp, the Reamper, designed for 500-series racks (d).

It is important to really understand the damage that can be done to audio signals when improperly handling balanced to unbalanced transfers. Incorrectly handling those conversions can quickly lead to an unusable signal. Once these problems are understood it becomes apparent why reamping devices are needed. For a more complete description of the balanced to unbalanced scenario, see the



Peter Janis, Radial Engineering President of Radial, is pictured with John Cuniberti (a). In 2010, Cuniberti contacted Janis and offered to sell the Reamp brand and patent. The sale was announced at the 2011 National Association of Music Merchants (NAMM) show. Radial manufactures and sells the Reamp JCR Studio Reamper passive unit (b) with direct reference to the US Patent #6,005950 and the original Reamp circuit designed by Cuniberti. (Photos courtesy of Radial Engineering)


Linear Audio - Volume 0
Linear Audio - Volume 1
Linear Audio - Volume 2
Linear Audio - Volume 3
Linear Audio - Volume 4
Free preamp PCB included!*
Linear Audio - Volume 5
Free Linear Systems Inc. JFET pair included!*
Linear Audio - Volume 6

Balanced vs Unbalanced

At this point, we should take a brief look at balanced and unbalanced circuits and their interfacing considerations to understand successful reamping. Much has been written about these two circuit topologies and much of that is incorrect. To really get the proper information on circuit interconnects with grounding and noise considerations the best place to start is the June 1995 issue of *Journal of the Audio Engineering Society*. This issue is a treasure trove of articles by Neil Muncy, Bill Whitlock, Cal Perkins, and others who are experts in the field of signal interconnects. It is a must have and is available through the Audio Engineering Society (AES) website.

Another resource is my *audioXpress* article series about a bi-directional balanced to unbalanced converter I made using interesting ICs from THAT Corp. I made several key points in that article and I discuss the need to pay close attention to impedance mismatches and grounding when working with any interconnects, especially when changing from unbalanced to balanced and the reverse. Failure to do so can result in excessive noise, signal overload, and high levels of distortion. For example, when considering impedances while feeding a signal into a balanced circuit, even a 1% imbalance in the input impedance of the two signal lines will result in a drop of common mode noise rejection down to 40 dB, which is an unacceptable level.

The oldest and still the best way to achieve proper interfacing of balanced to unbalanced circuits is to use transformer coupling. A well-designed transformer with controlled leakage capacitance and inductance will have an input impedance that is essentially balanced as there is only one primary winding. However, those good transformers with wide bandwidth, freedom from ringing, immunity to inductive noise pickup and low hysteresis are quite expensive to make. They are still invaluable where potentially high stray voltages between the interconnected devices are present.

Active devices that don't use transformers have come a long way and can provide most of the benefits of transformers at a lower cost and with higher fidelity. They can also provide gain functions to enable parallel operation of many devices simultaneously. Choosing the correct active devices and using appropriate designs are necessary, as implementation errors will quickly result in poor performance. With the advent of these new active circuits, there are now multiple ways to achieve proper balanced to unbalanced conversions.

Resources

"Audio Acoustics Applications," *Journal of the Audio Engineering Society*, December 1995, www.aes.org/journal/toc/downloads/Contents_JAES_V40-43.pdf.

T. Perazella, "Maintaining Your Balance (Part 1)," *audioXpress*, July 2009.

——, "Maintaining Your Balance (Part 2)," *audioXpress* August 2009.

"Balanced vs Unbalanced" sidebar that accompanies this article.

Passive Reamping

The earliest reamping devices were physical (i.e., they used electronic means to make the necessary conversions of the electrical signals between the recorders and amplifiers while maintaining as closely as possible the original signal's fidelity). They were also passive in nature (i.e., they used devices that did not require external power to function). Those passive devices can be as simple as a resistive network to attenuate the recorder's output signal so as to not overload the amplifier's input and force it into gross distortion.

A more sophisticated method was to use a transformer as an impedance-matching device and also to convert the recorder balanced output to an unbalanced input required by the amplifier, combined with level adjusting resistors. In some cases, tone controls were included that enabled signal modifications even before going to the amplifier. While making changes to the equipment, mute switches were also added eliminating the need for the engineer to go off stage.

The History of Reamping

Although references to passive device usage to match impedances in guitar amplifier interfaces date back to the 1980s with the introduction of the Jensen JT-DBE transformer, others also were involved in the early efforts to solve technical hurdles. This idea's initial commercialization may have began when John Cuniberti patented a device (US Patent 6,005,950) to accomplish electrical conversions for quality reamped recordings in 1994.

Cuniberti introduced a transformer-coupled box and a volume control that he called the Reamp. It converted the balanced signal from the recorder output to match the guitar amplifier's unbalanced input while adjusting the level to prevent inadvertent overdrive of the amp input. The Reamp enabled an engineer the ability to experiment with combinations of amplifiers, speakers, microphones, and so forth until the desired result was obtained without compromising the original performance's integrity.

