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Balanced or Unbalanced?

Some *audioXpress* readers and subscribers have been expressing their disbelief that this publication is now devoting several of its pages to "professional audio" equipment, often worth tens of thousands of dollars. Others have praised the fact that we are helping them rediscover a passion for audio engineering, recording, acoustics, and the roots of good sound. Some question whether we are going to abandon our "DIY" roots of audio electronics, stating, "If it's not something I can afford for my living room, it's not for me."

The questions remind me of the Warner Bros. cartoon in which Elmer Fudd asks Bugs Bunny (who is in disguise), "Say, have you seen a wabbit wun by here?" To which, Bugs Bunny responds, "Say, don't he have, uh, great big long ears [and he pulls his ears] like this? And does he hop around like this? And does he say, 'What's up, doc?' like this?" Bugs Bunny finishes with, "Nope, never heard of him."

When I'm asked to explain the concept behind *audioXpress* magazine, I begin by saying we are about audio electronics, DIY audio, and audio innovations. Next, comes the debate about whether the magazine is for professionals or consumers, to which I usually answer: Both "enthusiasts" and "audio engineers."

Usually I receive that Elmer Fudd "have you seen a wabbit?" look.

The point is, people tend to assume that "consumers" are pure "listeners," while "pros" actually get their hands on the stuff. But building speakers and amps or recording and mixing sounds doesn't really distinguish between "pro" or "consumer."

Many recording equipment manufacturers target "consumers" as their home-studio clients, wannabe-DJs, or simply kids who download free apps to experiment with music or audio recording on their smartphones. And guess what: when we look at their budget equipment we can see balanced connections!

When we attend one of those hotel shows for "listeners," where the terms "hi-fi" and "home stereo" are still used, and intertwined with "lifestyle" and "mobile," we don't need to look at the connections to know the majority is unbalanced.

Mentioning "balanced" or "unbalanced" immediately triggers that "magic" discussion about the merits of voltage levels, impedance, electromagnetic interference, crosstalk, induced noise, and differential modes. Suddenly, whether we are among "listeners," "users," or "builders," we find a common passion about audio and the never-ending quest for sound purity.

At which point, inescapably, I get the question: "So, balanced or unbalanced? That's where you draw the line?"

To which I have to summon my Bugs Bunny glance and reply—"Eh... No."

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.

Celestion Celebrates 90 Years of Loudspeaker Manufacturing

During the Prolight+Sound 2014 show in Frankfurt, Germany, British company Celestion celebrated 90 years of loudspeaker manufacturing with a large party attended by many distributors, consultants, clients, and press.

A speaker-shaped cake was cut by Dee Potter, Celestion's production manager, who celebrated 40 years working at Celestion that same day. According to Nigel Wood, Celestion's managing director, Potter built or supervised the building of more than 8 million speakers during her time with the company.

The company reached this landmark year as one of the world's leading suppliers of premium guitar speakers and, after shipping almost 500,000 compression drivers in the past 12 months, one of the world's leading professional audio component suppliers.

Celestion opened in 1924 as a small company, designing and building loudspeakers for the radio market. In 1932, a permanent magnet moving coil loudspeaker with improved sound quality (compared to existing technologies) was developed and patented. By 1946, Celestion was manufacturing more than half the loudspeakers in Great Britain.

During the late 1950s, Celestion introduced what is today known as one of the most famous guitar speakers of all time: the Celestion Blue. A toughened-up version of an existing general-purpose 12" loudspeaker that had been in production for more than a decade, the Blue was an immediate hit with many of the up and coming musicians during this period, including The Beatles.

Since then, the company has been at the forefront of loudspeaker development, continuing to develop world-leading guitar speakers, and embarking on the development of professional audio drivers to meet the growing requirements of sound reinforcement in all kinds of venues.

Today, Celestion has a growing reputation for creating advanced products; often incorporating patented innovations, a direct result of having the largest loudspeaker transducer engineering team in the pro audio industry.

With its worldwide headquarters in Ipswich, England, Celestion regularly supplies compression drivers, pressed steel and die cast chassis woofers, and coaxial and full-range speakers to some of the leading names in sound reinforcement system manufacture. Celestion is recognized as the premium guitar speaker supplier, but also one of the major suppliers of professional audio components.

Celestion
www.celestion.com

Celestion celebrated 90 years of loudspeaker manufacturing with a large party during the Prolight+Sound show in Frankfurt, Germany. Dee Potter, Celestion's production manager, also celebrated 40 years of working at Celestion, attending the Frankfurt show for the first time and cutting the speaker-shaped cake (a). Celestion's worldwide headquarters is located in Ipswich, England (b).

New NTi Audio XL2 Data Explorer PC Reporting Software

NTi Audio has released its new Data Explorer PC reporting software for precision-grade noise measurements. With a design inspired by users of the XL2 acoustic analyzer, the Data Explorer offers clear visualization, fast data analysis, and professional reporting of environmental noise measurements.

The Data Explorer software visualizes the entire data set including spectrogram. It provides fast zoom and pan response, even for data sets with millions of data points, creating professional reports, customized reports with titles, comments, and personalized logos with a mouse click. Audio files are time-aligned to the graph for instantaneous playback including proper leveling. "Include Markers" enable level calculations in specific areas of interest while "Exclude Markers" eliminate unwanted areas from the level calculations. All measurement results are immediately recalculated after Markers are set, which is a useful tool for "What-if" analysis.

The XL2 Data Explorer PC software automatically adds relevant header data (e.g., the measurement data, calibration information, instrument setup, etc.) to produce a tailored and professional-looking report.

"The introduction of the Data Explorer software is an important milestone for NTi Audio as it is an evident tool for professionals to manage and report on large data sets," comments Philipp Schwizer, CEO of NTi Audio. "The very positive feedback from the market confirms that we are meeting a big demand with this package."

The installation of XL2 Data Explorer software on multiple PCs provides consultants with the freedom to distribute their measurement projects among team members and clients. The data import into the software is enabled by an option installed in the XL2 sound level meter. The installation of the XL2 option requires firmware V2.72 or higher. Legacy data import recorded prior the option installation is supported.

The XL2 Data Explorer is an ideal software tool for acoustic consultants and noise measurement professionals.

NTi Audio
www.nti-audio.com

Radial Engineering Introduces the Headload Guitar Amplifier Attenuator

Radial Engineering introduced the Headload, a guitar amplifier attenuator that enables the artist to reduce the volume levels on stage or during practice, while driving the amplifier hard for maximum tone.

According to Radial Engineering President Peter Janis, "With the proliferation of in-ear monitors, there has come a tremendous need for reducing sound pressure levels on stage. Outside of drums, the next biggest culprit is the guitar amplifier. To get tone, guitarists like to drive their amps hard. This is particularly true for smaller amps that sound great when pushed to the max. Having tried all kinds of load boxes—we have never been all that impressed with the ones currently on the market. Although they all reduce the sound level coming out of the amp, they fall short when it comes to producing a usable signal for the PA system. We felt that if we could leverage the incredible success we have enjoyed with the Radial JDX direct box and combine this with the Phazer phase adjustment tool, a new type of load box could be created. After a year of trial and error, we are pleased to announce the Headload—the culmination of our quest for great amp tone."

The fan-cooled design begins with a series of high-power cement-encrusted epoxy-coated copper coil resistors used with a six-position Grayhill rotary switch to dissipate the power generated by the amplifier. A variable range control enables incremental power reduction at low levels. This is augmented with separate high- and low-resonance switches. Power to the speakers may be turned off for quiet recording or to eliminate the speaker cabinet on stage.

In addition to the 8-Ω speaker cabinet outputs, the Headload comes with a built-in Radial JDX direct box. This is coupled with a six-position voicing switch. It is supplemented with a two-band EQ for fine-tuning and a low-pass filter to eliminate overly harsh harmonics that are produced by some amplifiers. Two JDX direct outputs offer the choice of pre- or post-EQ settings to enable the artist to control his wedge or in-ear monitors while sending a nonequalized tone to the front of house mix position. For engineers who prefer to combine a microphone with a direct feed, the Headload has a Radial Phazer, which is a phase alignment tool. This enables the stage tech to align the direct feed with the microphone and dial-in the ideal mix on stage. In the studio, the Phazer can be used for creative equalization (e.g., creating Boston-type out-of-phase rhythm tones or an ultra-thick wall-of-sound). A second set of 0.25" line outputs may be used to feed additional stage amplifiers or effects if needed.

Radial Engineering, Ltd.
www.radialeng.com

a)

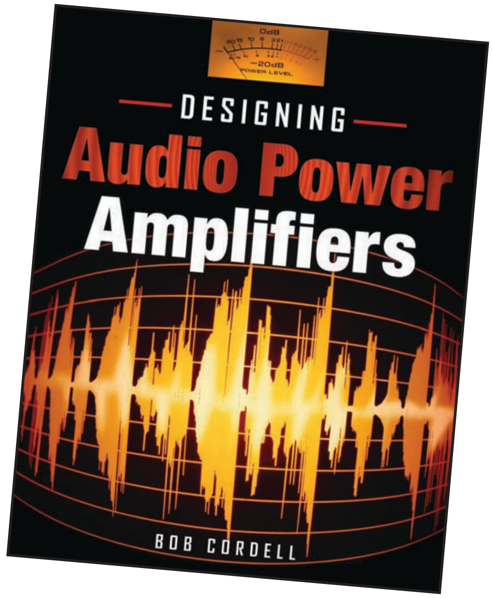
b)

c)

Radial Engineering's Headload guitar amplifier attenuator's front (a) and back (b) panels. The design begins with a series of copper coil resistors used in conjunction with a six-position Grayhill rotary switch (c).

Designing Audio Power Amplifiers

A Review of Bob Cordell's Audio Reference Book



This month we explore another great audio amplifier design reference book. Designing Audio Power Amplifiers by Bob Cordell provides a thorough, well-rounded tutorial on the intricacies of audio power amplifier design. It emphasizes the art of analog design. Those entering the audio field will find it helpful when learning the fundamental elements needed to understand amplifier design. The author also provides considerable detail about advanced techniques and modern methods, making this book useful for the experienced designers. Tube amplifiers are not discussed, but the book provides the essentials for discrete transistor-level power amplifier design.

By
Peter Delos
(United States)

Bob Cordell, who wrote *Designing Audio Power Amplifiers*, found his enthusiasm for audio circuit design as a teenager building Heath kits, test equipment, and tube amplifiers. His accomplished professional career includes more than two decades at Bell Labs, and he has worked with many great people through the evolution of the electronic age. Cordell is active in audio DIY groups and has published articles in the *Journal of the Audio Engineering Society* (*JAES*).

Cordell's book intentionally connects with those less experienced while covering a lot of material in sufficient depth to be valuable to more experienced engineers. Cordell is clearly dedicated to helping others and improving the art of amplifier design. His website (www.cordellaudio.com) provides design notes and SPICE models to supplement the book.

Designing Audio Power Amplifiers begins with a six-chapter tutorial introducing the fundamental discrete transistor building blocks and the overall power amplifier architecture. Operation of differential pairs, current mirrors, current sources, and V_{BE} multipliers are intuitively explained. The book also explores the use of these blocks within the power amplifier. Options for the output stage are evaluated along with amplifier class definitions. Feedback and stability is explained along with practical considerations in the power amplifier design process.

Once Cordell establishes the foundation, he delves deeper into more advanced topics. He provides a detailed evaluation of options for the input stage and voltage amplifier stage. The output stage is examined in further detail with specific consideration of crossover distortion. Field-effect transistors (FETs) are compared with bipolar junction transistors (BJTs) and a FET-based output stage implementation is thoroughly examined. Additional methods for feedback compensation and stability considerations are presented. Sources of distortion throughout the entire design are discussed and advanced error correction methods are presented.

Modeling methods are allocated two chapters. An introduction to using Linear Technology's LTspice is presented along with ways to develop accurate transistor models. Cordell's website serves as a great supplement to the book by providing many transistor SPICE models along with models of complete amplifier circuits.

Audio test methods and test equipment are considered and also allocated two chapters. The first chapter in this section introduces the basic equipment and provides information about circuits for speaker emulation. This section's second chapter

Designing Audio Power Amplifiers
Bob Cordell
McGraw-Hill/TAB Electronics, 2010
ISBN-10: 007164024X
ISBN-13: 978-0071640244
640 pages
www.cordellaudio.com

is dedicated to distortion measurements.

An entire four-chapter section focuses on Class-D audio design. Class D offers efficiency and is widely used in mass-produced commercial applications. Class D is implemented with varying forms of pulse width modulation (PWM) and the output transistors are used as switches rather than linear devices. The disadvantages to date include distortion, poor power supply rejection, and overall inferior perceived audio quality. Many audiophiles may dismiss Class D. However, the efficiency advantage has resulted in large financial backing of the semiconductor industry and detailed electrical engineering techniques continue to improve the Class-D performance.

Cordell systematically introduces the Class-D operation, the benefits, the limitations, and the trends for future development. He cites many recent references. Anyone working in this field will benefit from the Class-D discussion provided.

An obvious comparison can be made between Cordell's book and Douglas Self's book *Power Amplifier Design*. In my opinion, the books are very complementary. Rather than compete about the same topic, both authors present their own versions. The compilation of these two sources provides an incredibly detailed view for anyone working in or entering audio power amplifier design. In fact, the use of both books could form the basis for an advanced university-level electrical engineering design course on audio power amplification.

In summary, *Designing Audio Power Amplifiers*, is a great leading-edge reference. Anyone working in audio design should have this readily accessible. People in adjacent discrete transistor level analog design fields—whether on a circuit board, or inside an IC—will also find the analog design methods useful. Those entering into power amplifier design will find this reference book provides a great tutorial to quickly learn the essentials needed for proficiency.

About the Author

Peter Delos is a Radio Frequency (RF) Circuit Design Engineer at a major electronics company in the greater Philadelphia, PA, area. He is also a guitar player and began modifying and designing guitar amplifiers for gigs and recordings when he couldn't buy what he wanted. He does guitar amplifier repairs, modifications, and custom designs for local musicians and friends.

Power Amplifiers

Another Perspective

audioXpress author Peter Delos recently reviewed two books about power amplifier design, *Audio Power Amplifier Design* by Douglas Self (*audioXpress*, May 2014) and *Designing Audio Power Amplifiers* by Bob Cordell (*audioXpress*, July 2014). Both books are essential references for anyone interested in the subject. This article discusses a third perspective on power amplifiers, which John Dawson presented at the 135th convention of the Audio Engineering Society (AES) in October 2013.

By
Gary Galo
(United States)

Photo 1: Respected audio designer John Dawson co-founded A & R Cambridge, which is now known as ARCAM.

John Dawson, a respected audio designer, presented a 2-h tutorial "Audio Power Amplifier Design Revisited: From Picowatts to Kilowatts—A Practical Guide to Driving Headphones and Loudspeakers Properly."

Dawson co-founded A & R Cambridge in 1972, along with several other science and engineering students at the University of Cambridge (not to be confused with the UK firm Cambridge Audio). The company's first consumer product, which was introduced in 1976, was the A60 integrated amplifier. The amplifier's circuitry was contained on a single PC board. A & R Cambridge initially built 50 units, but the amplifier was acclaimed for its excellent sound quality and 32,000 units were eventually sold.

The company was renamed ARCAM in the 1980s. Since the introduction of the A60, Dawson has served as managing director, then chairman, and in 2004 he became its president. During his tenure as president he has been actively involved in electronics engineering for the company. Dawson entered semi-retirement late last year, but he remains a consultant to ARCAM and continues to be active in the analog and mixed-signal areas of product design. He also works as an electronics consultant to other firms (see **Photo 1**).

Dawson began his tutorial by saying high-resolution digital audio is here to stay and that a 120-dB dynamic range is required from an audio system to handle today's high-resolution sources. He noted that 1 kW into 8 Ω equals 89.4 V RMS. Therefore, the level at –120 dB will be 89 μV, or 1 nW (10^{-9}W) into 8 Ω. It's difficult to achieve a 120-dB SNR in a complete audio system, since noise will be influenced by the preamplifier source impedance and volume control. It's also difficult to achieve low distortion free of high-order harmonics over such a wide range. "Green" issues can't be ignored these days, so amplifier efficiency must also be considered. Electromagnetic compatibility (EMC) is also an important concern, in terms of radiation by the amplifier and its susceptibility to propagation from outside sources. Dawson also stated that analog electronics is generally not taught at universities.

Amplication Overview

Dawson provided an overview of the various classes of amplification, including single-ended Class A and push-pull Class A, B, and AB. He briefly touched on Class-D switching amplifiers. He also covered Class AB/G, where the rail voltages on the

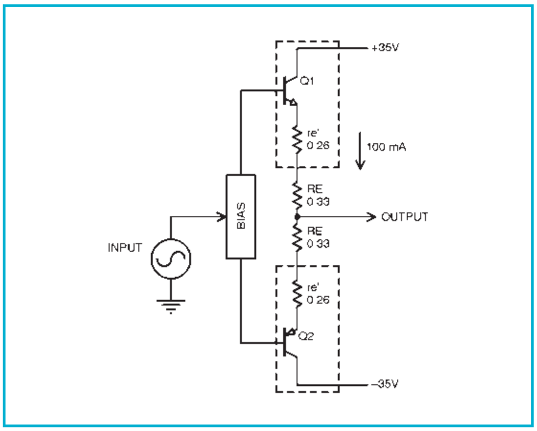


Figure 1: In his tutorial about audio power amplifier design, John Dawson used this schematic from Bob Cordell's book *Designing Audio Power Amplifiers* to illustrate the importance of considering the emitter resistance of the output devices (re´) when determining the value of the external emitter resistors (R_E). (Image courtesy of Bob Cordell)

output transistor collectors (in an emitter-follower configuration) normally run low and are switched high when required by signal levels. He said that some distortion is produced in the switching, but that it is normally inaudible.

The advantage of this scheme is even greater efficiency than conventional Class AB. Dawson briefly discussed power amplifier architecture, illustrating a basic, three-stage amplifier with a differential input, current source and current mirror, a single-ended voltage-gain stage with current source, and a push-pull emitter-follower output.

Dawson observed an amplifier's performance characteristics include SNR, frequency response, and various types of distortion. Loudspeakers and headphones are not resistive loads and an amplifier's output behavior must be load-tolerant. An amplifier's frequency response must be phase-linear across the audio band and independent of load. The output impedance must be significantly less than the load Z, so the ratio of output impedance to load impedance is at least 100. He also noted problems caused by Zobel output networks. Dawson presented graphs of several real loudspeaker loads, illustrating impedance and phase shift. Impedance variations of 6:1 are typical, along with phase shifts of up to 60°.

Dawson cited the Cordell and Self books, adding that together they give a great overview of the subject. Since they are rather different, he believes that they are both required reading. He noted that Self lists 11 distortion mechanisms in power amplifiers, but Dawson confined his discussion to the three he felt were particularly important: crossover distortion, rail induction, and premature overload triggering. (He assumed that "gross" problems such as slew-rate limiting and instability would be absent.)

He emphasized the importance of proper selection of the emitter resistor values in a push-pull Class-AB output stage, aided by an illustration from Cordell's book (see **Figure 1**). Dawson stressed the need to consider the emitter resistance of the output devices (re´) when determining the value of the external emitter resistors (R_E). He also said that for lowest crossover distortion 0.1 Ω is an ideal value for the emitter resistors, which moves the voltage gain closer to unity. The downside is higher quiescent current consumption and drift.

Dawson cautioned against low-impedance loudspeaker loads and recommended using "faster and better" bipolar junction transistor (BJT) transistors. Using graphs from Self's book, he discussed output–stage gain linearity in BJT vs. field-effect transistor (FET) output stages. In these illustrations, the FETs fared worse. Dawson said he did not recommend FETs in output stages. This view no doubt raised a few eyebrows at the well-attended session, since FET output stages have a devoted following.

Headphone Requirements

Dawson also discussed headphone requirements. He observed that most music is now consumed via headphones of various types. Unfortunately, and unlike loudspeakers, impedance is not standardized and varies widely, with Grados at 30 Ω and others covering the entire range up to 300 Ω.

Moving-coil types have more uniform impedance with frequency, but the balanced-armature, in-ear types are the worst in this regard. The latter are quite inductive at high-frequencies. The worst example he cited has an impedance curve varying from 7.7 Ω at 10 kHz to 97 Ω at 1 kHz. All these requirements make it difficult to design a "one size fits all" headphone amplifier.

The International Electrotechnical Commission (IEC) standard for measuring headphones calls for a 120-Ω source impedance. While this may equalize levels, frequency response variations are problematic. An ideal amplifier should be able to deliver 5 V RMS to the load, and in some cases this needs to be as high as 10 V RMS. The noise level for the most sensitive headphones should be less than 0.3 µV, which is nearly impossible.

Dawson pointed out the considerable variability in the performance of the low-voltage, low-power

About the Author

Gary Galo retired in June 2014 after 38 years as Audio Engineer at the Crane School of Music, SUNY, in Potsdam, NY. He now works as a volunteer, transferring vintage analog recordings in the Crane archive to digital format. He is the author of more than 280 articles and reviews on both musical and technical subjects. Gary is an active member of the Association for Recorded Sound Collections (ARSC) and has been a regular presenter at ARSC conferences. Several of his conference papers have been published in the *ARSC Journal*.

consumption ICs often used in battery-powered equipment. He believes, these days, it's not too difficult to design a decent mains-powered headphone amplifier if you can afford to use quality parts. It's harder when restricted to 3.6-or-5-V supplies.

Passive Components

During his tutorial, Dawson emphasized the importance of passive components, and his recommendations were consistent with what many in the audiophile community have known for a long time. He stated that resistors have temperature and voltage coefficients, and that the exact value is somewhat dependent on the instantaneous voltage across the resistor and the power dissipated (most resistor manufacturers don't specify voltage coefficient). Carbon film and composition types are to be avoided, metal foils are best but expensive, and metal film types are a good compromise.

For small value capacitors up to 100 nF, he prefers polystyrene followed by polypropylene. The only acceptable ceramics are the NP0 and CP0, which have low temperature and voltage coefficients. Like many of us, Dawson does not like electrolytic capacitors and prefers to do low-frequency bandwidth limiting with DC servos built with polypropylene capacitors. He also emphasized the importance of star grounding.

Parts Preferences

The presentation included recommendations of a few favorite parts. Dawson favors ON Semiconductor's NJL3281D (NPN) and NJL1302D (PNP) complementary ThermalTrak power transistors, which have been designed to "eliminate thermal equilibrium lag time and bias trimming in audio amplifier applications." These 15-A, 260-V, 200-W bipolar devices have an on-board, thermally matched bias diode, which provide instant thermal bias tracking, and are manufactured in five-lead TO-264 cases. The manufacturer notes their "high safe operating area" and "superior sound quality through improved dynamic temperature response," and recommends them for high-end consumer audio products.