There are currently discussions about the validity of the patent issued to Cuniberti based on prior art (see Resources). It is not my intention to enter that fray, but I will simply say that there is a common misconception that if someone had previously developed a product that did an action it would invalidate all future patents that also achieved that action. However, it is almost impossible to find any device that works today without relying on some prior technology. What makes a device patentable is the ability to utilize a new and unique technology or implementation method to achieve either the same effect in a better way or to further expand the effect. As an example, drinks were stored in cans for many years before the advent of the pop top can. That fact that cans were used to store drinks did not prevent the pop top from being patented as an improvement to the can's functionality.

Little Labs Redeye 3D does double duty as a direct injection (DI) box and reamper. (Photo courtesy of Little Labs)

Enter Active Reamping

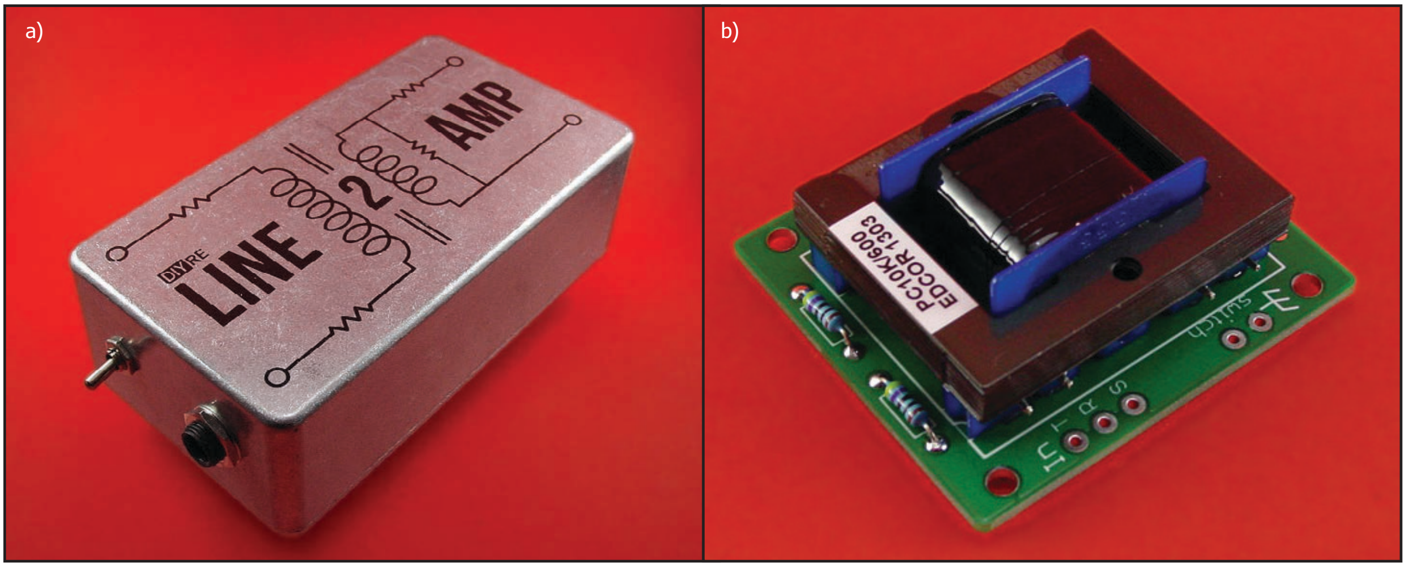
It became apparent that in some situations (e.g., where multiple devices had to be driven at the same time), an active device was needed to provide gain and buffering. In addition, although passive devices (e.g., transformers) are considered superior, they can and do have problems with fidelity. The better designs work quite well and have advantages, which are described in the "Balanced vs Unbalanced" sidebar.

Contrary to popular belief, transformers are not distortionless especially when driven into overload. I have heard it said they don't distort, they just saturate. If you have ever measured distortion in a transformer in saturation, you will understand that they actually do distort and potentially to very high levels. Many engineers find transformers endearing because they produce distortions that are more pleasing to the ear than the hard limiting that can occur with active devices.

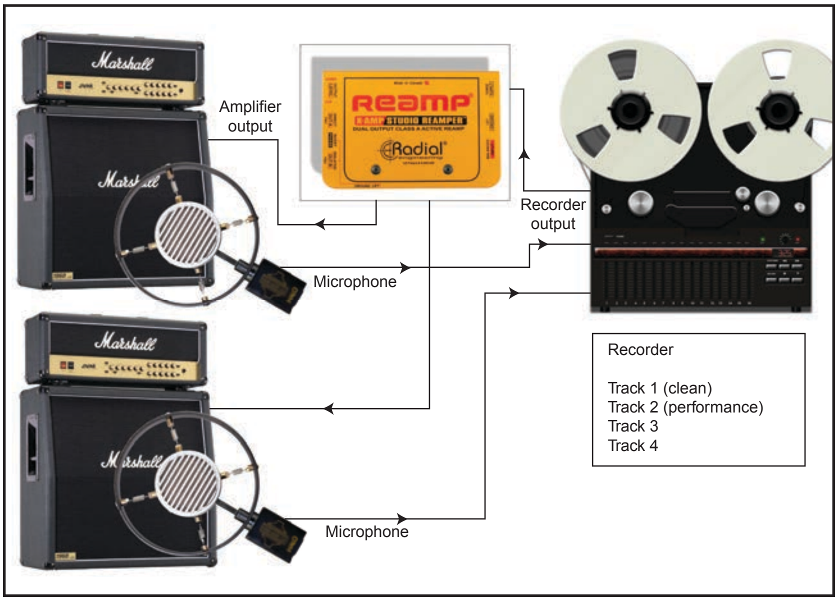
Robert Neve's famous mixer designs, which are still being duplicated, extensively used transformers for their ability to properly handle the unbalanced to balanced interfaces and to introduce signal modifications (distortions) determined to be the "right sound." However, many people prefer to control the modifications instead of having them built into the transformers. For those reasons, the active reampers became popular.

In 2005, Radial Engineering of Port Coquitlam, BC, Canada, began building the X-Amp active reamp to fill the need for gain and to provide additional features (e.g., tone control and muting). In 2011, Radial Engineering purchased the assets of Cuniberti's company assuring it the rights to continue to develop and market the Reamp line.

Currently there are many other physical reamping devices available with varying features and costs (e.g., the Coopersonic Reamping Device, the Little Labs Redeye 3D Phantom direct box and re-amplifier, and the LINE2AMP passive re-amplifier).



The Line2amp provides low-cost reamping in assembled (a) or kit form with the PCB (b). The Line2amp uses a US-wound transformer by Edcor to convert a balanced, line-level signal to an unbalanced, guitar level while isolating the grounds to prevent noise and hum. (Photos courtesy of DIY Recording Equipment)



The signal flow in a reamping setup is shown here.

Passive vs Active

It is interesting to note how much snake oil is floating around the audio industry. One of the most difficult things to overcome is the notion that a particular component or topology is superior to all others no matter what the circumstance. That is just not true. It is important to identify the factors that are key to your project and then find the device that will provide the result you need with the fewest compromises.

For example, the idea that transformers cannot provide high fidelity is simply untrue. The statement that active devices will produce nasty or glassy hard overload distortions depends on the circuits. Some recent devices such as the Analog Devices ADA4700-1 op-amp can work with a 100-V power supply making it virtually impossible to drive them into overload with any input voltage used in the recording process while providing vanishing low distortions with very wide bandwidth. For simplicity, we will only examine the interfacing requirements and leave other possible reamper special effects (e.g., tone controls) out of the equation.

A passive reamper's benefits include a high overload capability with potentially benign overload effects. Transformers enable you to isolate relatively high voltage offsets and in some cases provide very low noise. Also, a power supply is not needed.

The disadvantages of a passive reamper include: potential problems with the balanced to imbalanced interface if only resistive components are used and costly high-quality transformers. If low-cost transformers are substituted, you could have limited bandwidth, saturation-induced distortion, susceptibility to magnetically induced noise pickup, and less output drive potentially limiting the number of devices that can be simultaneously driven.

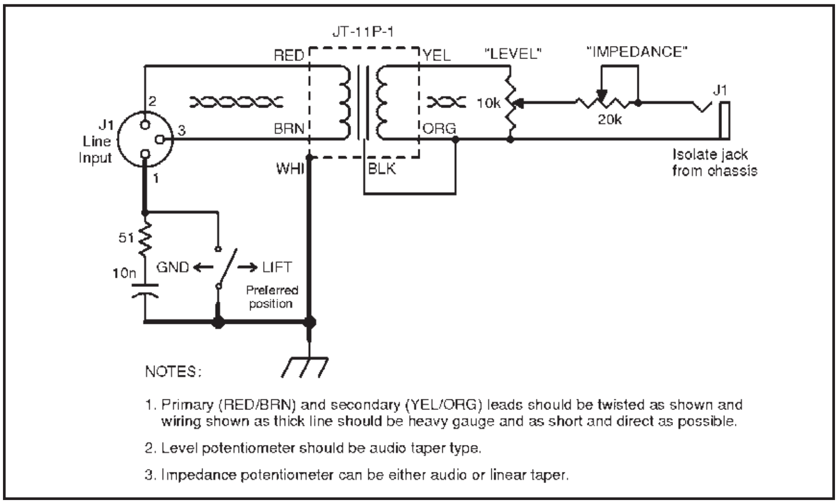
The benefits of an active reamper include: no transformer distortion, very wide bandwidth and low distortion if properly designed and used within design limits, generally a lower cost for equivalent performance, potentially low susceptibility to magnetically induced noise, excellent balanced to unbalanced performance if properly designed with available devices such as the THAT Corp. 1200 series InGenius balanced line receiver ICs, and lower output impedance to drive multiple loads.