Dawson also recommended Texas Instruments's LM4702 stereo power amplifier driver IC, which is manufactured specifically for high-performance audio applications. The device's output power can be scaled by changing the supply voltages and the number of output devices. It can be powered from a supply voltage up to 200 VDC (±100 V), and is capable of delivering more than 300 W per channel into 8 Ω. With maximum allowable rail voltages, slew rate is specified as 17 V/μs, and total harmonic distortion (THD) is 0.0003% at 1 kHz with an output voltage of 20 V RMS. The LM4702 is manufactured in a 15-lead TO-220 package.

He concluded with a photo of one of the power amplifiers in ARCAM's AVR600 seven-channel A/V receiver built with the Texas Instruments driver ICs and ON Semiconductor output devices, operating in a Class-AB/-G configuration.

Dawson provided an excellent overview of his power amp design philosophy. My brief description barely scratches the surface. This was an extremely worthwhile presentation for anyone interested in the subject.

Resource

University of Cambridge, www.arcam.co.uk.

Sources

AVR600 AV Receiver
ARCAM | www.arcam.co.uk

NJL3281D (NPN) and NJL1302D (PNP) Complementary ThermalTrak power transistors
ON Semiconductor | www.onsemi.com

LM4702 Stereo power amplifier driver IC
Texas Instruments, Inc. | www.ti.com

Cinema Sound

Sound for Pictures in Motion

From an audio perspective, movie theaters and residential cinema applications have something in common. They use more speakers and more channels per speaker than any other application. In cinema reproduction, multichannel is the key word. Even though the cinema speaker market has not been immune to recessions and technology disruptions, it continues to grow worldwide and expand in new ways. With the move from surround sound to immersive audio, even more speakers are needed. Here is an overview of the cinema sound market.

By
João Martins
(Editor-in-Chief)

Cinema sound is probably one of the least discussed topics in the speaker industry and still it "speaks volumes." For those professional audio manufacturers involved in the commercial theater and residential cinema markets, in particular, it remains a solid revenue source. It's not exactly a very competitive landscape.

Philips Electronics's earlier wireless home cinema solutions, such as its Fidelio range, appeal to today's digital lifestyle.

This is in sharp contrast to the consumer home-cinema market, which basically "imploded" over the last few years. It has become saturated with low-margin "multichannel-in-a-box" solutions. Even the high-end home cinema segment suffers from a technology transition caused by the demise of high-quality physical media and from consumers refocusing their priorities.

High-end home theater and residential installations remain an interesting revenue source for integrators (e.g., the US and Canadian markets) because they involve more than just projectors, large screens, and multichannel speakers. The market also includes acoustic treatment, special seating, remote control, and so forth. Outside North America, high-end home theaters remain a niche luxury market for the upper classes, since houses around the world don't exactly have the necessary space for a dedicated "cinema" room. Also, that residential market is served by the same manufacturers that design cinema speakers for commercial theaters and a restricted number of luxury speaker brands.

For some time, the promise of a worldwide "home-cinema" market originated with the DVD and Blu-ray formats, which carry multichannel sound formats from Dolby and DTS. From the mid-1990s

Large-sized custom home theaters remain an important market in North America, but are just a luxury niche market in other parts of the world. These examples are from Colorado-based Terracom Systems, where acoustics, lighting, and integration are more highly valued than speakers.

until recently (approximately 2010), "home-cinema" created a temporary boom that was quickly exploited by the major consumer electronic (CE) brands, including most TV manufacturers.

At the low-end, many consumers experimented with multichannel home solutions from Sony, Panasonic, or Samsung at the same time they were buying large HDTVs and Blu-ray players. There was a notion that the market would eventually enable high-quality speaker manufacturers to "step-in," especially with the promise of new multichannel audio formats such as Super-Audio CD (SACD). But that never materialized.

Many hi-fi brands tried to create generic middle-level multichannel home solutions but the complexity of the wiring and the total system cost was not an attractive proposition so sales volumes never reached the expected level. That market never "boomed," not even with stereoscopic 3DTV and all the hype about immersive home experiences. Consumers seem to be more interested in mobile devices and are ready to compromise sophisticated living room experiences for the convenience of wireless and integrated solutions. This is apparent by the success of "under display" soundbars.

With the proliferation of tablets and smartphones, smaller all-in-one audio options—particularly soundbars, connected music servers, digital media docks and, more recently wireless solutions—became the priorities for mainstream consumers. The market for high-priced audio separate components and powerful surround-sound audio systems for the living room has been dramatically decreasing in the last five years, especially since many audiophiles realized they would no longer be able to collect Blu-ray discs as they did with vinyl and CDs.

According to Research and Markets, in the US, the home audio and cinema markets are set to decline in volume at a 2% compound annual growth rate (CAGR), falling to 26.6 million units and to $3.4 billion by the end of 2017. In 2013, the home audio and cinema markets continued to decline in volume sales.

In some European markets, Philips Electronics even dethroned Sony as the most successful home audio and cinema products manufacturer by offering wireless home cinema solutions that appeal to today's digital lifestyle.

However, commercial movie theaters continue to expand worldwide. The movie industry is still growing in ticket sales, the number of new theaters, and revenue. High-quality cinema sound is and will always be a fundamental part of the movie theater experience and continues to evolve and be reinvented, largely benefited by a continuous stream

a)

b)

In the crowded consumer electronics (CE) market, home-cinema solutions like this Bose LifeStyle V35 integrated system shown as a complete setup (a) and as separate components (b) were "crushed" by low-price offerings and changes in consumer priorities after DVDs and Blu-ray discs were replaced by streaming services and subscriptions.

In 2011, Bose introduced the Lifestyle 135, it's first soundbar system, which combines a full home entertainment system and cinema speaker system with a new wireless Acoustimass module.

of innovations and the commitment of Hollywood's major studios. This segment is showing the most promising perspectives in the commercial theater environment.

Global Motion Picture and Movie Industry

The global motion picture and entertainment industry reached close to $90 billion in sales in 2013, according to a MarketLine report (referring not only to the theatrical sector but also to other sectors including home videos, video games, pay television, merchandising, and network television). Market growth is forecast to slow to a yearly rate of –0.3% through 2015, to below $86 billion. But with industry value added—a measure of the industry's contribution to the wider economy—is expected to increase at a 0.5% compound annual rate over the 10 years through 2018–2019. Box office sales represent the leading market segment, generating close to $32 billion, or more than 36% of the overall market in terms of value.

According to Research and Markets' Global Digital Cinema Screen Market 2012–2016 study, which was released in January 2014, one of the key factors contributing to this market growth is the cost reduction and benefits for film producers. With the help of digital cinema, distribution costs, duplication costs, labor costs, and storage costs can be reduced.

The global movie industry continues to be fueled by the popularity of films and constant technological developments. Box office growth expanded for a while with 3-D screens and 3-D films involving higher admission charges. This helped support the transition from 35-mm projection to digital projection, server-based systems, and higher quality sound.

Positive trends in consumer spending and a revival in movie attendance are expected to benefit the industry over the next five years. Movie theaters are investing in digital technologies and immersive experiences to boost demand and increase ticket prices, supporting profitability.

With more than 40,000 US theater screens and an estimated 180,000 screens worldwide, the movie theater industry has a high level of capital intensity. Since 2009, industry firms have invested heavily in upgrading screens and projection systems to digital, IMAX, 3-D, and 4K projection and in renovating seating. These expenditures have raised the industry's capital intensity during the past five years. Although 3-D investment slowed in 2012 in response to the technology's plateaued popularity, digital upgrades continue apace because they offer better visual and audio quality and enable theaters to more quickly and inexpensively access distributed film content. Shortly after the digital migration, many theaters equipped with earlier models of 2K servers and projectors were forced to reinvest in 4K technology, which is now the current standard in production, and increasingly for consumer viewing.

Reflecting the economic downturn, the movies and entertainment market suffered in recent years but it was still able to maintain a modest growth. Historically, this market was thought to have been recession-proof. Average US ticket prices have been rising since 1992, often at a greater rate than inflation. Production and marketing costs have increased because of the Hollywood studios' shift to blockbuster films.

The growth was primarily a result of strong increases in China, South America, and Eastern Europe, which helped offset the decline experienced in the large US and Western Europe markets. The US is the largest market, representing a third of the global industry's revenue. Currently, the Chinese movies and entertainment market is one of the fastest growing in the world. China has become the world's second-biggest movie market and in 2013 grew to $3.57 billion, an increase of 27% over 2012's $2.8 billion. This is primarily due to the significant growth in the box office category, which currently accounts for more 90% of total revenues. The number of cinemas in China has more than doubled since 2008.

Despite audience declines in many regions, the increase in ticket prices has slightly offset the effect. Digital screens are rapidly growing worldwide and are already the dominant projection format (digital projection was available in only half the theaters worldwide in 2012). And with most theater digital installations, a new sound system was also installed. The reason is simple. Even though moviegoers rarely choose to see a movie because of the sound, with digital projection, the availability of multichannel sound formats became the norm. Many theaters were not even Dolby-equipped during the 35-mm film days. But now with digital projection, theaters offer multiple audio formats and audiences everywhere notice the sound quality improvement.

After serving more than 120 years as the dominant projection format in movie theaters, the reign of 35-mm film is expected to end by 2015, according to the IHS Screen Digest Cinema Intelligence Service.

According to David Hancock, IHS's head of film and cinema research, "movie theaters now are undergoing a rapid transition to digital technology, spurred initially by the rising popularity of 3-D films. This is resulting in the rapid decline of 35 mm, first losing its status as the dominant cinema technology in early 2012—and then causing it to dwindle to insignificance in four years."

The trigger for this transition to digital technology was the popularity of a single movie in December 2009: *Avatar*. At the end of 2013, 90% of the US screens had been converted to digital, while countries including Denmark, Luxembourg, Norway, the Netherlands, Canada, and South Korea have achieved full conversion. In Belgium, Finland, France, Indonesia, Switzerland, Taiwan, and the UK 90% have been converted to digital screens.

New generation movie theaters, such as Regal Cinema's RPX Premium Experience, combine large IMAX screens with 4K projection and advanced sound installations. This enables them to offer immersive experiences like Dolby Atmos and future standards.

The Philips LightVibes immersive cinema entertainment lighting solution uses LED light fixtures, moving light spots, back lighting, and large LED media panels to create an immersive light show that enhances event screenings.

New Horizons

At the same time, the global online movie market is expected to reach almost $14 billion by 2017, according to research from Global Industry Analysts. Moving forward, market growth will be fueled by broadband Internet and increasing rentals. The number of online movies is steadily growing, along with the online movie consumer base. With Netflix and Apple (among many other emerging new distribution platforms) facilitating movie downloads and viewing, consumers can choose movies to rent or buy for home viewing. But the home experience will remain inferior to what was even possible with Blu-ray for some time, and there is little incentive for consumers to invest in superior audio solutions that would enhance the experience.

For that reason, global movie ticket sales are expected to increase every year, with online movie booking making it easier for theaters to detail available seats and continuously improve the experience.

While movie theaters are evolving to 4K projection with better quality picture and sound, consumers are also being offered 4K television displays or Ultra High Definition (UHD). Although 4K content will be soon available in existing streaming services, there is no physical media format schedule to launch 4K-quality content for the home. This means that even if a consumer buys a large 4K UHD display, the full benefits are not currently available.

Some forecasters even point out that with consumers currently preferring to view content on "personal" devices (e.g., tablets), there will be a very slow replacement cycle for large television displays in the living room, meaning HDTVs will remain the standard. This provides an opportunity for movie theaters, which can offer 4K projection—4K resolution, including 3-D 4K projection, is better on a large-format screen—in conjunction with a quality audio experience.

Also, a new generation of digital laser projectors is accelerating the availability of 4K in theaters. These changes will bring better color rendition and increased brightness (twice the brightness), in addition to a better overall visual experience resulting from higher frame-rates—from 48 frames per second (fps), as seen in *The Hobbit*, to as much as 120 fps—helping to translate fast action in a much more realistic way.

New generation laser digital cinema projectors from companies including Barco, Christie, and NEC are available—the first theaters in the US have already installed them. While new more affordable laser projection engines will enable more digital theaters to open in emerging markets, expanding the industry. Sony is also expected to make a strategic move in the laser projector market, bringing 4K projections to commercial theaters and to high-end residential installations. Laser projection could revitalize the consumer's appetite for 3-D movies, since the images will be much sharper and brighter, compared to today's average experience.

This transition from 35-mm projection to digital, from digital 2K to 4K and laser projection is predicted to bring important investments to the market, since it is not only a question of renovating existing theaters and multiplexes. By the end of 2013, more than 100,000 screens had been converted to digital.

It is estimated that more than 5,000 lower-standard e-cinema screens in India and thousands more in other poor regions of the world will be

Resources

Custom Electronic Design and Installation Association (CEDIA), www.cedia.net.

Dolby Laboratories, http://dolby.com/atmosmovies and http://dolby.com/atmos.

IHS Technology, http://technology.ihs.com/AboutUs.

Laser Illuminated Projector Association, http://lipainfo.org.

Motion Picture Association of America, http://mpaa.org.

Motion Pictures and Movie Industry: Market Research Reports, Statistics and Analysis www.marketline.com.

Music and Film Industry Association of America, http://mafiaa.org.

National Association of Theater Owners, www.natoonline.org.

Research and Markets, www.researchandmarkets.com/research/4hzdd3/global_digital.

converted to digital. This will help increase this market from the estimated 180,000 existing commercial screens to more than 200,000 in the next five years.

Evolution and Event Cinema

The transition to digital projection brings with it the need to complement those theaters with upgraded sound installations with digital processing. The impact in the production industry has been tremendous, forcing a complete renovation of sound recording and mixing studios to adapt to the evolving standards and new formats.

Most studios are driving the move to include more technology in their filmmaking with most movies now digitally captured in 4K. They also are engaging in experimentation with high-frame rates.

Cinema operators are experimenting with immersive or 3-D sound and 4-D cinema with motion seating is still having success. With more ideas and innovations in the works, it will probably find traction in the coming years.

Also, as part of digital cinema technology's second-phase rollout, cinema operators have focused their attention on the electronic distribution of movies. Instead of putting content on hard drives, the industry is using file distribution by satellite and IP networks. That type of connection is extremely relevant, since it enables those theaters to increase their non-film programming with events such as opera, ballet, rock concerts, and sports events. This is increasingly being done on movie screens in Europe and around the world, driving the growth of a lively market for alternative content or event cinema (EC).

According to the IHS Screen Digest Cinema Intelligence Report, across Europe, opera was the most dominant genre on cinema screens in 2012, averaging 36.7% of all event cinema performances in a sample covering eight countries. The UK was the most advanced event cinema market in Europe, with 131 shows screened in UK cinemas in 2012, earning about $20 million in revenue. After the UK, the Netherlands was the most active European market with 129 events. EC is now available in Europe and more than 50 countries including the US, Japan, Israel, Brazil, and Mexico.

"The market for non-film content in cinema has developed as the digitization of cinemas has progressed," says David Hancock, senior principal analyst for Cinema at IHS. According to Screendigest, by 2018 global box office takings for alternative content is forecasted to exceed $1 billion.

To improve the entire movie theater event experience, Philips recently announced the launch of a new immersive cinema entertainment lighting solution called LightVibes. The concept, which was introduced at the CinemaCon trade show in Las Vegas, NV in March 2014, uses LED light fixtures, moving light spots, back lighting, and large LED media panels to create an immersive light show. The light show enhances and differentiates advertising and event screenings (e.g., pop/rock concerts). The light fixtures run automatically and scripted light shows are synchronized with the screen content.

"Just like the audience at a sell-out gig being immersed in the energy of the band's live performance, Philips LightVibes brings the drama and emotion of a fantastic light show to the cinema theater-setting, complementing the content on the silver screen," says Philips Lighting. According to the company, 81% of viewers at a screening said that LightVibes had significantly improved their experience of a concert event in cinema and they would be six times more likely to attend a concert in a cinema with LightVibes available. So, apparently, it's not only about picture and sound.

Theater owners are already adapting their facilities for other sources of revenue, with the new event market helping fill cinemas in off-peak times and making their business models more efficient. There are even successful business cases for live interactive gaming events and scheduled screenings of pre-recorded concerts and shows.

Such activities require those theaters to be equipped with traditional multichannel sound and much higher powered systems that include the ability to correctly route different audio sources and formats across the room. Some of those theaters are being equipped with additional equipment (e.g., wireless microphones, monitors, and mixers) similar to typical event facilities.

KAB Electro Acoustics RSX-1 Remote Analog Switch

KAB Electro Acoustics specializes in analog playback equipment for disc records of all types—from vintage 78s to LPs. I reviewed KAB Electro Acoustics's EQS MK12 disc remastering phono preamplifier, which provides a dozen different playback equalization curves for vintage records (*audioXpress,* October 2002) and the SpeedStrobe (*audioXpress*, July 2009). The SpeedStrobe remains the best device I've seen for checking and adjusting turntable speed. In addition to the usual 33 1/3- and 45-rpm rings, this 10" strobe disc has a dozen more covering the most common speeds encountered with 78-rpm records. This article focuses on KAB Electro Acoustics's RSX-1 remote analog switch.

By
Gary Galo
(United States)

Running out of inputs on a preamplifier or integrated amplifier is a problem for audiophiles. Collectors who play all types of disc records often have two turntables—one for LPs and 45s and another for 78s—but only one phono preamplifier. Others may have two different turntable/arm/cartridge setups just for LPs.

A few high-end turntables enable the mounting of more than one tone arm. However, switching audio signals at phono cartridge-levels can be fraught with problems if it's not correctly done.

High-quality switches and connectors are essential, but the length of the output leads between the switchbox and the phono preamplifier input is a potential source of trouble. A switchbox needs to be within easy reach, which often means a half-meter or more of output cable. This increases the total capacitance of your tone arm cabling,and can lead to hum and noise problems.

The KAB Electro Acoustics's RSX-1 remote analog switch was designed to address these problems by providing a clean, noise-free signal path in a remote-controlled device that can be placed close to your phono inputs (see **Photo 1**).

The switch will work equally well with line-level signals. The small remote control is wired, supplying 3 V to the switching relay. It is connected to the switchbox with a stereo mini-phone interconnect cable. The supplied cable is 6' long, so the remote can be placed in a convenient spot. The remote has two buttons and an internal CR2430 coin-type, 3-V lithium battery.

Versatility

Functionally, the RSX-1 was designed to be more than just an input switcher. There are three pairs of RCA jacks labeled J1, J2, and J3, and three modes of operation, configured using three DIP switches on the PC board. The switches are accessible by popping three black plastic caps on the top of the case. The illustration painted on the case indicates each mode's switch positions. In Stereo mode the unit functions as a switch box, with J2 and J3 functioning as inputs or outputs and J1 as common. Remote Button 1 connects J1 and J2, and Button 2 connects J1 and J3. This enables you to route either of two inputs (J2 or J3) to output J1. Or J1 can function as an input, with the output routed to either J2 or J3.

In Stereo/Mono mode J1 is the input, J2 is the output, and J3 is unused. J2 is a stereo output when Button 1 is pressed on the remote control. Button 2 combines the two channels, providing a mono output at J2. In Stereo & Mono mode, J1 is common, J2 is connected to J1 in stereo, and J3 is connected to J1 in mono. Remote Button 1 selects the stereo connection and Button 2 selects the mono. With this configuration, a source connected to J1 can be routed to output J2 in stereo or J3 in mono. Conversely, stereo and mono sources can be

RSX-1 Remote Analog Switch
KAB Electro Acoustics
P.O.Box 2922
Plainfield, NJ 07062
(908) 754-1479
www.kabusa.com
info@kabusa.com
Price: $295

connected to J2 and J3 and routed to the J1 output.

KAB Electro Acoustics said the RSX-1 can be used to switch two turntables between one preamplifier, switch one turntable between two preamplifiers, compare two turntables or line sources, configure as a mono/stereo switch, configure to deliver separate mono and stereo outputs, or get good mono sound from your stereo phono cartridge.

Quality Parts

The RSX-1 is a completely passive device. KAB Electro Acoustics has used premium parts to ensure the highest possible signal integrity (see **Photo 2**). The six RCA jacks are Cardas-type GRFA PRT, which are right-angle, PC-mount connectors made with nonmagnetic eutectic brass, plated with rhodium over silver. The jacks cost $25 per pair and are the most expensive components in the device. The switching relay is a Panasonic ASX22003, which is guaranteed for low-level switching capability (a datasheet is available from Digi-Key or Mouser). The stationary contact is silver plus palladium, gold clad (AgPd+Au clad) and the movable contact is AgPd. The switch has a 100-mΩ maximum static contact resistance.

The ASX22003 is a double pole, double throw (DPDT), two-coil latching relay. Voltage is momentarily applied to one coil to move the contacts to one position. Once in position, the contacts remain latched without any applied voltage until they are moved to the opposite position by momentarily applying voltage to the other coil. With only intermittent current drain, the battery should last years before needing replacement.

The unit is housed in an all-steel enclosure, which provides complete shielding. To avoid ground loops, the left and right channel grounds are isolated from each other and from the chassis. The chassis ground post is exactly the type I use on equipment that I build—a 6-32 machine screw and a knurled thumb nut, ensuring a solid connection and long-term durability. The RSX-1's ground post should be connected to the ground on your preamplifier. Connect your tone arm ground wire directly to the preamplifier. Do not connect the tone arm ground wire to the RSX-1, since daisy-chained ground schemes will usually cause hum problems, especially with the low signal levels produced by magnetic phono cartridges. Remember, the RSX-1's chassis is not connected to signal ground.

The ground connection's purpose is to enable the chassis to function as a shield. In many line-level applications, the ground connection will be unnecessary.

The PC board layout has been designed to add minimum capacitance across the signal path. I used

Photo 1: The KAB Electro Acoustics RSX-1 analog remote switch provides clean, noise-free switching for phono and line-level signals. It can also function as a stereo/mono switch.

a digital LCR meter to measure the capacitance, carefully zeroing the meter to compensate for the short test leads before making the measurements. In Stereo mode, each channel measured a very low 13 pF. In the Stereo/Mono mode, capacitance was 12 pF per channel in stereo, rising to 20 pF when the unit is switched to mono. In the Stereo & Mono mode, capacitance is 13 pF in stereo and 17 pF in mono. Even a short cable between the RSX-1 and your phono input will add some capacitance, so you should consider the tone arm cable's capacitance, the RSX-1, and the output cable when determining how much additional loading capacitance, if any, is needed in your phono preamplifier. In most line-level applications the capacitance will not be an issue.

KAB Electro Acoustics proprietor and designer Kevin Barrett said that he was mainly concerned with the RSX-1's performance with high-output

Manufacturer Specifications

Bidirectional switch
Wired remote control with CR2430 lithium coin cell battery (included)
Cardas GRFA PRT RCA jacks
Panasonic ASX22003 latching relay
Fully isolated left and right channel grounds
Isolated chassis ground
Steel chassis
Dimensions: Switch box 7.5″ × 4″ × 1.4″; Remote 2.5″ × 1.75″ × 0.75″
Weight: 1.75 lb
Warranty: 1 year

Fresh from the Bench

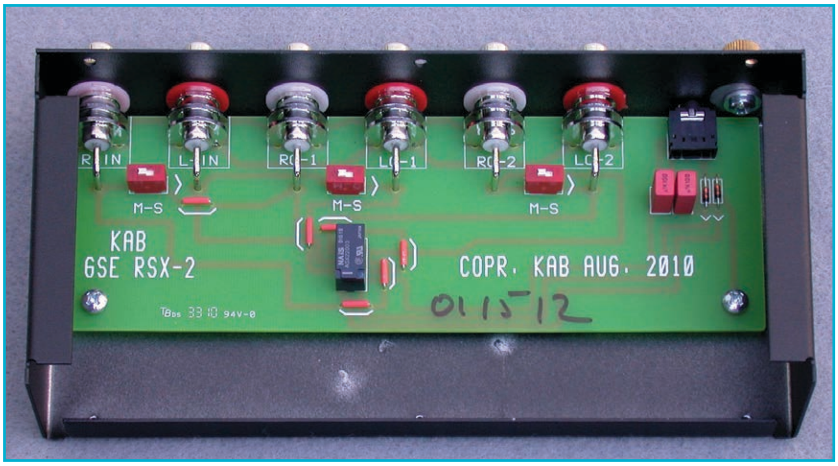


Photo 2: An internal view of KAB Electro Acoustics's RSX-1 reveals Cardas RCA jacks and a low-capacitance PC layout to ensure signal integrity. The steel enclosure provides complete shielding and the switching relay minimizes drain on the remote control's battery.

moving-magnet phono cartridges, where high impedances create the greatest potential for noise problems. Although he did not specifically test the unit with low-output moving-coil cartridges, he noted that, despite the high gain required in the phono preamplifier, the RSX-1's noise performance should remain excellent due to the low impedances involved.

Testing the Switchbox

I tested the RSX-1 in the system I use in my studio at work to do digital transfers of LP records. The system consists of a Rek-O-Kut Rondine 3 belt-driven turntable fitted with a Rek-O-Kut S-240 Fluid-Damped Tone Arm (actually a Jelco SA-750E) and a Grado Prestige Gold1 cartridge, all purchased from Esoteric Sound. Esoteric Sound revived the Rek-O-Kut name a few years back for its own line of turntables, arms, and accessories—these new designs should not be confused with the old Rek-O-Kut clunkers from over a half century ago!

The Rondine 3 has a unique feature—a removable spindle, which enables precise centering of off-center pressings (if you look and listen carefully, a perfectly centered pressing is the exception rather than the rule). The phono preamplifier is a Parasound Z-Phono, which feeds a Benchmark ADC-1 with RCA-to-XLR interconnects wired to correct the Z-Phono's inverted absolute polarity. Analog interconnects are made with D.H. Labs Pro Studio cable fitted with Canare F-10 RCA and Neutrik NC3MXX-B XLR connectors. The D.H. Labs cable has a Teflon-foam dielectric. The Canare F-10 is insulated with polytetrafluoroethylene (PTFE) and both connectors have gold-plated contacts. From there, I can connect the Benchmark ADC to a Tascam CD-RW901SL CD recorder or a high-resolution Tascam DV-RA1000HD digital recorder.

This LP transfer system is extremely quiet. If I advance my preamplifier volume control to maximum (without music playing, of course!), there is a trace of very low-level hum picked up by the unshielded Grado cartridge. At anything close to a safe playback level, the system is silent. When I put the RSX-1 in the signal path between the tone arm cable and the Parasound phono preamplifier, there was no increase in noise. I played several LPs, using Stereo Switch mode and Stereo/Mono mode and found that the RSX-1 did an excellent job maintaining signal integrity. I would have no reservations about using the RSX-1 in the phono cartridge signal path when making digital transfers or simply listening.

Overall Impressions

The RSX-1 is a versatile analog switcher that will solve many audio switching problems without degrading the signal and introducing noise. Its unique, remote-controlled design enables it to be placed close to a preamplifier's phono input, so cable lengths can be kept at a minimum. It works equally well with line-level signals. Considering the high-quality parts and construction, the price is more than fair. The RSX-1 is another fine product from KAB Electro Acoustics.

About the Author

Gary Galo retired in June 2014 after 38 years as as an Audio Engineer at the Crane School of Music, SUNY, in Potsdam, NY. He now works as a volunteer, transferring vintage analog recordings in the Crane archive to digital format. He is the author of more than 280 articles and reviews on both musical and technical subjects. He is an active member of the Association for Recorded Sound Collections (ARSC) and has been a regular presenter at ARSC conferences. Several of his conference papers have been published in the *ARSC Journal*.

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Electrostatic Speakers (Part 1)

Technology and Construction

Cone speakers, dome tweeters, compression drivers, and ribbon planars are electrodynamic transducers. Almost all of them use a magnetic structure and some sort of tubular voice coil with the exception of the ribbon planar, which uses a flat printed circuit coil. The first part of this article series details the technology and construction of electrostatic speakers and discusses their early development.

By
Mike Klasco and Steve Tatarunis
(United States)

Electrostatic speakers work differently than electrodynamics and are based on condenser microphone technology. Their construction and components are more like those of a capacitor. Most but not all electrostatic speakers are dipoles—open in the front and back. Electrostatic speakers typically require a high-voltage power supply, although there are a few self-biased electret loudspeakers and electret headphones.

Currently, the electrodynamic speaker dominates the market, with many boasting high performance, relatively non-critical construction, and no need for a power supply. So, one might ask "why bother with electrostatics?" Full-range electrostatic speakers are large and expensive. However, electrostatic speakers are well regarded because of their combination of well-defined, transparent sound due to the diaphragm being driven linearly over its entire surface, the low moving mass resulting in "fast" response and settling time, and very low distortion. These benefits are audible and measurable and only equaled by a few competing technologies, such as planar ribbons.

Electrostatic Technology and Construction

Electrostatic speakers use a thin, flat, or curved diaphragm. An electrostatic speaker, as with any capacitor, consists of a couple of layers. The layers usually include a plastic sheet coated with a conductive material (e.g., graphite) sandwiched between two electrically conductive grids (i.e., stator plates or electrodes) with a small air gap between the diaphragm and grids. If polyesters are used, they are typically polyethylene terephthalate (PET) such as Mylar or the higher performance polyethylene naphthalate (PEN) polyester (e.g., Teonex). The diaphragm substrate is usually made from a polyester film (with a 2-to-20-µm thickness). For reference, 25 µm is 1/1000". Carbon or another conductive compound is applied (sometimes silk screened) onto the stator followed by a protective insulated coating.

This second coating has a high dielectric breakdown strength to prevent arcing and rapture of the diaphragm from the high bias voltage. If the surface resistance is too low, distortion is created as the membrane surface charge moves in response to the combination of the modulating electrostatic fields from the stators and the deflections of the membrane. However, the composite coating needs a high enough dielectric strength to make it more reliable and resistant to changes during its operating life. In the early days, before air conditioning was common, many electrostatic speakers did not perform well in high-humidity environments (e.g., New Orleans, LA, and Miami, FL).

A fixed DC bias is applied to the diaphragm, approximately 5,000 V. The bias reaches the diaphragm through the terminals and the mounting contacts to the diaphragm assembly. One approach for this circuit path is electrically conductive silver paint epoxy.

For low-distortion operation, the diaphragm must operate with a constant and evenly distributed charge on its surface. This consideration makes the design, process, and manufacturing tolerances of curved diaphragms (e.g., the Martin-Logan electrostatic speakers) more challenging. The force exerted on the diaphragm by a single grid (i.e., single-ended) is too nonlinear, causing harmonic distortion, which defeats the purpose of designing and constructing a state-of-the-art speaker. The use of grids on both sides cancels out the voltage-dependent part of nonlinearity but leaves a charge-dependent (attractive force) part. The result is near-complete absence of harmonic distortion.

The grids are driven by the audio signal. Front and rear grids are driven in opposite (push-pull) phase. As a result, a uniform electrostatic field proportional to the audio signal is produced between both grids. This causes a force to be exerted on the charged diaphragm, and its resulting movement drives the air on either side of it. The grids must be able to generate as uniform an electric field as possible, while still enabling sound to pass through. Suitable grid constructions may be perforated metal sheets, a frame with tensioned wire, wire rods, and so forth. The grid design, from openness, size of holes, and so forth contributes to the electrostatic speaker's performance optimization.

The conductive-coated film diaphragm is tensioned onto a frame to determined resonance. The tensioning is typically achieved with an exciting signal from a test jig. A few electrostatic tweeters and headphones have used "semi-tensioned" designs, but these are the exceptions (see **Figure 1**).

From Amplifier to Speaker Matching

To generate sufficient field strength, the audio signal on the grids must be a high voltage. The electrostatic construction is in effect a capacitor, and current is only needed to charge the capacitance

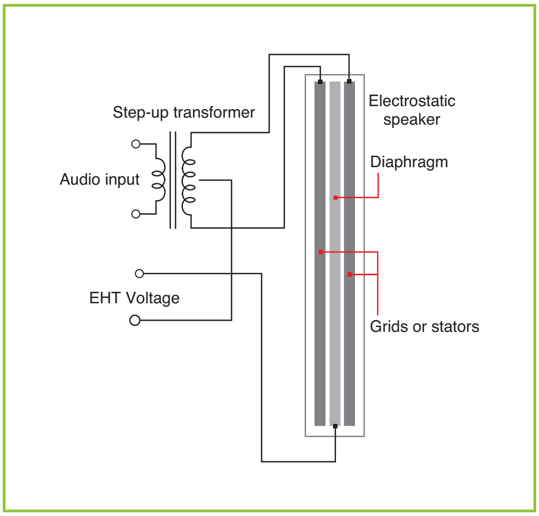


Figure 1: An electrostatic loudspeaker's sound is generated by the force exerted on a membrane suspended in an electrostatic field. This drawing shows an electrostatic speaker's construction and its connections (Image courtesy of *Wikipedia*)

created by the diaphragm and the stator

The electrostatic transducer is a high-impedance device. In contrast, a modern electrodynamic cone loudspeaker is a low-impedance device with higher current requirements. As a result, impedance matching is necessary to use a normal audio power amplifier. A transformer is often used for matching.

When the electrostatic speaker system does not include the amplifier, the transformer is integrated into the speaker. This transformer's construction is critical as it must provide a high turn ratio, which is challenging. It must be done so as to not to degrade the speaker's hi-fi potential. The transformer is almost always specifically designed to work with a particular electrostatic speaker.

Early Developments and Pioneering Electrostatics

In 1925, Edward W. Kellogg invented the moving coil loudspeaker with Chester W. Rice at General Electric's Advanced Technology Labs, which later became a part of RCA Labs. Kellogg was granted a patent on an electrostatic loudspeaker in 1934. However, from 1925 to 1935, there were a number of prior electrostatic-related patents and technical papers from other inventors. Despite some ingenious designs during this period, the electrostatic speaker made little practical progress. The major reason for this was the lack of available materials with suitable physical properties. Substances (e.g., gelatin, Japan varnish, gold size, silk, and rubber) are mentioned in these early patents. Lacking a suitable insulator, these early designs were forced to rely on the air gap and diaphragm to protect against electrical breakdown. Clearly, this limited the voltages that could be used and the resulting audio power.

The large size needed for electrostatics to

Photo 1: This JansZen electrostatic speaker ad dates back to the 1950s. (Photo courtesy of Retro Vintage Modern Hi-Fi)

produce any sort of low-end response and lack of appropriate diaphragm materials at the time made the electrodynamic cone speaker more attractive to most of the audio industry. In researching this article, we found that one of the first electrostatic speaker patents, which was issued to Colin Kyle, used a thin, flexible diaphragm, fly wire, and mosquito netting! The conductive coating consisted of a thin metal leaf. It was insulated from the stator by placing it under a thin gelatin layer. Around 1931, P. E. Edelman produced an electrostatic speaker using gold foil, Japan varnish, tensioned springs, and expandable corners.

World War II accelerated electrostatic speakers' further development and commercialized many new materials that had been invented but had not made it out of the lab. One example was polyvinylidene chloride (PVDC), which was discovered at Dow Chemical in 1933. It was initially developed as a spray used on US fighter planes and, later automobile upholstery to protect them from the elements. Dow Chemical later named the product Saran. In 1949, Dow introduced Saran Wrap, a thin, clingy plastic wrap primarily used for wrapping food. The wide commercial availability and the thin gauge made this low-cost film attractive to many early electrostatic speaker experimenters.

Photo 2: The complete KLH Nine system consists of two large, flat panels, closely resembling decorator-type room-divider screens both in shape and in appearance. A pair of triple panel KLH Nine speakers are shown. (Photo courtesy of Essence Electrostatic Loudspeakers)

Today's Saran Wrap is no longer composed of PVDC due to environmental concerns with halogenated materials. It is now made from polyethylene. An early alternative to Saran Wrap was PET film. DuPont first used the trademark Mylar in June of 1951 for its PET polyester. With practical and commercially available films on the market, the electrostatic speaker design progressed.

In 1947, naval engineer Arthur Janszen took part in a research project for the US Navy. The Navy tested microphone arrays for their work on the acoustic torpedo, which was perhaps the first "smart bomb." The test instrument needed an extremely accurate speaker, but Janszen found the cone speakers were too nonlinear in phase and amplitude response to meet his criteria.

Janszen believed that electrostat speakers were inherently more linear than cones, so he built a model using a thin plastic diaphragm treated with a conductive coating. By 1952, he had an electrostatic tweeter element ready for commercial production and was granted a US patent.

His company, JansZen, still makes an evolved version of the original design. Since the JansZen 1950s tweeter element was limited to high-frequency reproduction, it was often used in conjunction with woofers—most notably those from Acoustic Research. The AR-1W acoustic suspension woofer with the JansZen 130 electrostatic tweeter array was the "cool" audiophile system of the time (see **Photo 1**).

Looking back, the woofer did not fit the electrostatic speaker. The woofer was beaming at is usable top-end while the tweeter array was quite wide at crossover so the off axis and power response was not ideal. Nevertheless, the commercial success was enough to inspire Pickering (Stanton) to develop and market the Pickering add-on tweeter models 580 and 581, which is the company's version of add-on electrostatic tweeters in 1956.

In 1959, Janszen gave Neshaminy Electronic the rights to manufacture and use the tweeter. He also agreed to help develop products that used it. At this time, Janszen had also developed a full-range electrostatic and joined KLH Research and Development as vice president of engineering. While there, he introduced the KLH Nine (see **Photo 2**).

In the 1950s the British were busy with their own electrostatic speaker developments. Peter Walker and David Williamson designed the Quad electrostatic loudspeaker (ESL). The first in the series was the ESL-57, put into commercial production in 1957. After several name evolutions,

Photo 3: The Quad Electrostatic Loudspeaker (ESL) is the world's first production full-range electrostatic loudspeaker, launched in 1957. The speaker was prized for its clarity and precision. (Photo courtesy of QUAD Miusikwiededergabe GmbH)

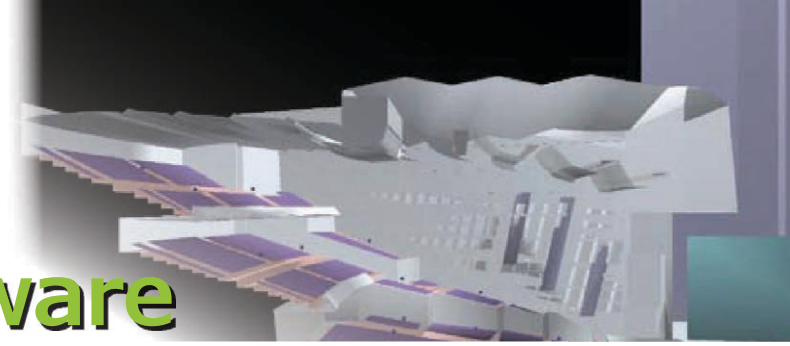
Photo 4: Stax's CSG-1 Tweeter was a high-frequency electrostatic speaker. It was first produced in 1954. (Photo courtesy of Stax Electrostatic Audio Products)

the company initially known as S. P. Fidelity Sound Systems came to be known as QUAD Electroacoustics, Ltd.

The Quad ESL had a "horizontal" design (see **Photo 3**). For anyone who has owned a pair, the clean sound compelled users to increase the volume but the speakers could not keep up. In the 1960s, the higher power tube amplifiers (Mike Klasco used Dynaco Stereo 70 tube amps bridged to drive his Quads) drove the Quad ESLs, but the less stable generation of transistors amplifiers did not handle the lossy capacitive load nor the higher impedance. To New Yorkers, the Quad's slightly angled expanded black metal grill resembled the woven straw cushions of the NYC subway seats.

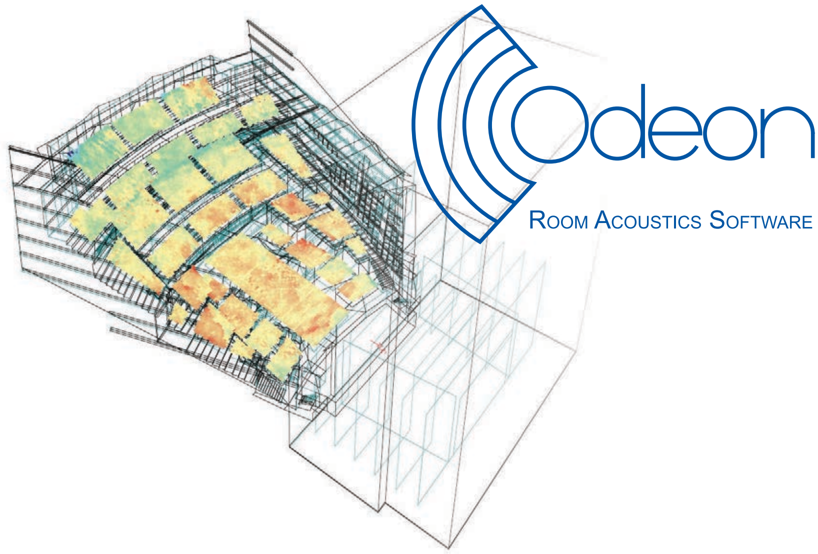
Let's not forget the early electrostatic speakers from the Far East. The Japanese used self-biased electret microphones in communications equipment during World War II with an innovative application of crude materials available in the 1940s. The famous electrostatic headphone company Stax, which was founded in 1938, focused on condenser microphones but launched its first electrostatic tweeter in 1954 (see **Photo 4**).

Stax continued to develop and introduce electrostatic headphones and full-range electrostatic speakers until 1992, afterward focusing only on electrostatic headphones. Recently Stax was acquired by the Chinese firm Edifier. We will discuss modern electrostatic developments next month.



ODEON Room Acoustics Software

This article details the software developed during the ODEON project, which began in 1984 as a collaboration between the Technical University of Denmark and a group of consulting companies. The purpose was to provide reliable prediction software for room acoustics. ODEON's first ray-tracing versions were targeted at solving acoustic problems in concert and opera halls.



By
Richard Honeycutt
(United States)

Author's Note: In this article, "Odeon" refers to the company and "ODEON" refers to the software.

In 1989, Odeon introduced hybrid models, combining cuboid-net-method ray tracing with image source methods (applied differently for early and late reflections) for more reliable results and faster calculation. In 1991, Odeon officially released ODEON V. 1.0, along with a comparison against data measured at the Royal Festival Hall in London.

Improved calculation methods and analysis capabilities, as well as auralization, were added to ODEON during the early 1990s. The current V. 12 incorporates an impulse response measurement system integrated in the software enabling easy comparison of calculated and measured responses. This addition to the software eases tuning models of real spaces to match their actual performance.

ODEON's Materials List

The user can enter model geometry into ODEON via the built-in extrusion modeler. The application allows drawing 2-D surfaces on a selected plane that can be extruded to provide a 3-D geometry directly recognized in ODEON. The application supports layers. When different extrusion planes are required for a complex model, the user can create a different file—one for each extrusion plane—and

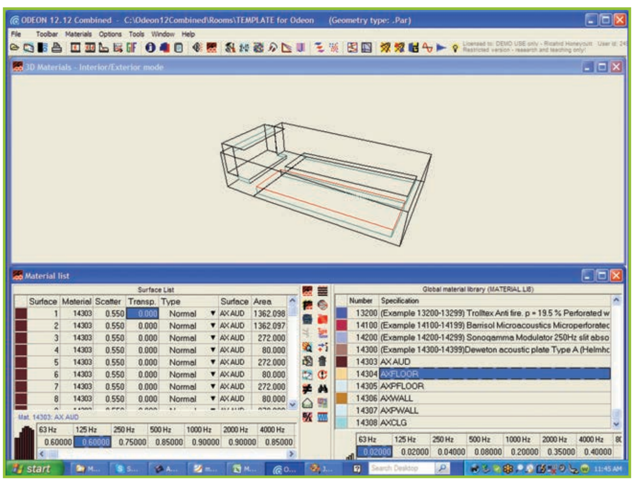


Photo 1: The material list is used to assign materials.

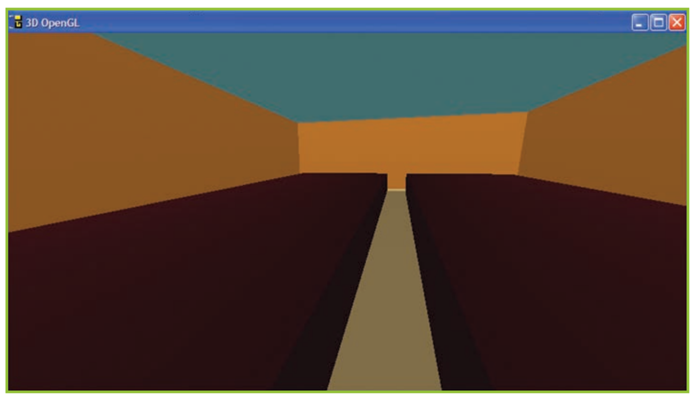


Photo 2: The 3-D Open GL view shows the modeled room by colors related directly to the assigned materials.