The disadvantages of an active reamper include: the need for a quality power supply, possible power supply induced noise, hard-limiting induced distortions possible with marginal designs, and susceptibility to poor noise performance with designs that don't adhere to proper balanced to unbalanced practices.

What device to use depends on those factors, cost, and other feature considerations. You should also decide whether you want the reamper to simply interface the devices or perform signal modifications. Remember, signal changes such as those caused by transformer saturation may be desirable in some cases and may cause problems in others. Generally, modifications you can control are better than those that are "hard wired" into a device.

"Virtual" Reamping

As powerful as traditional reamping methods are, they involve a lot of time and physical effort to move and connect all the various elements of the signal chain in a real environment. In addition,



The Reamper schematic uses a Jensen JT-11P-1 transformer.

many different devices must be on hand to try the various effects.

Enter the computer. The electrical parameters that must be met to create the "dry" track in the physical reamping process must also be addressed when using virtual reamping. In both methods you must first have a properly recorded dry track. In many cases, a DI box will suffice as an interface for the original recording.

By using one of the many powerful recording and editing applications available, the dry track can be passed through the computer using plug-ins to modify the sound. There are plug-ins to mimic almost every physical effect and even some that only exist in the virtual world.

For example, some plug-ins have algorithms that enable the digital signal to be modified as though it had been played back through classic tube guitar amps (e.g., a 1959 Fender Bassman or a 1994 Mesa/Boogie Tremoverb). A few examples of those applications include IK's AmpliTube 3, Peavey's Revalver III.V, and Line6's POD Farm 2.5. There are obvious benefits to doing the reamping this way. There is no need to have multiple physical devices, which require storage space; no moving and setup; and no need to maintain old electronics. The cost of a complete set of plug-ins can be less than the cost to troubleshoot and fix an old amplifier. The time between auditioning different combinations of equipment and effects is minimal. Monitoring can be done with headphones further eliminating room effects in the mastering environment. It also enbales the mastering to be done outside a mastering studio. So the question is: Will virtual reamping replace physical reamping?

The Verdict

Predicting the winner of the physical vs virtual reamping battle is an exercise in futility. In reality, they both have a place. This is especially true because reamping's primary goal is to create a uniquely satisfying musical experience. Sometimes the physical sensations involved with actually being in the recording space with equipment and other artists can stimulate creative juices in a way that working at a computer cannot match. At other times, the physical drudgery of working with all that equipment can dull the senses much as continually redoing the original performance can tire the performer. The choice really depends on the performer, the day, and the mood.

The best professionals never let the tools drive their creativity. They will choose the tool that is appropriate for the particular task at hand. I'm sure books could be written on all the ways that either

Using virtual reamping on a computer enables a properly recorded dry track to be passed through plugins to modify the sound with virtually every effect conceivable.

method can produce spectacular results. So if the engineers understand not only the process of reamping but also what tools are available and how they can enhance creativity, all of us who enjoy the results of those creative endeavors will win. That includes those who create the original musical data and those who use reamping to create the musical information.

Resources

AMS Neve, Ltd., "Getting Wound Up About Neve Transformers," http://ams-neve.com/news-and-events/getting-wound-about-neve-transformers.

Dan Alexander Audio, "Reamp," http://danalexanderaudio.com/reamp.html.

Gearslutz.com, "Who Invented Reamping?" www.gearslutz.com/board/so-much-gear-so-little-time/7555-who-invented-reamping.html.

Radial Engineering, Ltd., "The History of Reamping, "www.radialeng.com/reamp-history.php.

Rupert Neve Designs, "Discrete Analogue Mixer," www.rupertneve.com/products/analogue-mixer.

Sources

ADA4700-1 Op-amp
Analog Devices, Inc. | www.analog.com

1200 Series InGenius balanced line receiver ICs
THAT Corp. | www.thatcorp.com

DISTORTION in Tube Guitar Amplifiers

By
Richard Honeycutt
(United States)

As discussed in a previous Hollow-State Electronics article, the first guitar amplifiers were built for Hawaiian guitars, or as they are now called, lap steel guitars. Soon the amplifiers were adopted by jazz guitarists, since luthiers had gone about as far as they could go to help the guitar be heard in a brass/woodwind/percussion jazz band. (The arch-top guitar and the resonator guitar were created in an attempt to help the guitarist in these loudness wars.) In Hawaiian, "country and western," and jazz music, there was no desire to change the sound of the guitar—just to make it louder. Thus, distortion was avoided as much as possible.