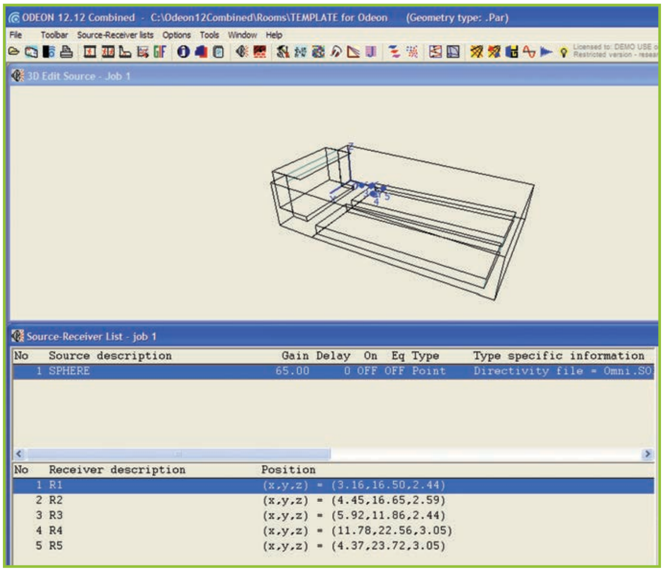


Photo 3: Sources and receivers are entered directly into the Source-Receiver List.

Parameters Plotted vs. Frequency				
EDT	T15	T20	T30	Curvature (C)
Ts	SPL	A-weighted SPL	D50	C7
C50	C80	U50	U80	LF80
LF-C80	Diffusivity	Echo-Dietsch		

Table 1: ODEON can plot several acoustical parameters vs. frequency.

Wideband Parameters				
Unweighted SPL	A-Weighted SPL	C-Weighted SPL	STI	STI Female
STI-male	RASTI	SPL expected	T30 average	LF80 average
Lj average	Bass ratio	SIL	AI	ALcons (from STI)
Reflection density	IACC	Early stage support	Late stage support	Total stage support

Table 2: ODEON also plots wideband parameters.

combine them into a single one.

ODEON can also import model geometry from .dxf, .3ds, and SketchUp files. (SketchUp imports require a plug-in that is freely available from www.odeon.dk). Alternatively, a model can be entirely built using the geometric modeling commands available in the ODEON editor.

After the model has been entered, it will appear on the screen in a wireframe view. Material for the surfaces in the model can be chosen via a Materials List screen (see **Photo 1**). ODEON includes a fairly comprehensive materials list, cataloged by type: simple absorbers, brick, ceramic tiles, wood, audiences, and so forth. The user can also create and use custom materials or material libraries.

The scattering coefficients for all materials are also available on the Material List, but only the midband scattering (average of 500 Hz and 1 kHz bands) is entered. ODEON calculates scattering in the other bands, weighting low-frequency values lower than high frequency values. Scattering is applied by the surface, not by the material.

With the materials entered, the user can view the open graphics library (OGL) rendering of the model (see **Photo 2**). Colors are derived according to the average reflectance of materials, following this RGB color model throughout the audible frequency range: red for low frequencies, green for mid frequencies, and blue for high frequencies, corresponding to the type of frequency range of light. Each color family is varied from black to white for 0% to 100% reflectance, respectively. High absorption at high frequencies leads to “warm” colors, while high absorption at low frequencies leads to “cool” colors.

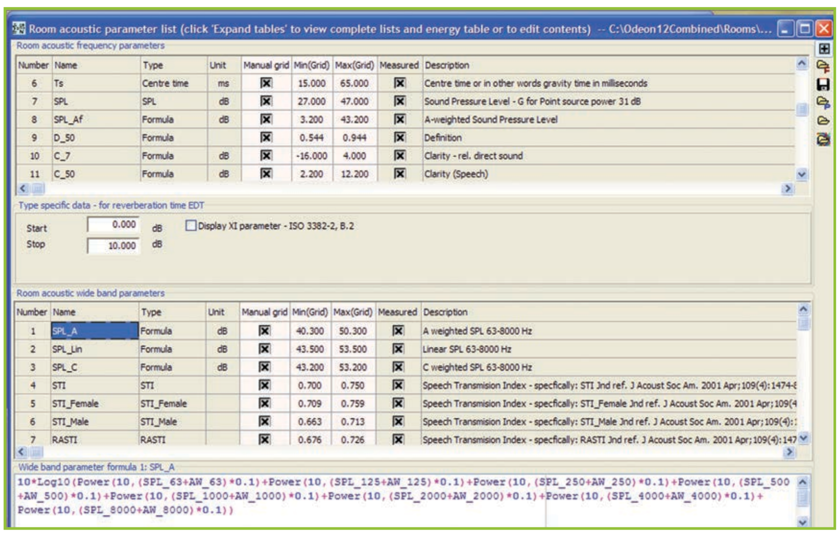
Analysis Preparation

The user can enter sources and receivers into the Source-Receiver List (see **Photo 3**). Use the X, Y, and Z coordinates to specify position. ODEON includes a spherical source that can be used for real-time (RT) calculations, plus a directory structure into which manufacturer-provided files can be downloaded. The default list includes files for normal speech, musical instruments, and known loudspeaker manufacturers. ODEON can accommodate .So8, CLF (CF1 and CF2), and XML files for beam-steered arrays. It does not use generic loudspeaker libraries (GLLs).

With the geometry entered, the materials assigned, and the sources and listener positions defined, the model is complete and ready for analysis. ODEON calculates an impressive array of acoustical parameters (see **Table 1** and **Table 2**).

For each analysis, the user can select the desired parameters via the Room Parameters List dialog (see **Photo 4**). Any parameter can be modified and new ones can be formulated. The Room Parameters List dialog shows the equation with which each selected parameter is calculated.

Photo 4: The user can choose the desired parameters for calculation via the Room Acoustic Parameter list.



Calculations

ODEON performs three types of calculations: single-point response, multi-point response, and grid response. The first two types of calculations provide results for specific listener positions. The grid response provides mapping data for a grid defined by the user. The user can control the way the calculations are performed via the Room Setup panel. Specific point response, multi-point responses, or a grid response can be run separately or as a batch, via the Job List (see **Photo 5**). Buttons on the Job List also display the calculation results.

ODEON treats scattering in different ways, depending on the distance from the source to the scattering reflector. Since by default ODEON automatically introduces scattering, according to the reflection-based scattering method, there will always be a reasonable degree of scattering in the simulation. Special scattering coefficient values are only required when the surface's geometry is simplified (e.g., when an audience area is modeled as a simple box, a 60% scattering should be specified to compensate for the lack of detail). Diffraction is automatically applied at the first and second reflection points, if applicable. Diffraction is treated by analytical formulae around a single edge (one-point diffraction) or two subsequent edges (two-point diffraction). Both cases occur only for the direct sound—not for reflections—on the assumption that in room acoustics, diffraction from reflections is much less important than diffraction from direct sound.

By default, ODEON uses the Image response method up to second-order reflections. Users can specify that image sources be used up to the 10th order. The order of image sources used in a calculation is called the transition order. Reflections up to this order comprise the early response. Subsequent calculations detail the late response. An algorithm called receiver-independent ray tracing (ray-radiosity) is applied, in which rays are transmitted all over the room. Secondary sources are placed at reflection points for all rays. A secondary source has

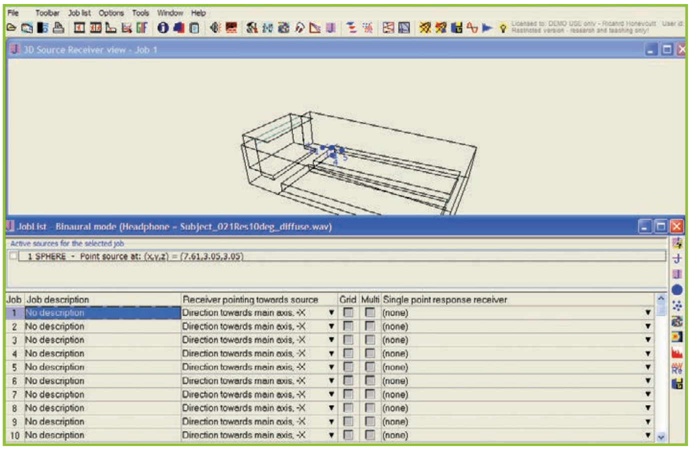


Photo 5: The Job List is the control point for selecting analyses to perform or results to display.

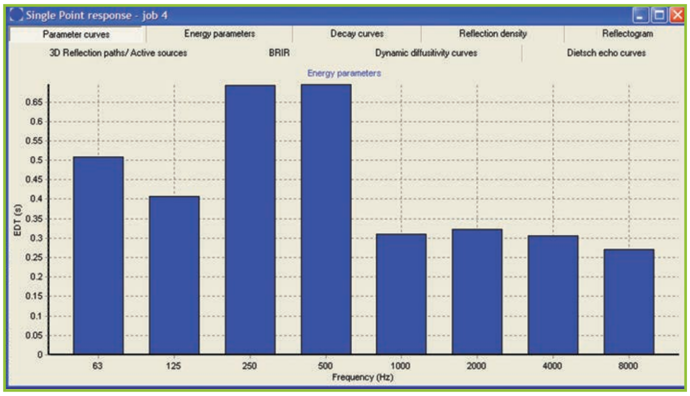
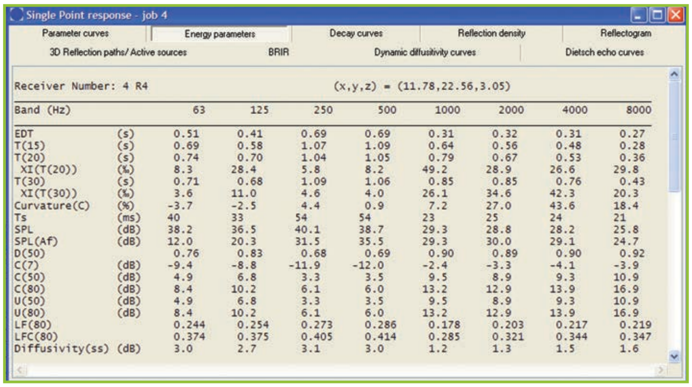


Photo 6: This single-point response shows EDT vs. frequency.



Band (Hz)		63	125	250	500	1000	2000	4000	8000
EDT	(s)	0.51	0.41	0.69	0.69	0.31	0.32	0.31	0.27
T(15)	(s)	0.69	0.58	1.07	1.09	0.64	0.56	0.48	0.28
T(20)	(s)	0.74	0.70	1.04	1.05	0.79	0.67	0.53	0.36
XI(T(20))	(‰)	8.3	28.4	5.8	8.2	49.2	28.9	26.6	29.8
T(30)	(s)	0.71	0.68	1.09	1.06	0.85	0.85	0.76	0.43
XI(T(30))	(‰)	3.6	11.0	4.6	4.0	26.1	34.6	42.3	20.3
Curvature(C)	(%)	-3.7	-2.5	4.4	0.9	7.2	27.0	43.6	18.4
Ts	(ms)	40	33	54	54	23	25	24	21
SPL	(dB)	38.2	36.5	40.1	38.7	29.3	28.8	28.2	25.8
SPL(Af)	(dB)	12.0	20.3	31.5	35.5	29.3	30.0	29.1	24.7
D(50)		0.76	0.83	0.68	0.69	0.90	0.89	0.90	0.92
C(7)	(dB)	-9.4	-8.8	-11.9	-12.0	-2.4	-3.3	-4.1	-3.9
C(50)	(dB)	4.9	6.8	3.3	3.5	9.5	8.9	9.3	10.9
C(80)	(dB)	8.4	10.2	6.1	6.0	13.2	12.9	13.9	16.9
U(50)	(dB)	4.9	6.8	3.3	3.5	9.5	8.9	9.3	10.9
U(80)	(dB)	8.4	10.2	6.1	6.0	13.2	12.9	13.9	16.9
LF(80)		0.244	0.254	0.273	0.286	0.178	0.203	0.217	0.219
LFC(80)		0.374	0.375	0.405	0.414	0.285	0.321	0.344	0.347
Diffusivity(ss)	(dB)	3.0	2.7	3.1	3.0	1.2	1.3	1.5	1.6

Photo 7: A tabular display of parameter values is also available.

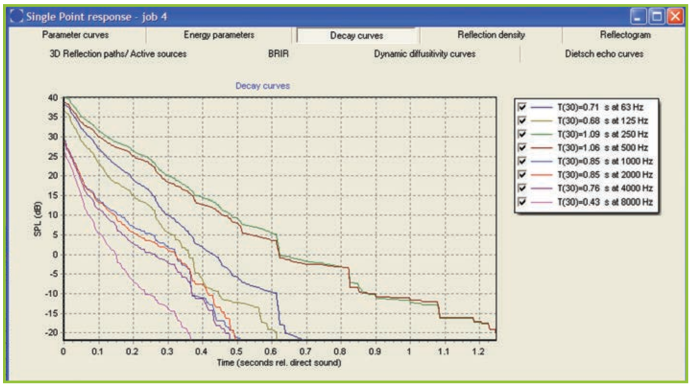


Photo 8: Sound pressure level (SPL) decay curves are plotted for each octave band.

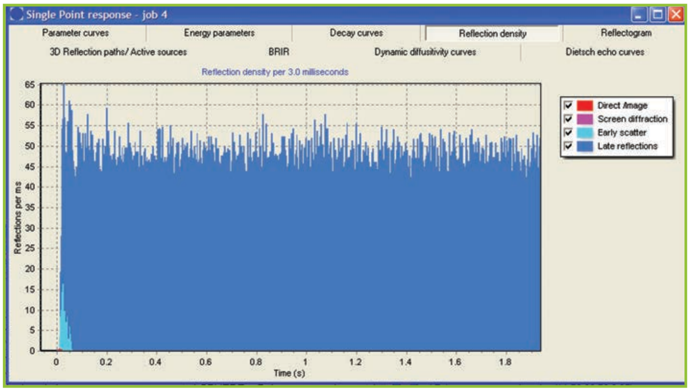


Photo 9: The reflection density plot shows the number of reflections per 3 ms plotted vs. times.

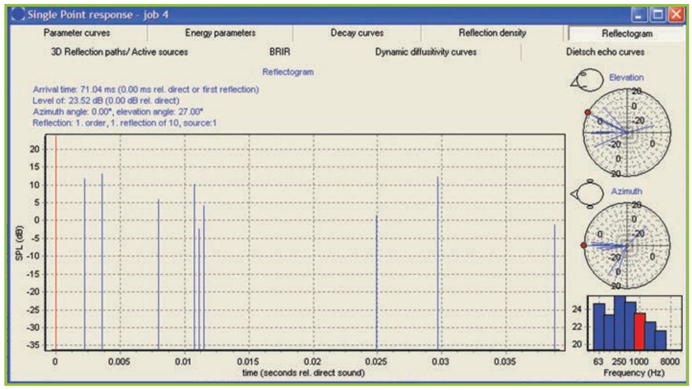


Photo 10: The reflectogram shows individual reflection arrival times.

an associated delay and strength. If a secondary source is visible from a receiver, the corresponding energy pack is added to the impulse response for that particular receiver. In this way, the impulse response for another receiver can be rapidly calculated from the same group of secondary sources, without the need to repeat the ray-transmission process. In addition to ray tracing, ODEON calculates RT by three statistical methods: Sabine, Norris-Eyring, and Puchades.

Responses

The results of the single-point response are presented in various ways, controlled by the tabs at the top of the panel shown in **Photo 6**. The Parameter Curves tab provides bar graphs showing the values of the various calculated parameters plotted vs. frequency. Use the right- and left- arrow keys to select the particular parameter being plotted.

Parameter values can also be displayed in tabular format (see **Photo 7**). The third tab of the single-point response panel shows sound pressure level (SPL) decay in each octave band (see **Photo 8**). The fourth tab presents a plot of reflection density (see **Photo 9**). Such a plot is useful in determining whether enough rays have been used for the calculations. At least 30 reflections per millisecond are required to ensure the room geometry's details have been correctly sampled for a reliable simulation. The reflectogram shown in **Photo 10** provides a detailed view of individual reflection arrival times. The reflection path from source to receiver is helpful in identifying surfaces that may need absorptive or diffusive treatment (see **Photo 11**).

Although the binaural impulse responses are primarily used to create auralizations, they can also be viewed via the BRIR tab (see **Photo 12**). The Dynamic Diffusivity Curves tab in the single point response panel can be used to examine the variation of sound diffusion in the room as the decay progresses. Echo disturbance, calculated by the Dietsch and Kraak method, is graphed vs. time, which is accessed via the Dietsch Echo Curves tab.

The multi-point response analysis provides stacked bar graphs of the energy parameters plotted vs. frequency, if the Energy Parameters (1) tab is selected on the multi-point responses panel (see **Photo 13**). The wideband values of the parameters are shown via the Energy Parameters (2) tab. Use the right- and left-arrow keys to select the specific

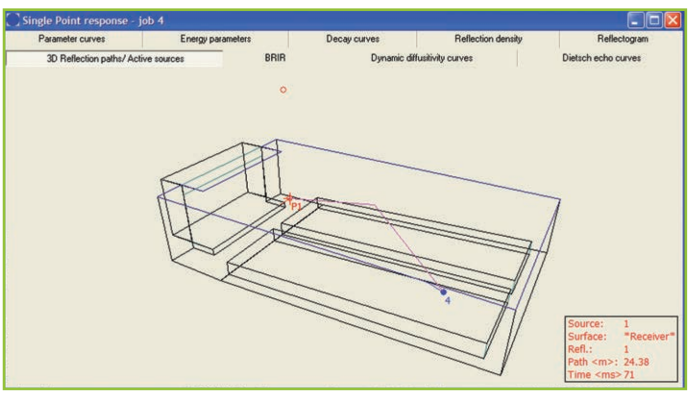


Photo 11: The path of each reflection can be displayed.

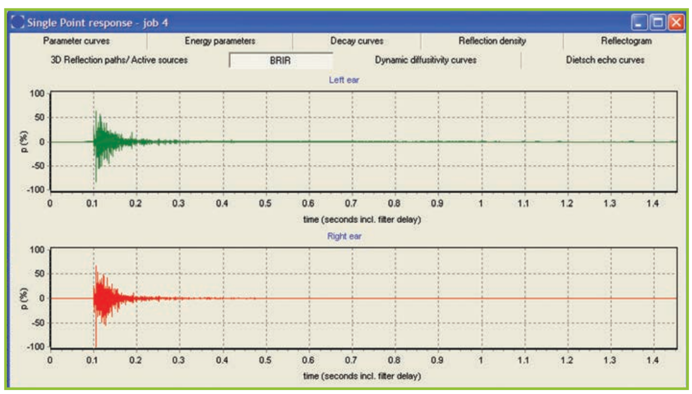


Photo 12: The binaural impulse responses for each ear can be viewed.

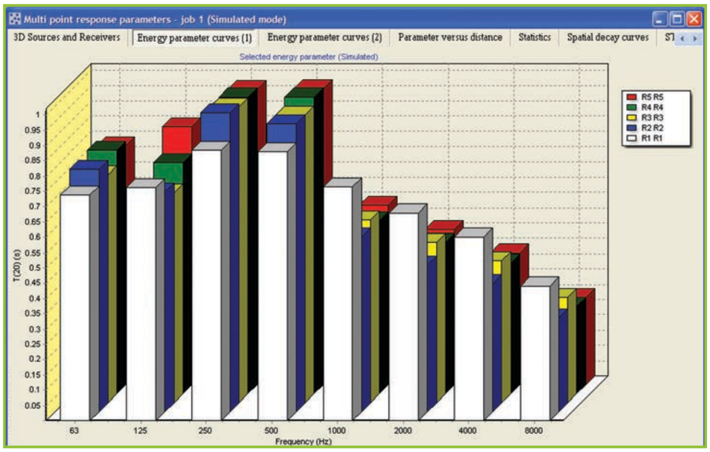


Photo 13: Values of each energy parameter at each receiver are plotted vs. octave band.

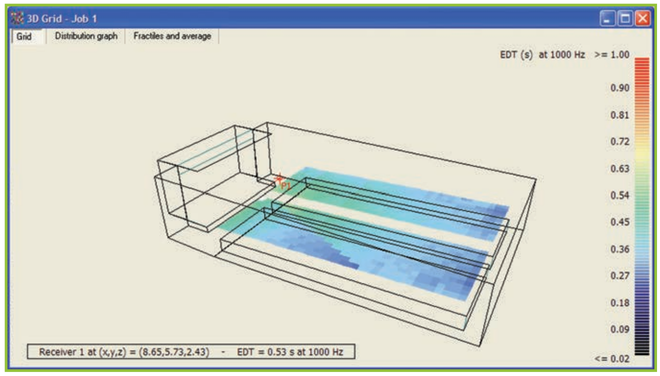


Photo 14: The grid response function provides parameter mapping.

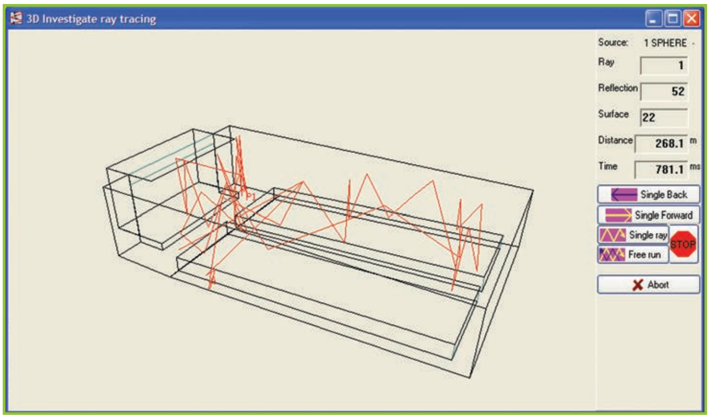


Photo 15: The 3-D Investigate Rays assists in visualizing ray paths.

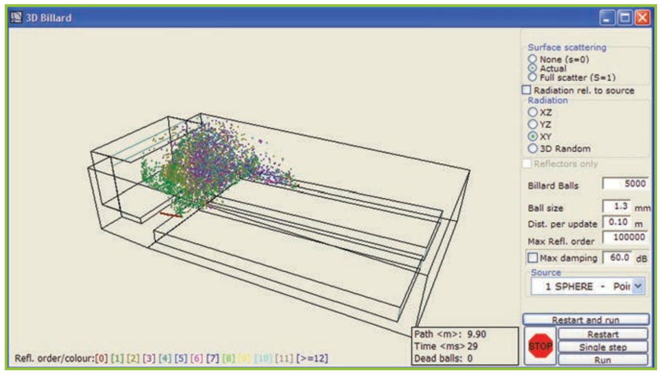


Photo 16: The 3-D Billiard animation shows how wavefronts progress through the room.

parameter to be graphed. The other three tabs on the multi-point display panel are self-explanatory.

The grid analysis provides maps of the selected parameters across a user-defined grid. As an example, **Photo 14** shows the early decay time (EDT) map for the sample we have analyzed. Use the left- and right-arrow keys to select other parameters. The up- and down-arrow keys can be used to change the displayed octave band.

In addition to the single-point , multi-point, and grid response panels, three buttons activate specific displays. The 3-D Direct response button on the Job List shows the mapped direct SPL. The 3-D Investigate Rays button on the main toolbar enables the user to see the progress of rays as they propagate through the venue.

Photo 15 shows how this helps visualize the paths of both helpful and detrimental reflections. The rays can be added one at a time or the display can be set to free-run. This display is useful for locating holes in the model. Although (unlike some programs) ODEON can run with small leaks, calculations are terminated if ray loss exceeding 20% is found.

The 3-D Billiard display button on the main toolbar brings up a display that represents wavefronts by small "billiard-ball" objects that progress in the room (see **Photo 16**). As for 3-D Investigate Rays, this display is particularly valuable for locating small leaks in the model (which can cause errors) and identifying sources of focused reflections.

ODEON provides online binaural auralization plus offline multichannel convolution with individual adjustment of the delay and level of each channel.

In addition to uses for architectural acoustics, ODEON provides capabilities for evaluation of industrial noise and noise isolation, through its ability to utilize transmission through semitransparent surfaces. Accordingly, ODEON is available in Basic, Industrial, Auditorium, and Combined editions. All versions support International Organization for Standardization (ISO) standards 3382-2 and -3, 14257, and 60268-16. The Auditorium and Combined editions also support ISO 3382-1.

In addition, the Auditorium and Combined editions provide calculations of a number of room acoustic parameters not available in the other versions. The industrial version includes the capability of using line sources (these are different from line arrays) and surface sources. A comprehensive comparison of the Basic, Industrial, Auditorium, and Combined editions, a demo version, and a free trial are available at www.odeon.dk.

Digre Positions His Father's Legacy for Future Success

An Interview with MISCO'S Dan Digre

MISCO President Dan Digre joined the company his father founded in 1984.

By
Shannon Becker
(United States)

SHANNON BECKER: Your father Cliff Digre founded MISCO. Tell us a little about him and why he started the company.

DAN DIGRE: My father founded what is known today as MISCO, a manufacturer of loudspeakers, audio systems, and related products serving dozens of markets and related industries. He was the kind of innovator who could make the most out of every situation, turning problems into business opportunities.

For example, in 1947, my mother bought a small radio with "horribly distorted" sound. When the problem turned out to be a bad speaker with a rubbing voice coil, my father—then a student at the National Radio School in Minneapolis—took it to a reconer in St. Paul to get it fixed. When the reconer lost the speaker and made excuses, my father went into business as Minneapolis Speaker Reconing.

Cliff Digre founded what is now MISCO, a company that manufacturers loudspeakers, audio systems, and related products.

My dad actively pursued the reconing markets, including the then-thriving drive-in theater speakers. His decision made Minneapolis Speaker Reconing the largest reconing service in the country.

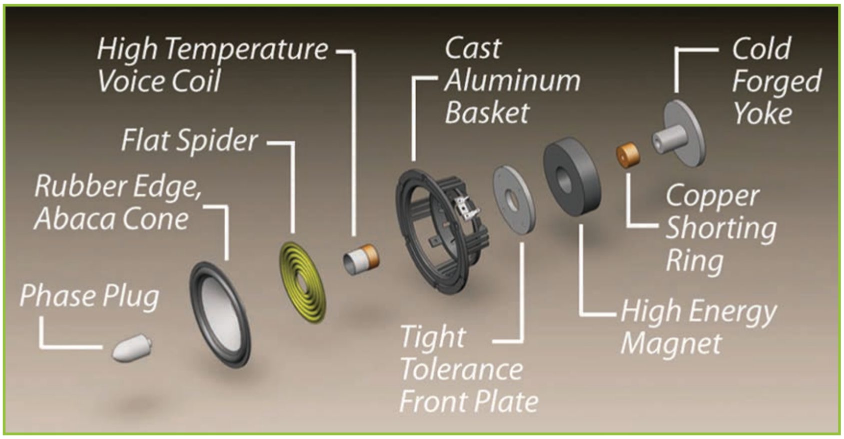
In 1956, a customer asked my father to design and manufacture an 8″ speaker with the same high quality as his reconed speakers. The initial production run of just 200 speakers was a huge success. However, just as the company began to assemble the second order for 1,000 speakers, the customer died and his company canceled the order.

My dad had 1,000 speakers so he sold the speakers to whoever would take them. Then, repeat orders began pouring in as buyers discovered the speakers' high quality. The line expanded to automotive speakers and 12″ speakers for hi-fi sets. Cliff's reconing company grew to become MISCO.

Although my dad was confident in his products, he wanted MISCO speakers to attract more attention in the marketplace and decided MISCO speakers also had to stand out visually. After considerable development, MISCO worked with Hawley products and began rolling out speakers with red molded cones, which came to be known as the legendary "Red Line."

My dad's career in the audio industry was book-ended by his service as a World War II combat veteran—first as a B-17 ball-turret gunner and radio operator based in England and 64 years later, as author of his memoir *Into Life's School*.

SHANNON: When did you take over the company and what products does it provide?



Every MISCO speaker combines precise engineering with high-quality components.

DAN: I took over the overall responsibilities at MISCO in 1990. Now I manage the company out of a new manufacturing facility in Minneapolis, MN. MISCO manufacturers more than 1,800 models of speakers and audio equipment for hundreds of customers in markets including pro sound, gaming, home theater, aerospace, medical, military, and transportation, as well as many other industries. My dad's speakers have touched the ears of millions of people worldwide.

SHANNON: When and how did you become involved in the family business?

DAN: I started cleaning drive-in theater speakers when I was in grade school. I used to take my speakers to school for "science day" and explain to my classmates how speakers worked. I even made a cut away that I could operate and show coil and cone movement. Through high school and college, I worked at every position on the assembly line so I was not only familiar with, but proficient, at every operation. Even so, my vocational choice was not to build speakers but to become a vocal music teacher and choir director and that's what I went to college to study. I spent four years as a high school choir director, which I really enjoyed. I still sing in choirs, but I decided to return to the family business and help my father.

The markets were changing, and I knew reconing and small-time manufacturing was not MISCO's future. We turned to OEM manufacturing and went from producing 1,000 speakers per week to 3,000 speakers per day. In addition to my responsibilities as president of MISCO, I have also been a past President of the Association for Loudspeaker Manufacturing & Acoustics (ALMA) International.

SHANNON: What sparked your initial affinity for audio technology?

DAN: I have always had a love for music and a fascination with sound. I suppose if you grow up with the technology around you it sparks your interest further. I have always been intrigued by the power of a loudspeaker—to produce stunning music, its simplicity of operation, and the complexity of interaction between the parts of the speaker and the acoustical environment. We do a lot less by trial and error today, but there is still a fair amount of "black art" in the design of cones and controlling the break-up modes out of the piston band.

SHANNON: Tell us about MISCO's first product. How did it come about and is it still sold today?

DAN: Aside from a variety of 4" and 5" drive-in speakers, MISCO developed an 8" full range speaker in 1956 for hi-fi. It had an Alnico magnet, a feathered-edge paper cone, and high sensitivity. It was featured in a kit in *Popular Science* magazine in the 1960s. Many of MISCO's customers have been buying the same speakers for more than 50 years and rely on MISCO to maintain exactly the same design for them because their product design depends on it.

SHANNON: MISCO has introduced major technological and design innovations in the loudspeaker industry. Can you discuss some?

DAN: As a custom designer and manufacturer of speakers for other companies, our approach has been to provide our customers with the best solutions available for their product. That's how we came up with our tag line "For the Sound You Want." Each one of our customers has a different view of what that sound is depending on their product and market—intelligibility, high power, smooth frequency response, durability in tough environments—whatever it is, it takes a different sense of the design and the materials needed to create that design.

In the early days of transitioning from alnico magnets to ceramic magnets, there were several challenges to be overcome. Not the least of it was how easily ceramic magnets would flake off and create "chips" in the magnetic air gap. My father was awarded two fundamental patents that addressed these problems and provided a method for attaching magnet assemblies to baskets that provided enormous flexibility in manufacturing.

MISCO also offers Class-D amplifiers with DSP.

MISCO'S new facility, built in 2001, was designed to maximize efficiency.

MISCO was one of the first companies to utilize high-energy neodymium magnets in non-automotive applications. Also, MISCO worked with Nuway Speaker products (now Loudspeaker Components) and Dupont to pioneer the use of Kapton as a high-temperature cone material for military applications. With customers in the alarm industries where durability and reliability are so critical, MISCO has also worked with voice coil and tinsel suppliers to develop long-life tinsel that can endure high power and high stresses for many years in the field.

SHANNON: How has the company changed to keep up with technology over the years?

DAN: Focusing on custom work has differentiated the company from its largely international competition base. Loudspeakers can be churned out by the millions by high-volume commodity producers, so we have to look at where we can apply our extensive knowledge and use of technology toward solving audio problems for other companies. The use of technology needs to support the overall goals of our company and our customers.

To keep our competitive edge, MISCO has turned to technology to support three key elements. First, we make no sacrifices on quality. Many competitors will use the "good enough" philosophy. At MISCO it's "what did the customer approve and does this speaker meet the agreed upon specifications or not." If not, it doesn't ship. To that end, we have utilized sophisticated automated end of line testing equipment to assure ourselves and our customers that what we ship meets specs. We understand that you can't inspect quality into a product, you have to build it in. That means working closely with our suppliers wherever they are in the world to assure raw materials are world-class quality and exactly meet our specifications.


The massive 3.4-kilojoule Magnetizer is used for large neodymium or ferrite magnet structures.

Second, to be cost competitive we have established an extensive chain of suppliers and manufacturing capacity in the US and Asia. Our suppliers are best in class and we utilize and share our technology with them to keep them that way.

Third, we have invested in our people in Minnesota to ensure they have latest technology tools, training and motivation to the best loudspeaker designers, assemblers, technicians, customer service people anywhere.

In the past few years, we have been supplying complete audio system solutions—speakers, enclosures, wiring, and high-end Class-D amplifiers with DSP. Being able to control the quality of the entire audio system as allowed us to give our customers high end, update to date technology that works as a system.

As far as pure loudspeaker technology goes, our soon to be released high-end audio product line will employ many new technological designs and benefits that speaker engineers and designers have requested. And the products will be made in the US.

SHANNON: Why did MISCO build a new facility in 2001?

DAN: Our focus through the 1980s and 1990s was on small industrial-type speakers for which MISCO became famous. As we saw the need for larger sizes (up to 21") and complete audio system solutions, this required more space for production as well as for the manufacturing equipment and more climate-controlled manufacturing. Our new factory is comfortable to work in and produces some incredible audio products.

The new facility has allowed us to layout our factory to maximize efficiencies in building speakers—the flow of materials, incorporating a Kanban system for material management, and flexible cell manufacturing to name a few.

One of the key features of building any pro-audio product is high-power/high-reliability testing. Pumping 1,000 W of audio signal through a professional loudspeaker during a power test will generate a dangerous amount of noise. That's why our power tests are conducted in a new power testing "bunker" that protects the outside workspace from loud noise and speakers that could catch fire.

Another tool used in the new MISCO facility is the massive 3.4-kilojoule magnetizer. This is used for large neodymium or ferrite magnet structures—any structure that can or will be used in a loudspeaker can be charged in this magnetizer.

We also installed new end of line test equipment. Every finished loudspeaker ends the production process with a comprehensive performance test. We use Listen's SoundCheck system to test each speaker against pass/fail limits (determined during the design phase), store the test data, and provide SPC data that can stored along with the speaker's serial number if needed. Each test can takes about 5 s. Data from the entire production run can be analyzed for production trends, average values, pass/fail yields, and so on.

SHANNON: What are the pros and cons of manufacturing in the US?

DAN: For US customers who are focused on high quality, require batch-to-batch, year-to-year consistency, and understand the difference between price and total cost of ownership, the benefits of US manufacturing far outweigh the challenges.

The benefits are three-fold: Communication—quick and direct communication means lower costs, less room for misunderstanding, and delivery of the expected product specifications; Quality—we don't "push" the specifications (which goes hand-in-glove with communications); and Cost—the savings from transportation, customs, potential change-outs, or worse, poor quality. The cost of poor quality in terms of sorting, lot inspection, reworking, field replacements, and damaged reputation is enormous. So many of our customers today have come back to US manufacturing because of all these issues.

For the American manufacturer to compete, they need to communicate the true benefits of "Made in America" and make certain that they deliver on those promises. MISCO has also applied this same commitment to the products we manufacture in China. This has allowed us to be close to customers who need product delivered in Asia. The difference is that any MISCO product manufactured in China is the same as the product that is manufactured in the US and uses the same parts suppliers.

SHANNON: MISCO recently celebrated its 65th anniversary. To what do you attribute the company's success?

DAN: It sounds cliché, but it's true: Success is achieved by meeting and exceeding your customer's expectations for quality, service, and value. It's about spotting market opportunities and changes and moving to fill those needs as quickly as possible. It's about developing trust throughout the entire supply chain from the raw material supplier all the way through to our customer's customer. Everyone does what they say they'll do and if there is a problem, they are focused on fixing the problem not ignoring it.

MISCO's Power Test Chamber with custom software assures reliability. The inset photo shows speakers during the testing process.

SHANNON: Are you currently working on or planning any new audio innovations?

DAN: As a custom manufacturer MISCO is always innovating. Customers come to us with new challenges and that requires new solutions. We work hand in hand with them to find the best solution typically involving smaller space, louder or deeper bass, and reducing the cost of ownership. Our customers rely on us to use the latest technology in amplification, magnetics, cone materials, voice coil materials, modeling software, and testing technology to help their product be innovative and competitive.

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

DAN: There will be a continuation of the need to achieve more performance from smaller products. The days of having a lot of space available for big speakers are over. Everyone is concerned with size and often weight whether in home, pro, commercial, or industrial markets. That requires more performance from smaller speakers, which means stronger motors and more sophisticated cone design.

There is much interest and a lot of research on the impact that nanotechnology could have on speakers. Can we "print" speakers? Can we "grow" speakers? Will we even need separate motors and cones or can they become the same part?

While the immutable laws of physics have to be respected, there will be new and alternative ways to get there. And wherever "there" is in the future, I can assure you, MISCO will also be "there."

Psychoacoustics (Part 2)

Noise Reduction and Sound Masking

In the context of this article, noise reduction applies to the playback of recorded sound. This is usually done in hardware (e.g., the Esoteric Sound Surface Noise Reducer or Packburn 325) or in computer software (e.g., DC8). The method is always some type of filtering and it is not psychoacoustic—the noise is removed or at least reduced. Also, these methods are general (i.e., no pre-coding is needed). They just operate on whatever input is presented and do the best they can with it.

By
Ron Tipton
(United States)

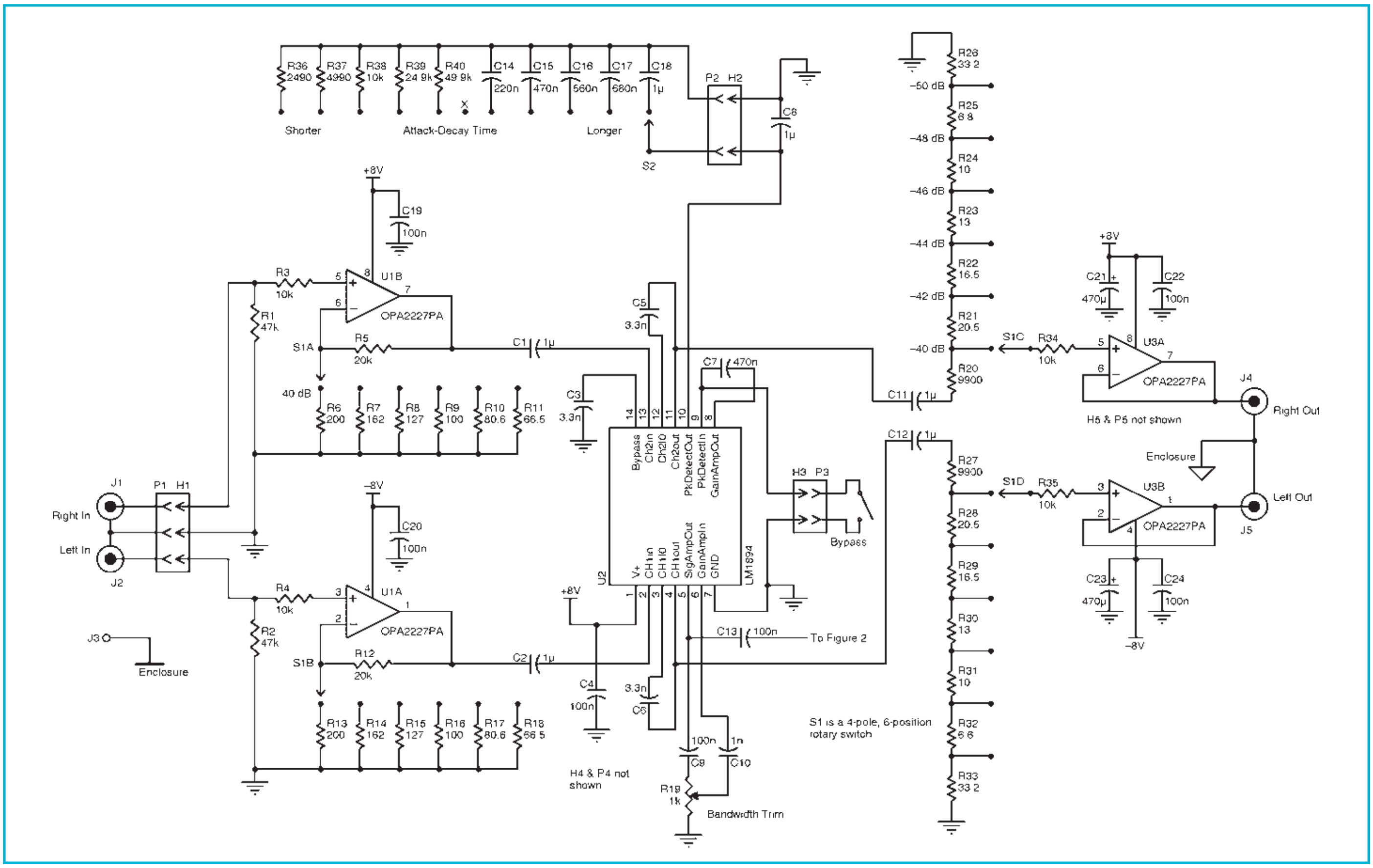
Texas Instruments LM1894 is partially psychoacoustic. Various compression-decompression methods (e.g., Dolby Digital, MPG3, and WMA) are psychoacoustic because they remove data thought to be the least audible while not seriously degrading the musical content. The various compression algorithms are not perfect, so it's a trade-off between fidelity and file size. (An in-depth discussion of audio compression is beyond the scope of this article.)

The LM1894

The LM1894, a 14-pin IC, is a stereo dynamic noise reduction (DNR) circuit for audio playback systems. It provides up to 10 dB of surface noise reduction, which is the broadband "hiss" sound commonly heard when playing magnetic tapes and 78-rpm records. It does not require an encoded input signal, such as Dolby, so it's compatible with most recorded material for surface noise reduction. The LM1894 uses psychoacoustic masking as well as an adaptive bandwidth technique so it is only partially psychoacoustic. This somewhat odd behavior makes it useful for listening but not as useful for restoration. That is, I can hear more noise reduction than I can record—more on this later.

The LM1894's datasheet and two application notes provide enough information to design a practical circuit. **Figure 1** and **Figure 2** show my design's circuit diagram. The LM1894 was designed for a 300-mV to about 1-V_P input signal but I wanted to drive it from a phonograph turntable with a low-output stereo cartridge. So the input stage, U1, is a low-noise stereo amplifier with a 40-to-50-dB switch-selectable gain. The metering circuit shown in **Figure 2** enables me to set the LM1894 input level to 300 to 500 mV_P. The output attenuator switches, S1C and S1D, are ganged with S1A and S1B to provide unity gain from input to output. This will make the LM194 compatible with a stereo phonograph preamplifier. Toggle switch S3 turns off the noise reduction for easy "on-off" comparisons. **Photo 1** shows my design's front panel.

In addition to the I/O level switch, there are only two controls: potentiometer R19 for bandwidth trim and S2 for selecting attack and decay times. Adjustment is rather simple. Adjust the bandwidth trim for minimum noise and then adjust S2 to hear if the noise can be further reduced. The two controls are only slightly interactive although I found shorter attack-decay times worked best with the source I was using, which were fairly noisy 78-rpm recordings.



Inside the LM1894

Around 1980 Richard Burwen designed and marketed the model 1201A dynamic noise filter. Its block diagram is similar to the LM1894 block diagram I am using for this article (see **Figure 3**). The LM1894 datasheet and application notes are dated 1984, so it's possible this IC was influenced by Burwen's design.

Figure 3 explains how it works. The function of this circuit could also be done with two operational transconductance amplifiers (OTAs) and a few regular op amps. But, the LM1894 is more compact and easier to use. Basically, this is all that happens. The left and right input signals are summed, amplified, and peak detected. The peak detector output voltage is converted to a current that controls the bandwidth of two low-pass filters: the OTAs and their external capacitors (C5 and C6 in **Figure 1**).

The low-pass bandwidth varies between 1,000 Hz and 30 kHz depending on the input music spectrum. This may not be readily apparent from the block diagram, but it works this way because of the response of the internal peak detector coupled with the external components: C7, C9, C10, R19, and the resistors and capacitors selected by S2.

Our hearing is most sensitive between 2,000 Hz and 10 kHz so noise in this bandwidth is very audible. Reducing the audio bandwidth lessens the audibility of noise and this DNR circuit low-pass filters this region. The control circuit's rapid response (attack

Figure 1: The circuit diagram for my dynamic noise reduction (DNR) prototype uses a Texas Instruments LM1894. The input amplification is balanced by the output attenuator for an overall unity gain.