Photo 1: These early Kustom amplifiers were well known in the mid-1960s.

Fast forward to 1962, the year I got my first electric guitar and amplifier. A distorted guitar sound was a sign that you could not afford a decent amplifier—as was my case. As soon as I could come up with the funds, I upgraded by replacing my no-name guitar amplifier (which used loctal tubes and 20-year-old technology when I bought the used amplifier and guitar for $15) with a Silvertone Twin Twelve amplifier (advertised as 50 W, using two 6L6GCs in push-pull). I noticed that when I pushed the volume control a bit above 5, a nice edge was added to the sound for Ventures-style lead riffs, I also noticed that I did not like the sound as well when I played chords because the distortion seemed unpleasant. At that time, very little if any distortion was heard on guitar recordings.

A few years later, the folk-rock group the Byrds heard Ravi Shankar's sitar music and introduced it to their friend George Harrison of the Beatles. The sitar's sound is rather buzzy, reminiscent of a guitar with a warped neck and fretted about the seventh fret, so the strings buzzed against the frets. Previously, guitarists hated this sound and went to great lengths to avoid it. However, the sitar's incorporation into Western pop music built an acceptance of this buzzy sound quality.

It is my theory that the sitar, plus the buzzy sound of some acoustic slide guitar players who became popular during the Folk Era, led musical electronics manufacturers to try to create an effect that would mimic the sound; hence, the first intentional distortion effects for guitar. (This theory is not musicologically rigorous in its derivation, but it is based on the first-hand observations of a then-teenaged amateur guitarist.) The Hawaiian, country and western, and jazz guitarists continued to prefer the clean sound.

Photo 2: The Electro-Harmonix LPB-1, introduced in the late 1960s, boosted signal levels to overdrive amp input stages.

Early Distortion Efforts

At first, there were three ways to create distortion for electric guitar: using outboard effects (i.e., "fuzz boxes"), overdriving the amplifier's input stage (often by using a booster amplifier) and just playing really loud. The second option sometimes involved daisy chaining two amplifiers and feeding the output of the first into the input of the second. The third option was increasingly adopted as acid rock music came into vogue.

In the mid-1960s, Kustom Electronics became a prominent manufacturer of solid-state guitar amplifiers (see **Photo 1**). Guitarists were offered a choice between vacuum-tube distortion and solid-state distortion. Solid-state distortion generally contains stronger upper harmonics than tube distortion, which is mainly second and third harmonics. The upper harmonics add a more aggressive, buzzing sound. The external fuzz boxes all used solid-state circuits and, in spite of valiant attempts to mimic tube distortion, none really succeeded.

The most successful external accessory, in terms of enhancing tube distortion, was probably the Electro-Harmonix LPB-1 (see **Photo 2**). The LPB-1 was a single-transistor preamplifier designed to connect between the guitar and the amplifier's input stage, thus permitting a more overdriven sound from the amp. The LBP-1 did not distort the guitar sound, instead it provided a maximum gain of about 30, which would easily drive almost any guitar-amplifier input stage into hard distortion.

Distortion Basics

Before going any further into differences in amplifier distortion, let's examine the basics of distortion. Distortion is defined as any process that causes the output signal of a device to differ from the input signal in any way except amplification. Technically, this definition would include the action of tone controls. Indeed, some texts refer to "frequency distortion," but what we usually mean by distortion is nonlinear distortion. This is distortion of the waveform by the nonlinear action of an electronic device (e.g., a vacuum tube or transistor) or by an electroacoustic device (e.g., a speaker). Although some guitar-amplifier speakers have been designed to produce some distortion (usually even-order harmonic distortion resulting from mis-centering the voice coil windings in the magnetic gap), most guitar-amplifier distortion is produced electronically.

If we graph an amplifier's instantaneous output voltage resulting from a specific input voltage, the graph is called the amplifier's "transfer function." The slope of the transfer function corresponds to the amplifier's gain. **Figure 1** shows transfer functions for a perfect amplifier and a real (imperfect) one.

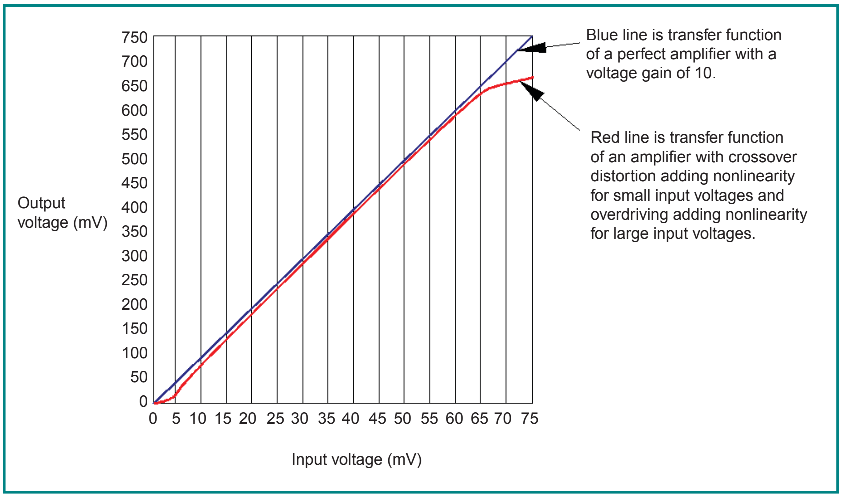


Figure 1: Amplifier linearity is represented by the straightness of the transfer function.