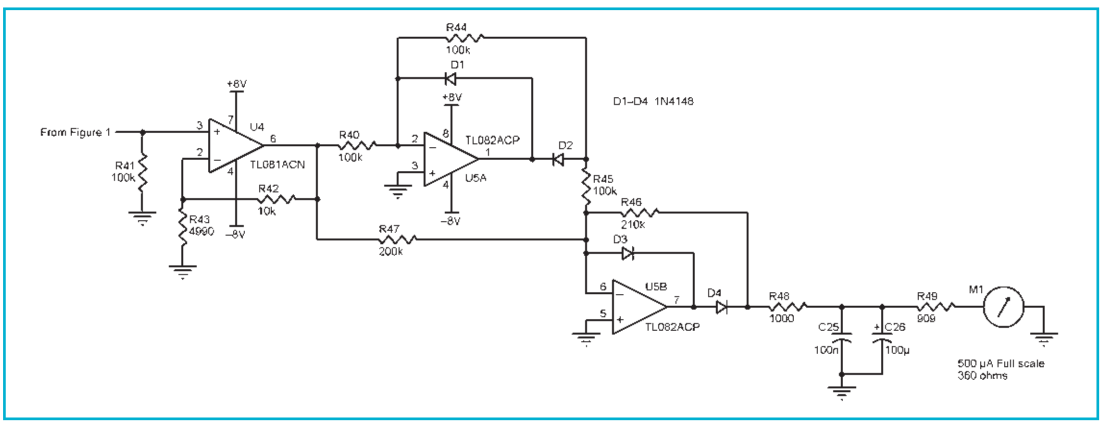


Figure 2: This is the circuit diagram for the dynamic noise reduction (DNR) input level indicator. The input gain is adjusted for a meter reading of 300 to 500 mV_P.

Photo 1: This is the front panel of prototype I built to test the Texas Instruments LM1894 dynamic noise reduction (DNR) system. The panel meter sets the input signal level.

and decay times) provides the noise reduction without introducing audible distortion in the music.

Figure 4 shows the frequency spectra of about a 2-s interval of music from an Art Tatum 78 rpm (see Resources). The red line is the "raw" input from the turntable and the blue line is the DNR output. Note that in the noise-reduced output the low frequencies are about 2 dB higher than the input while frequencies above 1,500 Hz are attenuated. When listening to this recording, I did not hear any reduction in musical content but there was a marked reduction in surface noise.

I connected the DNR output to the sound card's line input and recorded about 3 min of music, about 30 s with the DNR on followed by about 30 s with the bypass on. This sequence is repeated for the duration of the recording. There is a noticeable amount of noise reduction in this recorded file (tst4012a.wav) but not as much as I heard in my listening test. This is a good indication of some psychoacoustic noise masking. (I didn't do any processing on the recorded file except to remove the loud clicks produced by turning the bypass switch on and off so some clicks and pops are still audible because these are not "surface noise.")

Next, I recorded about 3 min of this 78-rpm record with the turntable output in the left channel and the DNR output in the right channel. This is file tst4012g.wav. I transferred it to my Tascam DR-05 digital recorder, connected its line-out left channel to the DNR left input and to the computer left channel input. The DNR's left channel output was connected to the computer right channel input. I played the wav file and recorded a "twice filtered" version as tst4012k.wav. There is less surface noise but I think the music quality was also slightly affected so two passes through the DNR is a practical maximum. **Figure 5** shows a spectrogram of a 2-s slice near the middle of the "k" file.

This is definitely a useful circuit for surface noise reduction, which seems more difficult to do in software without introducing undesirable "artifacts." The remaining clicks and pops are more easily removed (or attenuated) with computer programs (e.g., DC8, Sound Forge, and others). Adding a unity input gain and an output attenuation to the 4012 circuit would make it easier to accommodate higher level signals without an external input attenuator and output amplifier. (This circuit will operate from most any ±8 VDC regulated power supply.)

Sound Masking

I wrote an article titled "Masking Unwanted Sounds," (*Nuts and Volts*, September 2002). A lot has changed in the sound masking field since then. It has become a big business. Masking is used in large and small businesses to provide the perception of a quieter work place. But it adds sound on top of sound so I wonder about the long-term effects on employees. Nevertheless, it now seems to be a part of our lives.

In the *Nuts and Volts* article, I used sounds from nature (e.g., rain on the roof, ocean surf, a babbling brook, etc.). Now, white or pink noise is used with

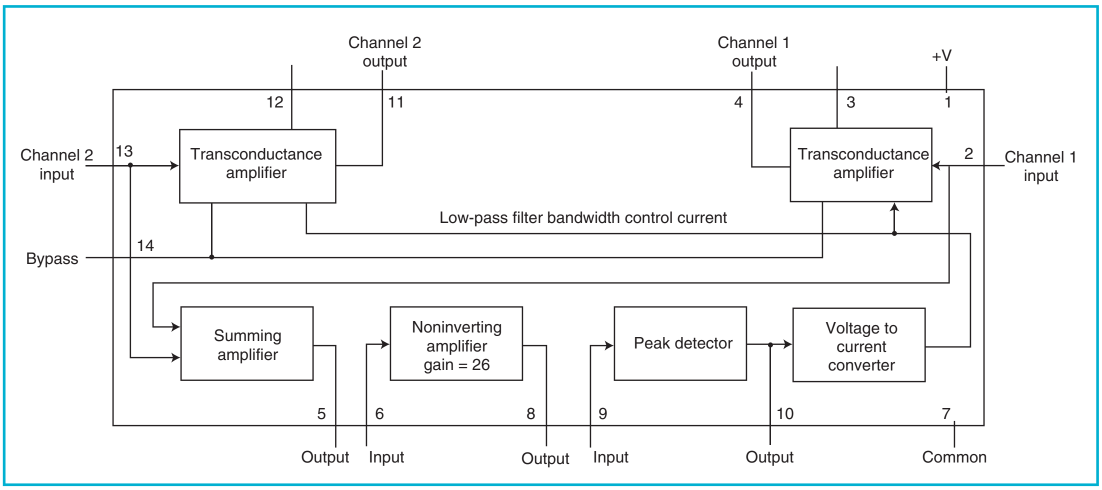


Figure 3: The Texas Instruments LM1894 block diagram shows the pin numbers of the 14-pin DIP package.

spectral shaping chosen to match the spectra of the sounds to be masked. Now, as then, the problem is to uniformly fill the space. Cambridge Sound Management calls its system Qt Technology.

In my *Nuts and Volts* article, I used an example of sleeping in a cottage on a Caribbean island about a 100′ from the Atlantic Ocean and listening to the waves. "We were immersed in that sound and it was almost like a physical presence," I wrote. I think presence is an apt description although I suspect I'm not the first person to think that. I conjectured this sense of presence could be achieved with a wall or ceiling of sound (i.e., covered with 8″ or 10″ speakers in suitable enclosures). The expense of this would be impractical for most purposes, so perhaps Cambridge and others have solved the presence problem in a more practical way.

Masking can be useful even without much presence under some conditions (e.g., getting to sleep). There are commercial units, usually marketed as sleep aids, which can cover a small area with a single speaker. They can play up to a dozen or so different nature sounds and some also include white or pink noise. I have used one on occasion with fair results. This tells me that I can experiment with some simple masking to find out what can be done. There are many freely downloadable Internet sound clips of office and restaurant background sounds to try to mask. And I have noise generators and filters to produce the masking.

However, a physical law does apply: The energy in the masking sound must be as large or larger than the energy in the sound to be masked. When the sound to be masked is, for example, from an office, it is generally broadband as is the shaped white noise masking. In practice, this means the mask's spectral density must be equal to or greater than the spectral density of the offending sounds. It has been experimentally determined that as the frequency to be masked decreases, the masking amplitude must increase. For example, at 400 Hz, the masking must be about 16 dB higher than the sound to be masked. Masking low audio frequencies is impractical so baffles and sound absorbers are required.

A lot of information about masking is available online. One of my more interesting finds was the installation guide for Dynasound's DS1042 soundmasking system. This is the only publication I found that suggested, in capital letters, that the masking must be introduced slowly so it won't call attention to itself. The guide also said it will take weeks to bring the masking sound up to its recommended level for effective privacy. Masking is accomplished by at least doubling the sound level

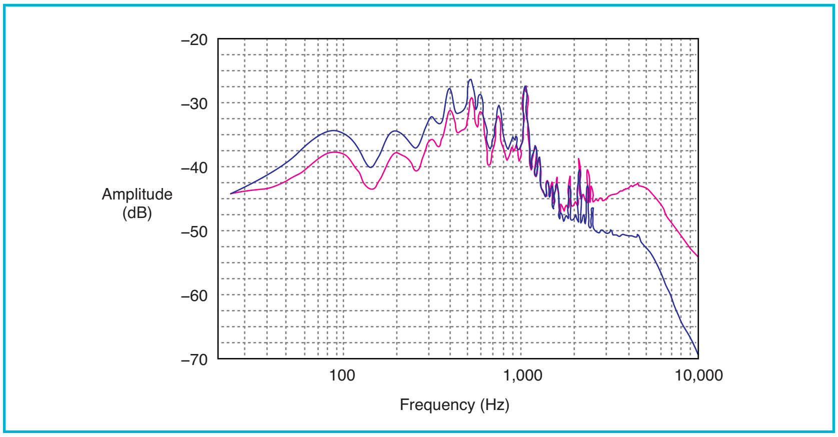


Figure 4: These are the frequency spectrum plots from filtering the turntable output (red line) with the model 4012 DNR (blue line). Note the filtered output is about 3 dB higher at low frequencies and much attenuated above approximately 1,000 Hz.

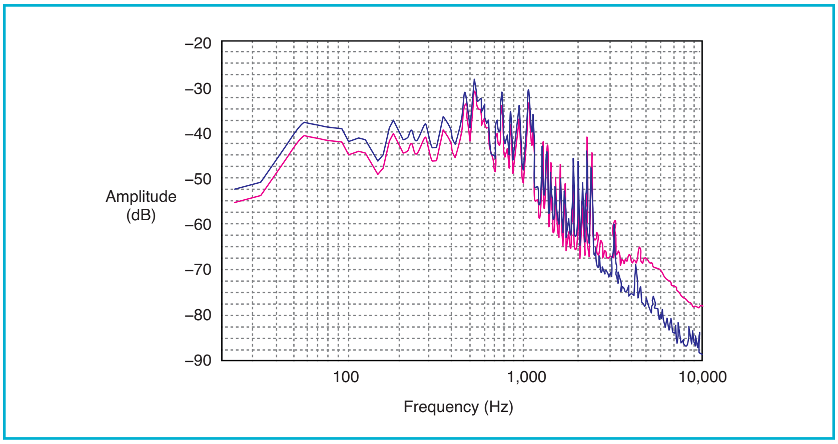


Figure 5: These are the frequency spectrum plots from twice filtering the turntable output. The first filtered output is the red line and the second filtered output is shown in blue. Additional surface noise was removed by the second filtering but at some loss of music quality.

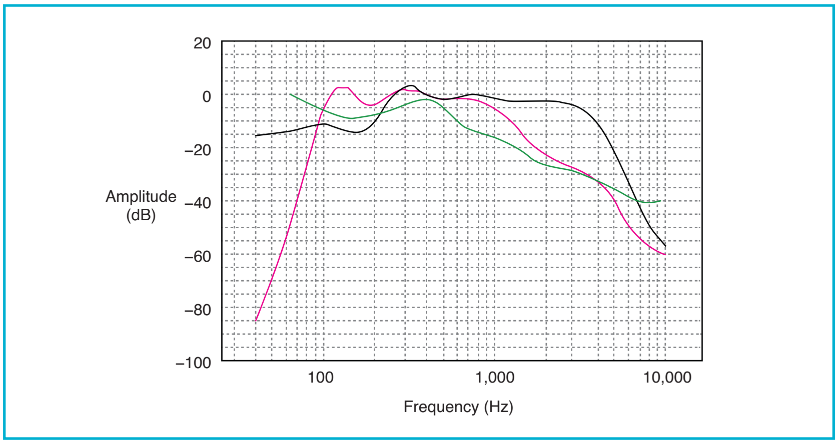


Figure 6: The red line is an average spectrum for human speech. The green line is the smoothed peak spectrum of the office-3a.wav file. The black line is the measured pass-band response of the band-pass filter used for the masking test. The filter's –3-dB cut-off frequencies are 157 and 3,536 Hz.

Resources

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DS1042 Sound-masking system
Dynasound, Inc. | www.soundmasking.com

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Freesound | www.freesound.org

Tenma72-1015 RMS Voltmeter
MCM Electronics | www.mcmelectronics.com

Sony Sound Forge audio editing software
Sony Creative Software | www.sonycreativesoftware.com

500E MK II Cartridge and D5127 stylus
Stanton | www.stantondj.com

DR-05 digital recorder
TEAC, Corp. | www.tascam.com

LM1894 Noise reduction circuit
Texas Instruments, Inc. | www.ti.com

DC8 Audio editing software
Tracer Technologies, Inc. | www.tracertek.com

so it's not unreasonable to expect it to be initially intrusive. And again I wonder if masking is the best way to achieve a quieter environment.

In his white paper, "What Makes a Sound Masking System Sound Good?" (Cambridge Sound Mangement, 2013), author Jack Heine says the masking should sound like moving air because almost everyone is familiar with air conditioning and that makes it easier to ignore. The next question to be answered is how to shape (filter) the noise so it sounds like moving air and has a spectrum that matches the sounds to be masked. An Internet search provided several human speech spectrums, which I averaged for the red line shown in **Figure 6**. The green line is the smoothed peak spectrum for

Photo 2: This equipment tested office sound masking with shaped white noise. The CD player for the office sounds is on the left side, bottom (a). A Model 527 sound level meter with the right side on a tripod recorded the mask-1.wav file. Its output is connected to a computer sound card in another room (b). The four center speakers were for the office sounds and the outside pairs reproduced the masking noise (c).

an office sounds file, office-3a.wav. It's apparent from this graph that office sounds are more than just speech, which is not unexpected because workers are moving around doing things. I also saw that the needed dynamic range was outside my graphic equalizer's 24-dB boost-cut range (a typical EQ range) so I used a TDL Model 601, which is a tunable band-pass filter. The black line shown in **Figure 6** is the filter's measured response that fits (except at the low-frequency end) the frequency range to be masked. (I tried turning the filter's low cut-off frequency lower, but the noise became more intrusive faster than the masking increased.) **Photos 2a–c** show the masking test equipment setup.

Photo 2b shows the "equipment stack" along with a TDL model 527 sound level meter I used as a microphone to record the mask-1.wav file. The four center speakers reproduce the office sounds while the two outside speaker pairs handle the masking noise. This is a sealed enclosure with eight identical Dayton 299-706 5.25" drivers with an overall response of –3dB at 100 Hz and 5 kHz.

To perform the test, I used the CD player for the office-3a file and shaped white noise from the noise generator and band-pass filter for masking. Both signals were amplified by the identical power amplifiers and then reproduced by the nearly identical speakers. I recorded the test in mask-1.wav starting with just the office sounds and then slowly increasing the noise until I had about 90% masking. Then, I increased the noise level until I had full masking but it seemed intrusive, so 90% is a more pleasing compromise. I think I could get used to it over time.

Figure 7 shows the frequency spectrums of unmasked and masked 10-s portions of mask-1.wav. To achieve about 90% masking, the masked signal needed to be about 10 dB higher than the unmasked sound.

When replaying the mask-1 file I found it to sounded like the "real time" test, so how is this psychoacoustic? It's psychoacoustic because we become used to masking noise just as we eventually become unaware of unpleasant odors. Our senses sort of shut down when there is not any new information present.

There is apparently some use of masking in restaurants and waiting rooms but it seems to me that most people would not be there long enough to acclimate so background music or television would probably be better—and these masking techniques are widely used.

The main branch of one of the banks I use here in Las Cruces, NM, installed a masking system about a month ago. It uses a waterfall sound and visually cues the customers with a 20' floor-to-ceiling simulated waterfall at the lobby entrance. The speaker are concealed in the ceiling and uniformly cover the entire banking area. It is intrusive to my ears so I asked one of the tellers about it. She replied that it bothered her at first but now she doesn't notice it at all. This supports the idea that masking is more useful to the work force than to the customers.

I'm also interested in quieting the environment by using sound absorption and active noise cancellation, when appropriate, but that's a subject for another article.

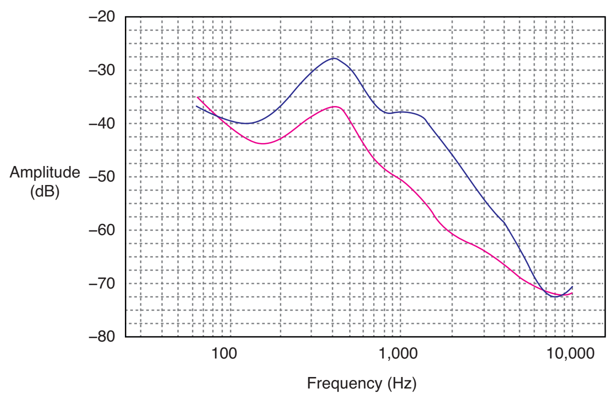


Figure 7: These are the frequency spectrum plots for a 10-s portion of mask-1.wav. The red line is unmasked and the blue line is masked as described in the text. The 10-dB difference achieves about 90% masking.

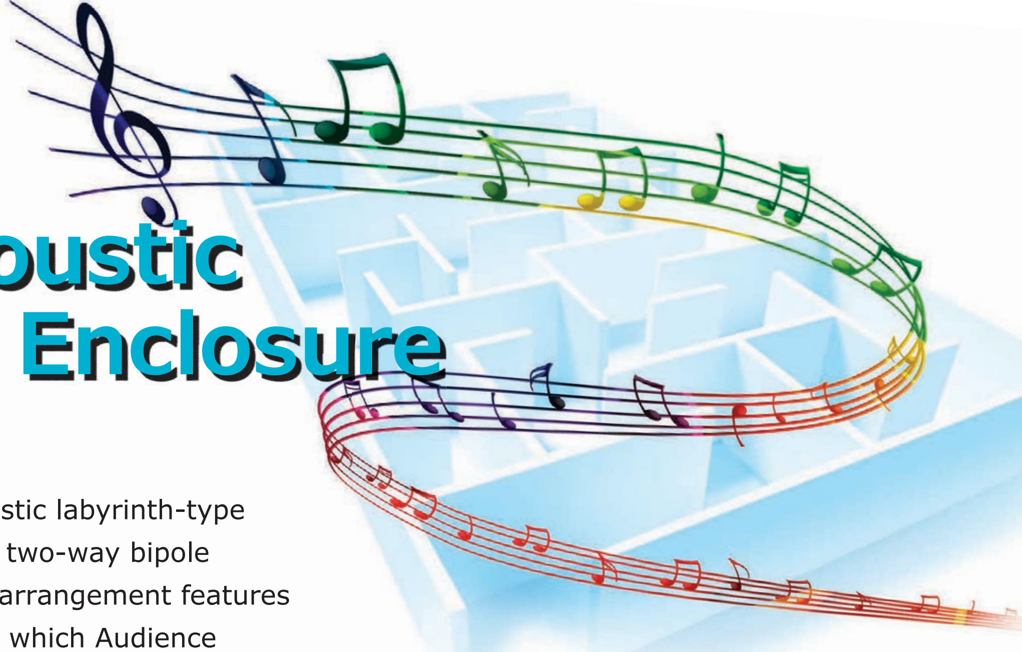
Final Thought

Perhaps I should have named this article "Psychoacoustics 101" because it is just an introduction to this interesting field rather than a "grand tour." Internet searches will provide you with more information than you probably ever wanted to know!

Author's Note: The wave files mentioned in this article are available from the author on a free CD. To request a copy, an e-mail info@tdl-tech.com with "psychoacoustics" in the subject line.

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 TDL's president and principal designer.



Bipole Acoustic Labyrinth Enclosure

This project is a quasi-acoustic labyrinth-type line that uses a chambered two-way bipole speaker arrangement. The arrangement features the A3S full-range speaker, which Audience developed for use in its line of high-end line-array and bookshelf speakers.

By
Ken Bird
(United States)

Benjamin Olney of radio manufacturer Stromberg Carlson patented the acoustic labyrinth speaker system in 1936. The system was incorporated into the pre-World War II high-end audio equipment of the day. The configuration was a folded tunnel whose length equaled the speaker's resonant frequency (Fs). Olney specified that the cross-sectional area be at least one-half or the full effective area of the speaker cone. The system worked well with the 10" and 12" dynamic speakers used in the 1930s with a resonant frequency that was usually 60 Hz at the lowest. At that frequency, the tunnel needed to be about 5′ long requiring materials and assembly line time that raised costs beyond that of the usual open-back console radios marketed during the 1930s. The design fell out of favor after World War II, as reflex, horns, and sealed designs began to dominate the audio market.

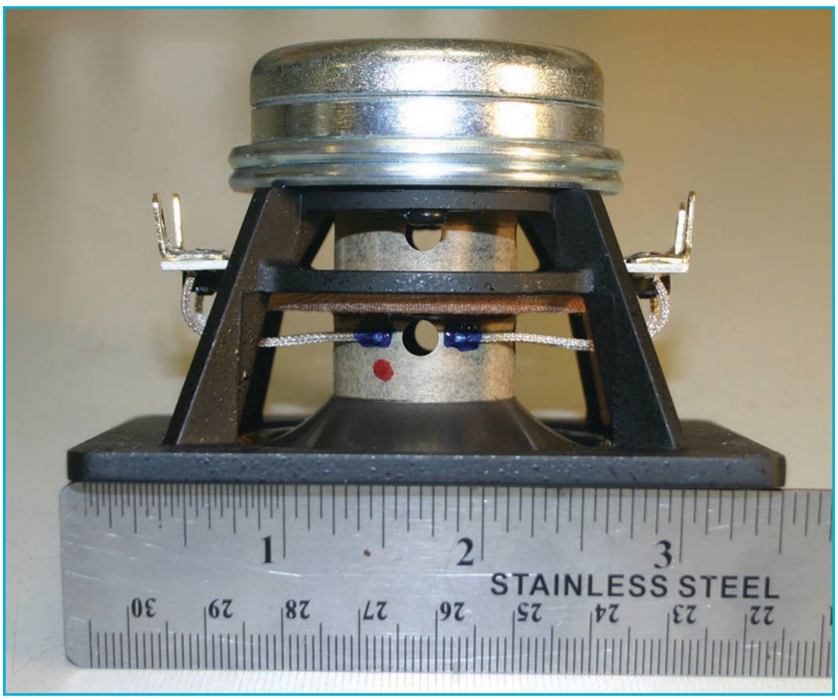


Photo 1: I used an Audience A3S 3" dual voice coil driver for this project.

T. L. Bailey described a variation of the acoustic labyrinth design in "A Non-Resonant Loudspeaker Enclosure Design" (*Wireless World,* 1965). The article prompted new interest in the labyrinth concept, which was then dubbed the transmission line. But, the design has always bedeviled amateur and professional speaker designers. Bose made the only major commercial application, with its line of Acoustic Wave radios and CD players.