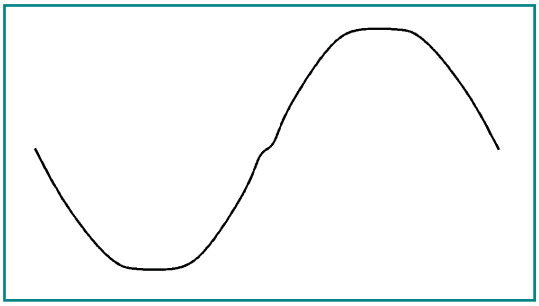


Figure 2: Nonlinearities in the transfer function distort the output waveform.

The nonlinearities in the red curve shown in **Figure 1** would cause "squashing" of the waveform at the top and bottom and "notching" where the output crosses over from the bottom to the top tube of the push-pull pair (see **Figure 2**).

Jean Baptiste Joseph Fourier, a French mathematician from the late 18th and early 19th centuries, developed a theorem that states any waveform can be deconstructed into a fundamental-frequency sine wave and additional sine waves (harmonics) at multiples of the fundamental frequency. Another way of looking at Fourier's theorem is that it says when we distort a waveform we add additional frequencies at harmonics of the fundamental. This is called "harmonic distortion." Since the additional frequencies it adds are multiples of the fundamental, they correspond to notes that are one octave, one octave-and-a-fifth, and so forth, above the fundamental frequency. In other words, the added notes are in tune with the original note. So a 1-kHz wave with harmonic distortion would contain the frequencies 1, 2 , 3 kHz, and so forth. Normally, the amplitudes of the harmonics decrease as the frequency increases.

The other kind of nonlinear distortion is called intermodulation (IM) distortion. In this kind of distortion, there are two or more original frequencies and the additional frequencies that are created correspond to the sum and difference of the original frequencies. Thus if the original frequencies were 740 and 1,047 Hz, the output would contain components at 307 (the difference frequency), 740, 1,047 (the original frequencies), and 1,787 Hz (the sum frequency).

Notice that these frequencies are not in octave relationships (factors of 2) to the original frequencies, so they sound like wrong notes. In this case, the original frequencies correspond to an F# and a C, which could be part of an F# diminished chord. The sum and difference frequencies correspond to somewhere between an A and an A# and between a D and a D#, respectively. Also they are not at the right frequencies for any notes. Not only do they not fit in an F#-minor chord, they do not fit in any chord. They are out-of-tune. Thus, IM distortion is not usually considered desirable.

The difficulty is that any circuit that produces nonlinear distortion produces both harmonic and IM distortion, so we can't really have one without the other. However, some types of nonlinearities produce a relatively lower IM compared to the harmonic distortion. These are usually considered to sound more musical.

Figure 3 shows a sine wave (sine wave A). **Figure 4** shows sine wave A added to sine wave B, which has one-tenth the frequency of sine wave A. **Figure 5** shows sine wave A multiplied by sine wave B. The multiplication process represents the creation of IM distortion.

Distortion Methods

Since the good amplifiers were designed to be clean, outboard devices (e.g., the LBP-1 mentioned earlier) were introduced to produce a distorted sound without playing really loud. Next came fuzz boxes containing circuits ranging from diodes bypassed with potentiometers, to transistor preamplifers, which were designed to be overdriven by the guitar signal, to digital gates. These outboard devices produced large amounts of IM, as well as a high proportion of higher-order harmonics, such as 10th and 12th.

Next, amplifier manufacturers added high-gain channels, some with channel switching, and master volume controls enabling stages after the preamplifier to be overdriven or not, according to the master volume setting. Finally, outboard attenuators appeared that were connected between the amplifier and the speaker, enabling the amplifier to be operated at near-maximum power levels, while the speaker was not putting out too much sound.

Each of these methods produced its own characteristic distortion sound. As for the amplifier itself, its characteristic distortion sound depends on the circuitry of the stage(s) where the distortion occurs: a transistor or op-amp, single-ended triode, single-ended pentode, phase splitter (e.g., paraphase, long-tailed pair, or split load), or output stage.

Different models of output tubes have different effects on the distortion sound. So unlike home music systems, in which the majority opinion is that distortion is bad and should be eliminated, in guitar amplifiers, there is a whole science about what is the most desirable distortion sound (individual opinions vary widely) and how to create that sound.