Stuffing, length, and variable speaker specifications made the enclosures difficult to produce. Recent papers and articles by audio engineering veterans such as George Augspurger, and Martin J. King have given transmission line builders better design criteria, yet the final product is left up to the listener's subjectivity. Those who have built transmission line systems based on criteria from Augspurger and Martin have experienced mixed results. For this project, I employed the classical design method using some

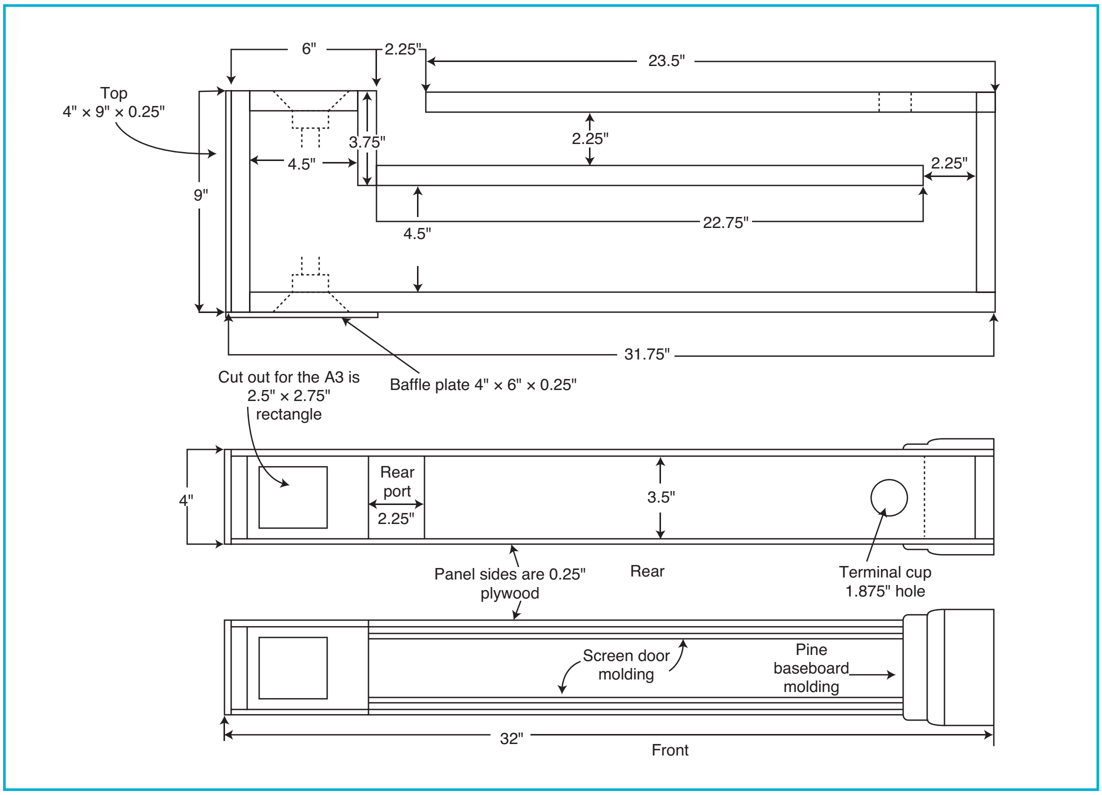


Figure 1: Draw out the enclosure plan's dimensions before you begin the project.

of the guidelines described in Olney's original 1936 labyrinth patent description.

My bipole quasi-acoustic labyrinth enclosure uses the Audience A3S full-range speaker, which features a large proprietary neodymium motor coupled to a lightweight titanium cone with 64 mm of effective piston area and a 25-mm voice coil (see **Photo 1**). The dual voice coil driver provides flexibility for multiple driver use. The 12-mm X_{MAX} is larger than many subwoofers, and the speaker's response is exceptionally flat from 40 to 22 kHz. The A3S costs $190, which may seem expensive for a 3" speaker. However, it is much less costly than a couple of Lowther full-range drivers and the A3 provides better performance specifications.

The Enclosure

In a pure acoustic labyrinth, the tunnel is not fully stuffed with acoustic absorption material. Instead, the tunnel's walls are lined with a sound-absorbent material and the tunnel has the same linear cross section to the port area. In the early 1970s, I was employed at Admiral. The company marketed the "Tunnel Reflex" system, which was a licensed variation of the acoustic labyrinth. The "Tunnel Reflex" system was later renamed the "Bull Horn" system so it would sound less "technical."

The Admiral Bull Horn used a 48" long linear duct, which was about the same cross-sectional area of the speaker. The tunnel was lined with 0.25" felt material. It used a 4" full-range driver that had a large Alnico magnet with a 70-Hz Fs. The system achieved excellent low frequencies in the lab. When used in the average home living room, the speaker sounded as big as console stereos with 8" or 12" speakers.

This system design differs from the original labyrinth description, in that the drivers are located in a chamber stuffed with polyfill loaded by a

About the Author

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

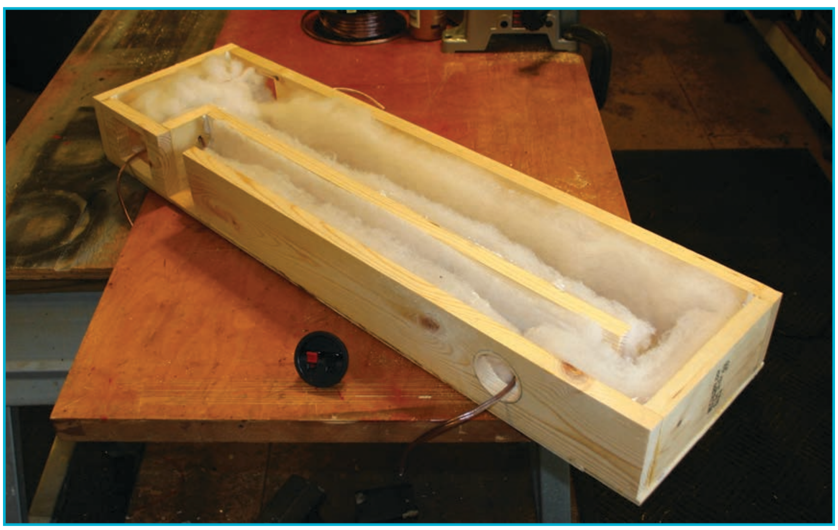


Photo 2: The enclosure is shown with an acoustic lining and the wiring installed.

Photo 3: The completed enclosures are shown in front and rear views.

tunnel whose cross section is halved at the midpoint. The tunnel's linear tapering tends to mass load the drivers. With the tunnel's total length at 50″, it tunes the enclosure to 65 Hz. As with the Bull Horn, the tunnel is not stuffed but rather is lined to absorb the harmonic nodes produced in the tunnel. Although not intentional, some may see a similarity to an older design called the Daline.

I used foam core board to build a pair of prototype speakers. The results indicated the speakers could be reproduced in wood. (See the sidebar article about using foam board for prototyping.) The area of the two drivers, or Sd, is 9.9 in^2. The tunnel area is 17.75 in^2 for the first half and 8.75 in^2 for the last half. I stuffed the chamber with 3 oz of polyfill and lined length of the entire area with 0.5″ polyester blanket.

Construction

For the enclosures, I used 1″ × 4″ pine board to build the frame and 0.25″ plywood for the sides. The enclosures' compact size does not require any heavier material for the sides but hardwood could be used for the box frame. The baffle front plates and enclosure sub tops were cut from 0.25″ plywood and rounded with a router (see **Figure 1**).

First, cut the individual frame pieces and then use wood glue and 0.07″ drywall screws to assemble them. Use a table or miter saw to make an accurate butt joint. Use corner clamps to hold the pieces and drill guide holes for the screws. The dimensions of the rectangular speaker mounting holes are 2.5″ × 2.75.″ Use a jig saw to cut the front baffle plates and frame pieces before assembling the frame. The speaker terminal cup that I used required a 1.875″ hole in the rear bottom section.

With the frame assembled, place the frame on top of the right-hand 0.25″ plywood side. (The front of the enclosure has the single speaker hole. The rear of the enclosure has the speaker and the port holes.) Line it up carefully with the frame and then trace the tunnel's outline and the chamber area on the side. These will serve as guidelines to position the tunnel on that side. Set that piece aside and run a bead of wood glue over the frame sides and attach the left-hand plywood side. Use 0.5″ brad nails, spaced every 4.″ Nail the side to the internal tunnel wall spacers. Use a nail set to push the nail head 0.0625″ below the wood surface. This gives the enclosure a rigid structure that will not audibly resonate.

The enclosure must be pre-wired, caulked, and lined with the polyester material before the other side is attached. Run a bead of window caulking along all the enclosure's interior seams. The tunnel is narrow and needs to be hand caulked. With the caulking complete, pre-wire the enclosure with 18-gauge speaker wire stapled in place along the tunnel walls to prevent any possible vibration.


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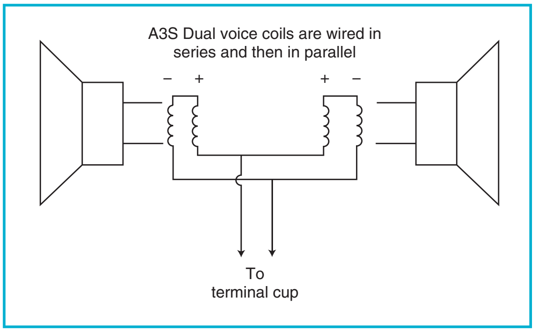
Figure 2: This diagram shows the enclosure's system wiring.



Lining the Tunnel

Line the tunnel with a 0.5″ polyfill blanket liner. Cut the material into 4″ wide strips and use craft glue to attach them to the tunnel's sides. The best technique is to dribble the glue onto the strips and press them in place along the tunnel walls. Line the entire length and the interior of the chamber walls (see **Photo 2**). Next, glue the lining on the inside of the right hand side following the earlier guidelines. Try to keep the material trimmed 0.25″ away from the drawn lines.

With the lining complete, attach the side by first running a 0.25″ bead of the window caulking along the frame sides and tunnel partitions. Keep a damp rag handy to wipe off any excess. Use 0.5″ finishing nails and a nail set to attach the remaining side.

Next, attach the sub top and the baffle plate. Glue the exterior top panel on and use window caulk to attach the front plate. Place four 0.5″ nails in each corner. Wrap the front plywood edges with molded screen door trim and use 3.5″ high pine baseboard to widen the bottom of the enclosure. Use wood filler to cover the set nail holes and exposed screw holes. Set the enclosure aside to dry for an hour or so.

Finishing

Sand the entire enclosure with 150-grit sandpaper. Prior to sanding, stuff newspaper into the open speaker holes and port areas to keep dust out of the chamber. I stained my enclosures with a blend of Mahogany and Walnut that I used on an earlier project and painted the back a flat black. I gave the enclosures a final finish of two coats of gloss polyethylene varnish. **Photo 3** shows the front and back of the completed enclosures.

Driver Installation and Measurements

The A3S drivers' dual voice coils are wired in series on each speaker providing a 16-Ω impedance per unit. This produces a system total of 8 Ω when the divers are wired in parallel. Contouring or crossover components are not needed. The simple wiring maintains the integrity of a full-range system. Solder all the connections to the speaker and the terminal cup to ensure the polarity is correct. You can check the polarity by applying 1.5 VDC from D-cell battery. Just tack solder some wire leads to the battery terminals. Attach the negative battery lead to the negative terminal on the speaker cup and intermittently touch the positive terminal to the other cup while observing the speaker cones. They should both move outward when the positive lead makes a connection (see **Figure 2**).

I used #4 × 0.5″ sheet metal screws to install the drivers and the terminal cup. To ensure a good seal, I cut foam self-stick insulation material into 0.125″ strips and applied it to the rear of each driver's frame. Once screwed down, the material flattens out and fills any gaps in the baffle area. This provides a good seal and simplifies driver removal. The terminal cup I used is equipped with a foam gasket and will not need any further material for a good seal.

System Performance

The RTA system I used to measure the system's near-field and in-room response showed it to be

Use Foam Board to Prototype Speakers

A fast way to test your speaker enclosure design is to use foam core board to build a prototype. The foam board is available at craft stores and come in thicknesses from 0.1875″ up to 0.5″ and a variety of colors.

Most projects, using up to 5″ full-range drivers, can be built using the 0.1875″ board marketed under the Elmer's brand. The board can be cut with an Xacto or carton knife and solidly glues together using tacky craft glue. Wood sub baffles must be used to better mount the speaker.

Photo 1 shows a set of foam board prototypes that evolved into the bipole line described in the article. They were constructed with the 0.1875″ Elmer's foam core board. By using the foam board, you can perform measurements and subjective testing without buying and cutting wood.

Photo 1: Prototype enclosures can be built from foam board.

Resources

T. L. Bailey, "A Non-Resonant Loudspeaker Enclosure Design," *Wireless World*, 1965.

Benjamin Olney-Stromberg Carlson Co., US Patent 2,031,500, 1936, www.google.patents.com.

Source

A3S 3″ drivers
Audience | www.audience-av.com

usable audio down to 40 Hz. Using two drivers in a bipolar arrangement maintains efficiency as both drivers fire in phase, while in opposite directions. This creates a broad coherent sound stage while maintaining fairly good imaging. In addition to using them in 2.1 audio systems, the setup can be used as the rear speakers in a surround-sound system.

The audio purist who does not use any form of equalization will have to rely on experimental positioning within the room environment to achieve the best performance. The use of the bass and treble controls can be beneficial with this system. In most rooms, the speakers should be positioned 12″ to 18″ from the wall or placed in a corner location.

The speakers can produce bass in the lower octaves with incredible power for their size, as seen in the 12-mm excursions of titanium drivers. You would expect to hear a lot of intermodulation

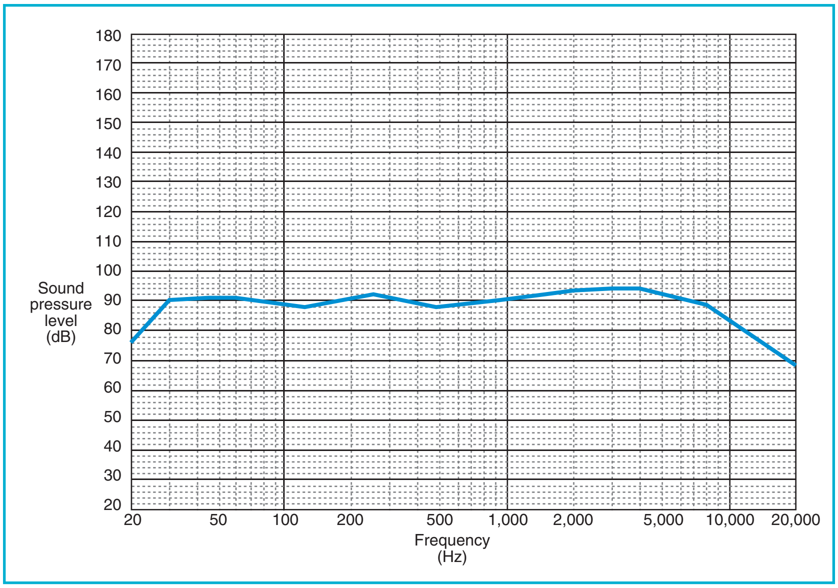


Figure 3: The in-room response of both enclosures is shown here.

(IM) distortion with that amount of bass excursion, yet the midrange is clear, the imaging is superb, and the treble sparkles when it should. The A3S drivers produce a high-performance full-range system without the need for a crossover network, at a fraction of the cost of a comparable factory-built system. **Figure 3** shows the room response of both enclosures with all tone controls flat.

Materials List for Two Enclosures

The dimensions specify 1″ × 4″ pine or hardwood boards—3.5″ wide and 0.75″ thick (see **Figure 1**)

Building Materials
Front frame piece: two each 33″ long
Interior partition: two each 24″ long
Frame top: two each 9″ long
Frame bottom: two each 7.5″ long
Frame lower rear: two each 24.5″ long
Rear baffle: two each 5.25″ long
Chamber shelf: two each 3.75″ long
0.25″ Plywood: four pieces 9″ wide × 33″ long
Optional: One 48″ piece of pine basboard trim, cut to length for enclosure bases

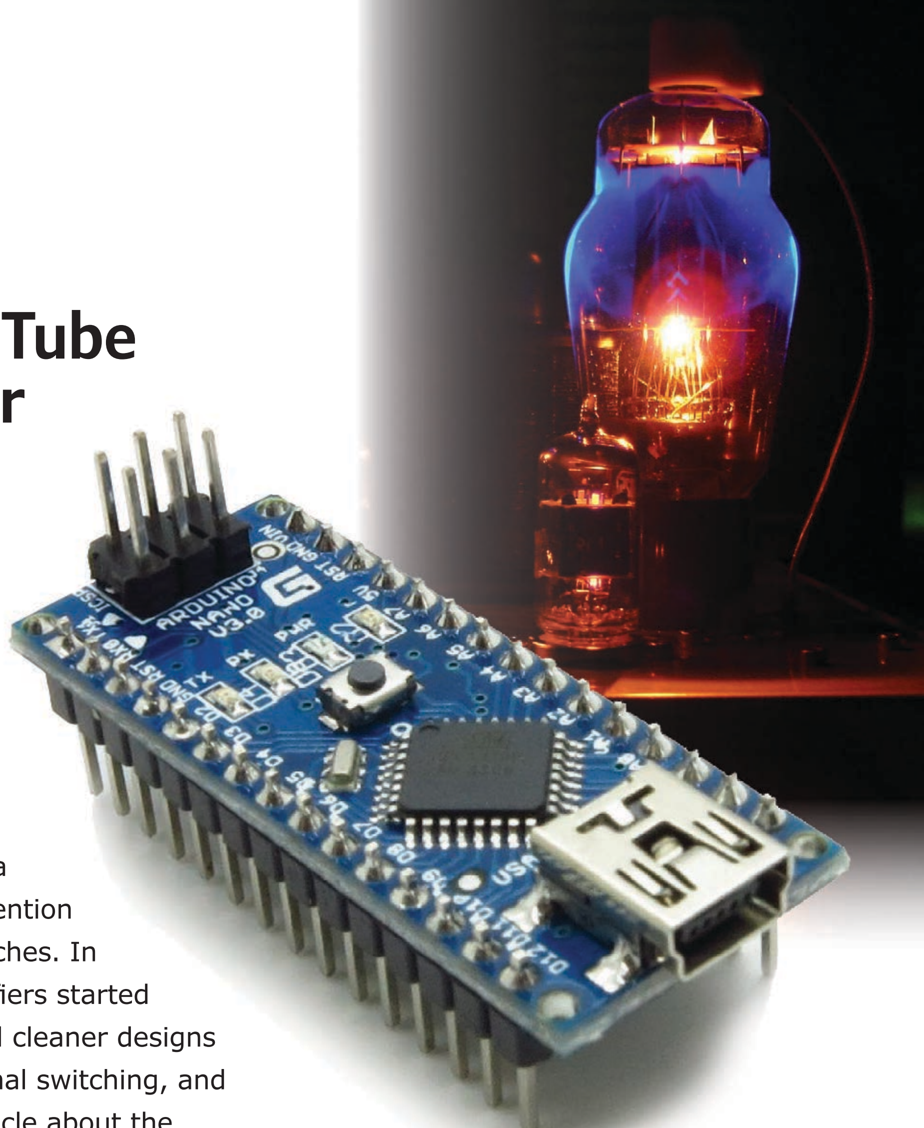
Subwoofer top: two each 4″ × 9.25″
Front baffle plate: two each 4″ × 6″

Speakers
Audience A3: four each 3″ drivers
Speaker Terminals: two each (Part Express #269-292)
18-gauge speaker wire: 8′

Other Materials
Polyester blanket filling: 2 yd
1.25″ Drywall screws: 24
0.5″ Brad nails: 100
Wood glue
White or brown silicone window caulk: 1 tube

Budgie

An Arduino-Based Tube Stereo Preamplifier



Over the last 10 years, I have built many tube power amplifiers but I had never built a tube preamplifier. The source switching seemed particularly daunting. A friend recommended that I refurbish a classic Dynaco PAS-3 which has been a popular choice with many upgrade kit suppliers. Unfortunately, the main part of these older designs is a clumsy rotary selector switch, not to mention the noisy potentiometers and slide switches. In the 1980s, commercial stereo preamplifiers started using IC microcontrollers that permitted cleaner designs with push-button control, relays for signal switching, and a wireless remote. While reading an article about the Arduino last year, I realized these modern features could easily be incorporated into a DIY preamplifier design.

By
Shannon Parks
(United States)

Photo 1: The Budgie preamplifier contains a single-ended, Class-A, all vacuum tube audio path.

In recent years, open-source Arduino prototyping boards have become an attractive option for hobbyists. You can start tinkering with an Arduino for less than $10 and a programmer tool is not needed. All you really need is a PC with a USB cable. The intuitive Arduino software enables you to compile and load programs called "sketches" onto the Arduino board. It consists of an integrated development environment (IDE) and C-like programming language and is compatible with Windows, Mac OS, and Linux. Arduino "shields," which are add-on modules, permit fast prototyping with many I/O capabilities. During a single afternoon, a useful gadget—possibly a garage door monitor or wireless attic temperature sensor—can be assembled with minimal effort. Browsing the Internet, you can find applications that range from ingenious to ridiculous. The Arduino prototyping system is incredibly versatile.

Several of the Arduino's most often used capabilities became the genesis of my Budgie preamplifier (see **Photo 1**). I found these features were incredibly useful:

- A bank of relays could switch between the four stereo inputs as well as control mute, standby, gain, and bass boost settings.
- A red power LED could use PWM to indicate if the preamplifier is muted or in standby.
- An IR receiver with a remote could control a motor-driven volume potentiometer, change the source input selection, and turn the unit on/off. Any IR remote could be used with a code learning mode.
- A backlit display could easily show all the settings at a glance.
- Momentary push buttons could select the input device, bass boost, gain, and mute settings.
- Instead of using several Arduino shields wired to an Arduino board, all the circuits could fit on one custom PCB along with the power supply and the microcontroller (see **Photo 2**).

While there are many versions of Arduino boards, I chose the Arduino Nano. At only 0.73″ × 1.70″, the tiny Nano can be embedded using a 32-pin dual in-line package (DIP) socket, which cleans up the design. It can be programmed in-circuit and be removed and easily replaced (see **Photo 3**).