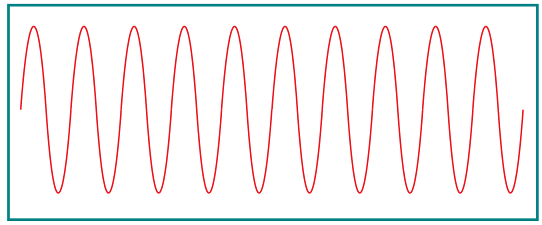


Figure 3: This is "sine wave A."

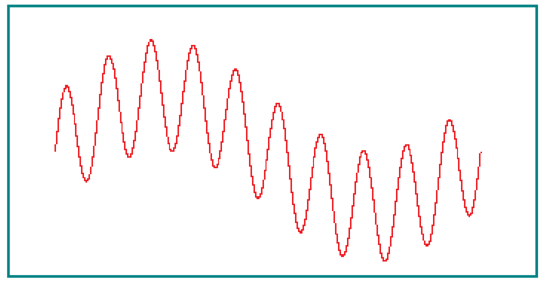


Figure 4: This is sine wave A added to sine wave B, which has one-tenth the frequency of sine wave A.

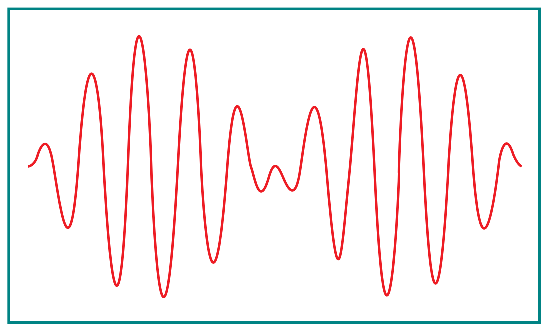


Figure 5: This is sine wave A multiplied by sine wave B, producing IM distortion.

Member Profile

Member Name:
Peter Delos

Location:
New Jersey side of Philadelphia, PA

Education:
Peter has two electrical engineering degrees—a BSEE from Virginia Polytechnic Institute and State University (Virginia Tech), 1990, and an MSEE from the New Jersey Institute of Technology (NJIT), 2004.

Occupation:
Peter is an RF circuit design engineer, which is a specialized area of electrical engineering.

Member Status:
Peter said he has been a member on and off for several years and recently renewed his subscription to *audioXpress*.

Affiliations:
Peter said does not currently have any professional affiliations. However, he said he should rejoin The Institute of Electrical and Electronics Engineers (IEEE).

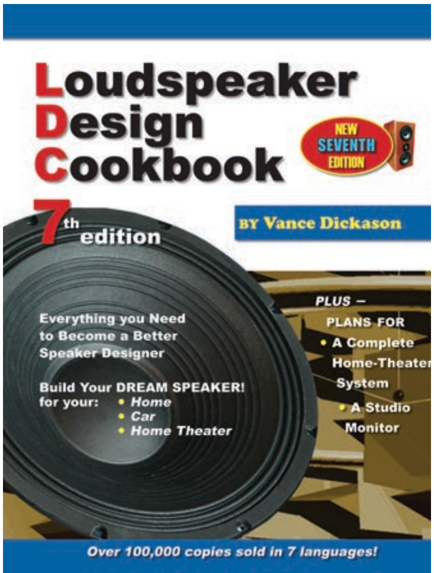
Audio Interests:
Peter plays the guitar and initially became interested in audio work while trying to optimize the sound for gigs and recording. He enjoys experimenting and likes that you can set up a home bench for a reasonable cost. His audio interests include guitar amplifier design, guitar effects, and stereos.

Most Recent Purchase:
Peter just bought *Audio Power Amplifier Design*, written by Douglas Self.

Current Audio Projects:
He is working on a custom tube-based guitar amplifier.

Dream System:
Peter said he wants a system that will bring in enough money to pay for all the parts and time expended.

AX Shop



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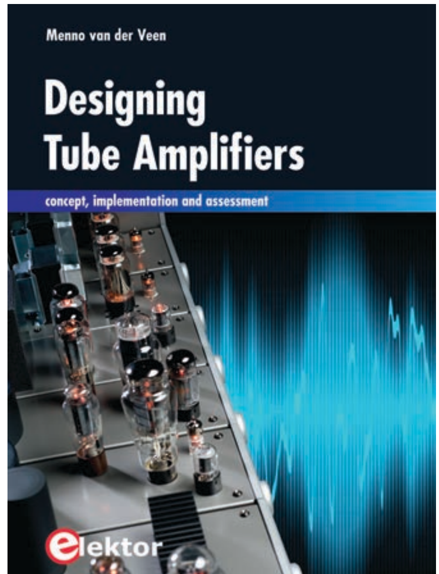

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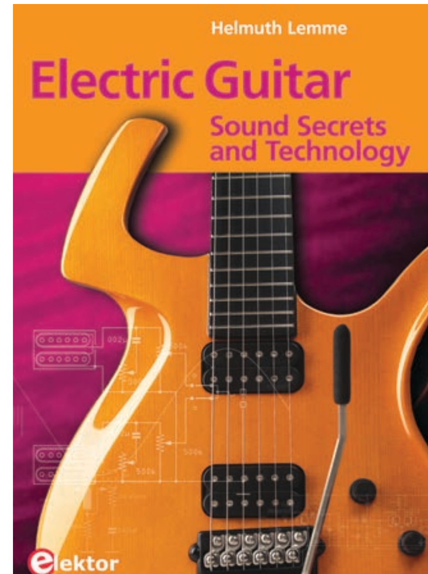

3

1 LOUDSPEAKER DESIGN COOKBOOK

This book's 7th edition provides you with everything you need to become a better speaker designer. It now includes Klippel analysis of drivers, loudspeaker voicing, advice on testing and crossover changes, and so much more! It is an excellent reference for loudspeaker engineering professionals.

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4

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Industry Calendar

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

March 12–15, 2014 — Musikmesse Prolight+Sound Frankfurt

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

This is by far the world's largest professional show dedicated to music instruments, music technology, and professional audio. *audioXpress* will attend in full force covering all the audio novelties in the Prolight+Sound halls and the technology-oriented halls of the Musikmesse, where "the return of the analog synths" is apparent. This is also the place to look for acoustical solutions, recording studio equipment, and a lot of sound reinforcement solutions. In fact, we believe no other show in the world unites as many audio companies from different segments as the annual Frankfurt event. Look for *audioXpress* in the International Press distribution areas.

April 5–10, 2014 — NAB Show 2014

Las Vegas Convention Center, Las Vegas, NV, US
www.nabshow.com

The National Association of Broadcasters (NAB) Show is one of those not-to-be-missed opportunities where you can find many of the most important audio companies involved in field recording, studio, post-production, film sound, consumer distribution technologies, acoustics, and so forth in one location. The NAB Show is typically one of the most vibrant, exciting, and diversified exhibitions for the professional audio industry. Plus, it's certainly the most important annual event for the radio industry. With digital radio and new distribution models on mobile networks and online streaming on the horizon, radio certainly will be one of the hot topics for 2014. Most of the audio companies will be in the Central Hall, adjacent to registration and the main LVCC entrance.

April 26–29, 2014 — 136th International AES Convention

Estrel Hotel and Convention Center, Berlin, Germany
www.aes.org

The third Audio Engineering Society (AES) International Convention will include a full technical program of papers, posters, e-briefs, tutorials, and workshops covering a comprehensive range of audio-related topics, and feature top industry professionals. The 136th AES Convention, co-chaired by Sascha Spors and Umberto Zanghieri, will bring audio engineers from around the world to share the latest knowledge in audio research, development, and applications. Additionally, the Berlin convention will strongly focus on sound for picture, recording, and broadcast aspects of audio engineering, in addition to the popular off-site technical tours and social events.

Following the success of the Project Studio Expo (PSE) at AES conventions in San Francisco, CA, and New York, NY, the PSE will make its European debut at the 136th AES Convention. A special technology showcase will also provide participating companies with a chance to directly interface with interested end users and customers.

May 13–15, 2014 — 18th AES Brazil Expo

Expo Center Norte Pavilhão Amarelo, Vila Guilherme, São Paulo, Brazil
www.aesbrasilexpo.com.br

The Audio Engineering Society (AES) show in Brazil is unlike any other AES convention held in the US or elsewhere. In fact, even though the conference sessions follow the traditional association model—almost 100% in Portuguese—the exhibition floor is predominantly dedicated to sound reinforcement and PA systems due to the importance of that professional audio segment in the Brazilian market, which is one of the world's largest music markets. The event, held in the megalopolis of São Paulo—Brazil's business capital—is an important annual gathering for the domestic pro audio industry, including speaker component and amplification manufacturers. In fact, AES Brazil is the main pro audio event in South America and provides an opportunity to find local partners and learn more about Brazil.

June 14–20, 2014 — InfoComm 2014

infoComm INTERNATIONAL®

Las Vegas Convention Center, Las Vegas, NV, US
www.infocommshow.org

The audiovisual communications marketplace, as the InfoComm association defines this major event, increasingly represents the intersection of AV and IT, technology and architecture, business and entertainment, government and hospitality, healthcare and education, live events, and installed systems. InfoComm is undoubtedly the most important professional AV show on the American continent. For 2014, more than 950 exhibitors from 300-plus audio companies and 35,000-plus attendees from more than 110 countries are expected. InfoComm 2014 will be provide a unique opportunity to see the latest technology, learn new skills, and grow our professional network.


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