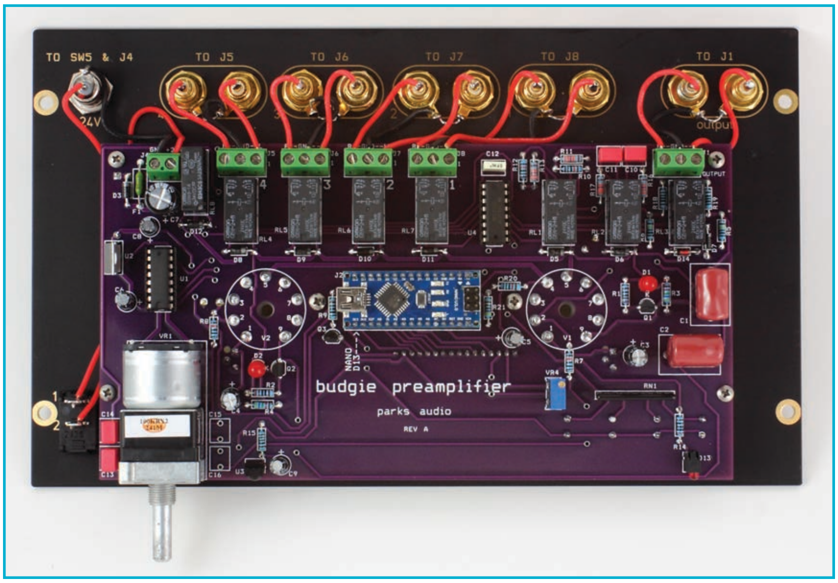


Photo 2: All the circuits have been placed on one custom PCB along with the power supply and the microcontroller.

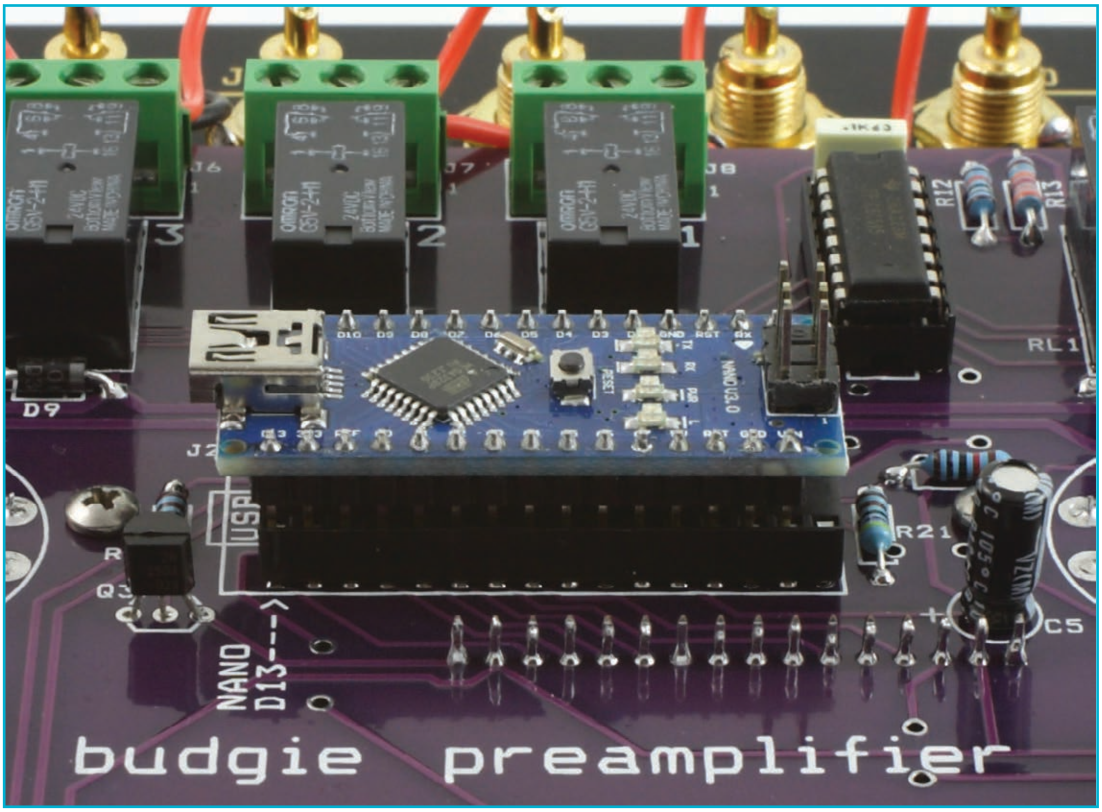


Photo 3: The Arduino Nano is a small, complete, and breadboard-friendly board based on Atmel's ATmega328 microcontroller.

The audio circuit remains the heart of any preamplifier. With a little creativity, the well-regarded 12B4 triode can be used. Its high perveance means that current will flow with a much smaller cathode-plate voltage differential than is typical with vacuum tubes. An external 24-V desktop supply can provide all the voltages the tube requires without need for further regulation, which simplifies the design.

The Arduino boards, as with all microcontrollers, are limited by their number of I/O pins, which are simply referred to as "pins." The Nano has 14 digital pins (D0 to D13) and eight analog pins (A0 to A7). Digital pins are either low (0 V) or high (5 V). As an output, a digital pin can source up to 40 mA and do tasks (e.g., turning on an LED or forward biasing a transistor switch). A digital pin can be set up as an input to process the remote control code stream of digital 1s and 0s received from the IR sensor.

The analog pins have an internal ADC, which enables 1,024 values (10-bit resolution) between 0 and 5 V. This is useful for reading voltages in approximately 5-mV increments. The Budgie preamplifier uses one such input to monitor a voltage divider circuit implemented with five 1-kΩ resistors in series. This pin receives four unique voltage levels from the push buttons to determine which was pressed. A special feature of the analog pins is that they can also be configured as digital pins. The Budgie does this for the display. For this application, A0 is renamed D14, A1 is renamed D15, and so on.

Shift Register Circuit

The Budgie preamplifier uses a serial-in, parallel-out (SIPO) shift register to drive a bank of relays (see **Figure 1**). Only four Arduino digital outputs—enable, clock, latch, and data—are needed to control eight DPDT relays. These correspond to the four outputs labeled D3, D4, D5, and D7 shown in **Figure 2**. The Texas Instruments TPIC6C595 shift register used in this project has heavy-duty field-effect transistor (FET) outputs that can handle voltages higher than logic levels. This is necessary for operating the 24-V relays. It also acts as a protective buffer between the Arduino and the relays.

Power LED

The red power LED is directly driven by the Arduino (see **Figure 2**). Normally the output (D6) is high with the maximum LED brightness limited by a 200-Ω resistor that biases the LED at about 16 mA. When the preamplifier output is active, the PWM control is not used as it can add 400-Hz noise (the PWM switching frequency) to the audio

About the Author

Shannon Parks built his first tube amplifier—a clone of the Dynaco ST35—in 2001 and was immediately hooked. He manages the DIY-tube forums and is the owner of Parks Audio, LLC (www.parksaudiollc.com). He lives with his wife and daughter in Mahomet, IL.

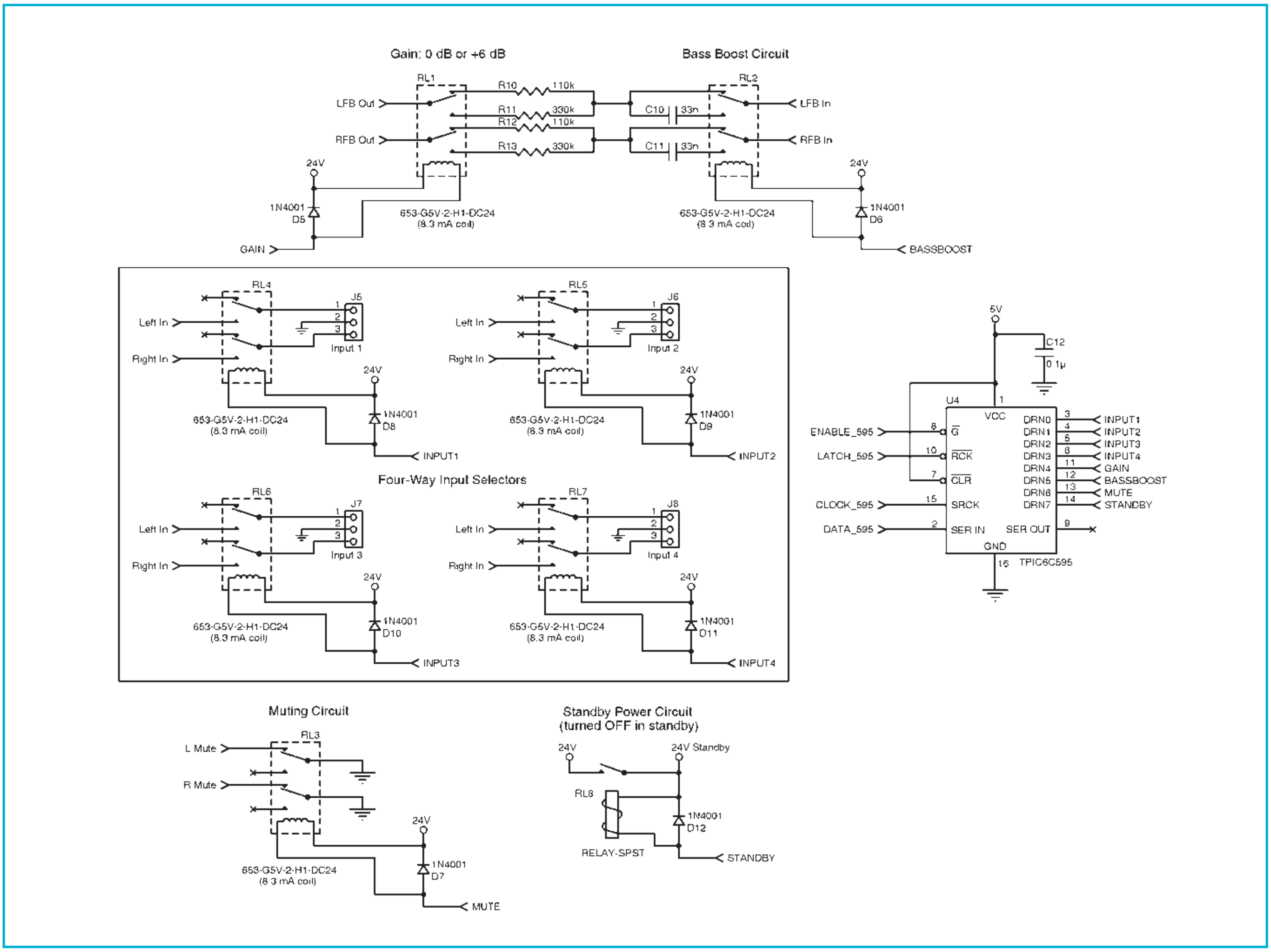


Figure 1: A serial-in, parallel-out (SIPO) shift register is used to drive a bank of relays.

circuit. However, it can be used when the output is muted or in standby.

IR Remote Circuit

The D8 input is the only digital pin configured as an input (see **Figure 2**). An IR sensor with a built-in preamplifier sends all sensed IR commands from a remote control to the microcontroller.

A special software library called "IRremote" simplifies the handling of the raw datastream into 4-byte codes. This data can be processed in real-time or stored in the EEPROM during learning mode.

Motor Driver for the Volume Potentiometer

A Texas Instruments SN754410NE specialized integrated circuit (IC) controls the motor of the Alps volume potentiometer and acts as a buffer between the microcontroller and a motor. The D10, D11, and D12 outputs are used. The IC simplifies a process that would otherwise need careful design. Two digital pins control the direction of the motor on the volume potentiometer while a third digital pin enables the motor driver.

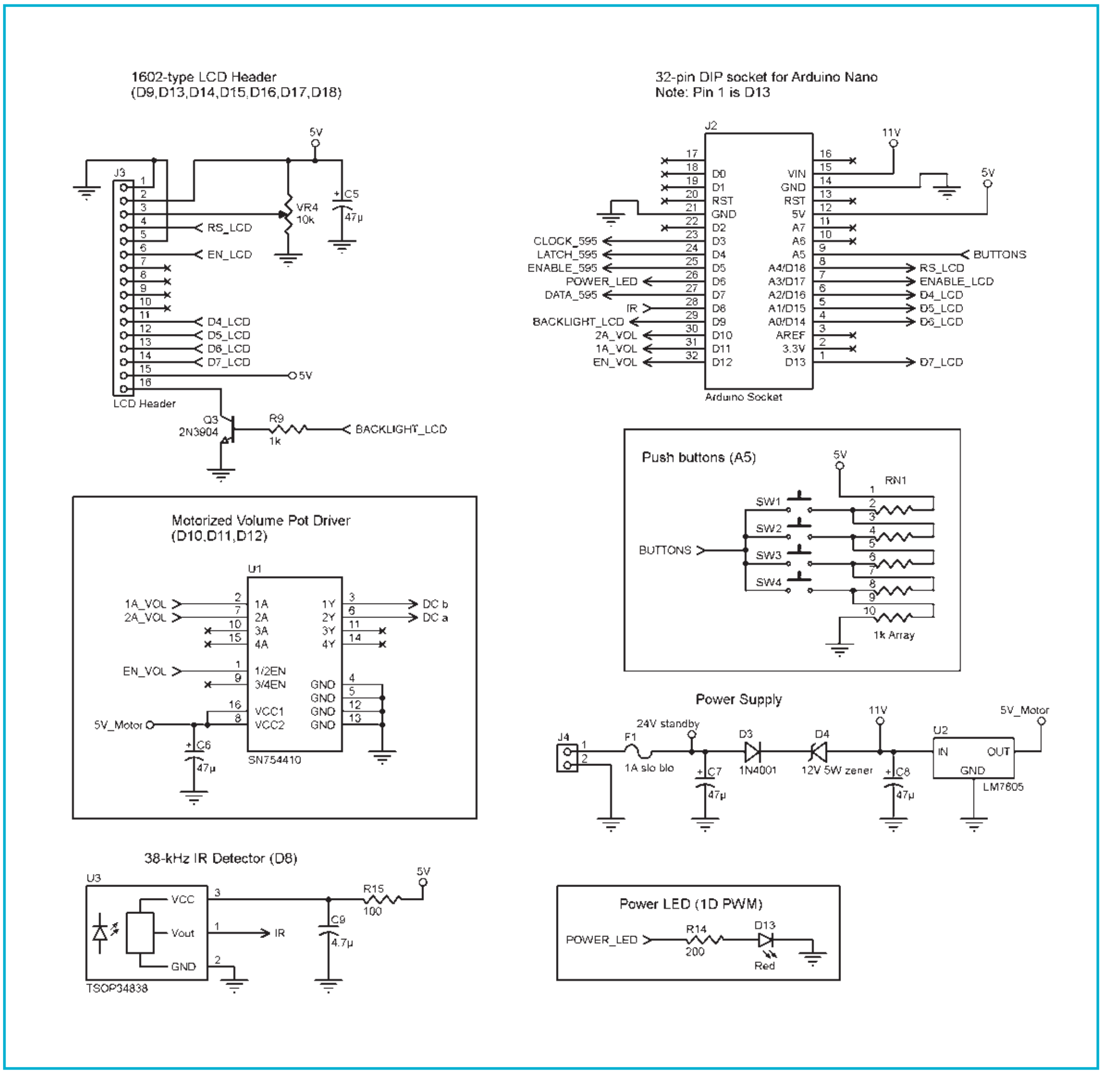
1602 LCD

The 1602 LCD—16 characters per line by two lines—is probably the most used display in the Arduino realm. It is controlled using the standard software library "LiquidCrystal" in 4-bit mode, so a total of seven digital pins are needed: register select (RS), enable (EN), backlight, and four data lines. These are assigned to Nano outputs D9, and D13 through D18 (see **Figure 2**).

Push Buttons

A voltage divider string of resistors is connected to four push buttons: input, bass boost, gain, and mute (see the A5 input in **Figure 2**). Each button has a different voltage level that the analog input reads to determine which button was pressed. Multiple

Figure 2: These Budgie diagrams show how to set up the Arduino Nano, LCD, power supply, push button, IR, and motor control circuits.



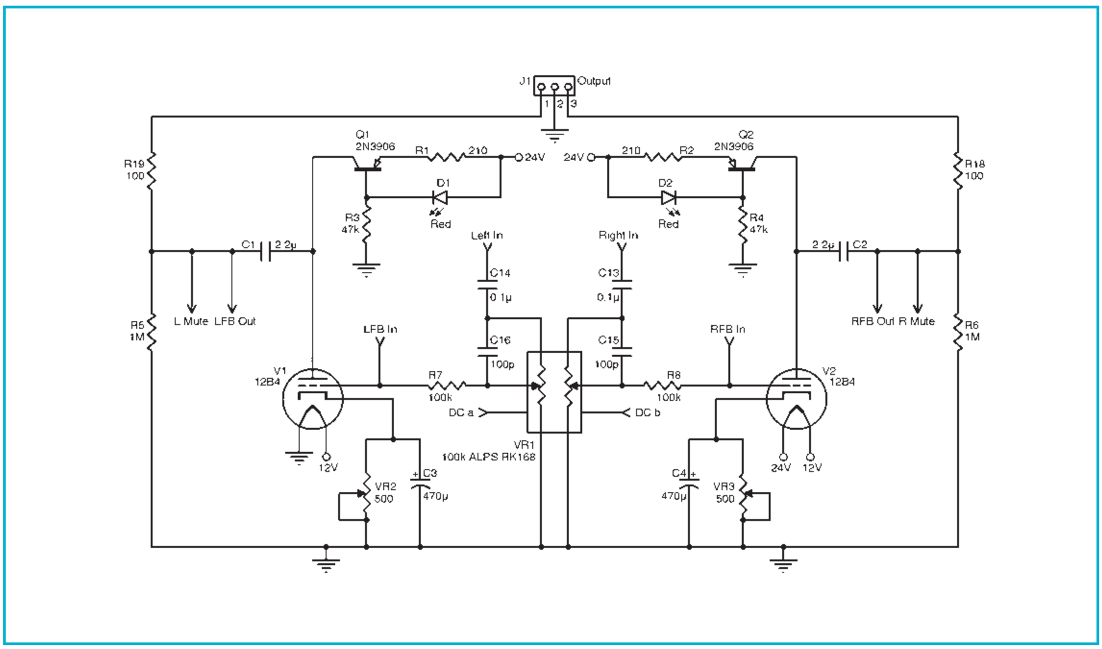


Figure 3: The 12B4 audio circuit has about 23-dB gain, which is more than is needed.

Budgie Preamplifier Parts List

Capacitors

C1, C2	2.2 µF, 250 V, 18.8 × 12.6-15LS	667-ECW-F2225JA
C3, C4	470 µF, 10 V, 6.3D 2.5LS	647-UVZ1A471MED
C5, C6, C8	47 µF, 35 V, 5D 2LS	647-UVZ1V470MDD
C7	100 µF, 35 V, 6.3D 2.5LS	647-UVZ1V101MED
C9	47 µF, 50 V, 5D 2LS	647-UVZ1H4R7MDD
C10, C11	6,800 pF, 63 V, 7.2 × 4.5-5LS	505-FKP26800/63/2.5
C12	0.1 µF, 63 V, 7.2 × 2.5-5LS	80-R82DC3100AA50K
C13, C14	0.1 µF, 100 V, 7.2 × 2.5-5LS	505-MKP20.1/100/5

Chassis

Hammond 6 x 10 x 2 Black Chassis		546-1441-16BK3
Quantity 2	Clear window plug	593-LPC020
Quantity 2	Clear window plug retainer	593-RTN150

Circuits

U1	Half-H motor driver	595-SN754410NE
U2	5-V regulator 7805	595-UA7805CKCT
U3	IR detector, 38 kHz	782-TSOP34838
U4	Power shift register	595-TPIC6C595N
Quantity 2	16-pin DIP socket	571-1-390261-4

Diodes

D1,D2	T-1 0.75 5-mm red LED	859-LTL-4223
D3,D5-D12	1N4001 50-V 1-A diode	512-1N4001
D4	12-V, 5-W Zener diode	863-1N5349BRLG
D13	Red power LED	645-551-0407F

Miscellaneous

J1, J5–J8	Terminal blocks 3P	571-2828363
J2	32-pin DIP socket	517-4832-6000-CP
J3	16-pin SIP header	649-68001-416HLF
J4	Terminal blocks 2P	571-2828372
SW1–SW4	Round push-button switch	10KB012
SW5	SPST on/off rocker	540-SRB22A2FBBNN
Quantity 1	Arduino Nano V. 3.0	992-ARD-NANO30
Quantity 5	RCA panel jacks—white	568-NYS367-9
Quantity 5	RCA panel jacks—red	568-NYS367-2
Quantity 1	2.1-mm DC connector	163-4021
Quantity 4	Rubber feet	517-SJ-5023BK
Quantity 6	Hex standoff 0.187 × 0.375	534-1892
Quantity 4	#6-32 × 0.5″ sheet metal screw, Phillips pan head	
Quantity 12	#4-40 × 0.25″ machine screw, Phillips pan head	
Quantity 1	Red LCD 1602 display module (HD44780 or compatible)	
Quantity 1	30 × 22 Aluminum volume knob	
Quantity 2	12B4/12B4A triode	
Quantity 2	9-pin Ceramic tube socket PCB (0.75″ top diameter)	
Quantity 1	24-V/1-A DC supply (three prong)	

Resistors

R1, R2	210 Ω, 0.25 W	271-210-RC
R3, R4	47.5 kΩ, 0.25 W	271-47.5K-RC
R5, R6	1 MΩ, 0.25 W	271-1.0M-RC
R7, R8	100 kΩ, 0.25 W	271-100K-RC
R9	1 kΩ, 0.25 W	271-1K-RC
R10, R12	110 kΩ, 0.25 W	271-110K-RC
R11, R13, R16, R17	332 kΩ, 0.25 W	660-MF1/4DC3323F
R14	200 Ω, 0.25 W	271-200-RC
R15, R18, R19	100 Ω, 0.25 W	271-100-RC
R20	10 kΩ, 0.25 W	271-10K-RC
R21	4.7 kΩ, 0.25 W	271-4.7K-RC
RN1	1 kΩ, 10-pin array, isolated	269-1.0K-RC
VR1	100 kΩ, dual log ALPS RK168	688-RK16812MG099
VR2, VR3	500-Ω potentiometer	72-T93YB-500
VR4	10-kΩ potentiometer	72-T93YB-10K
F1	1.5-A fuse, Slo-Blo	576-047301.5MRT1L
RL1-RL7	Omron G5V 24-V DPDT	653-G5V-2-H1-DC24
RL8	Omron G5NB 24-V SPST-NO	653-G5NB-1A-DC24

Transistors

Q1,Q2	2N3906 PNP	512-2N3906BU
Q3	2N3904 NPN	512-2N3904BU

Budgie PCBs are available from Shannon Parks (www.parksaudiollc.com/arduino). Part numbers in the third column indicate the items are available from Mouser Electronics (www.mouser.com).

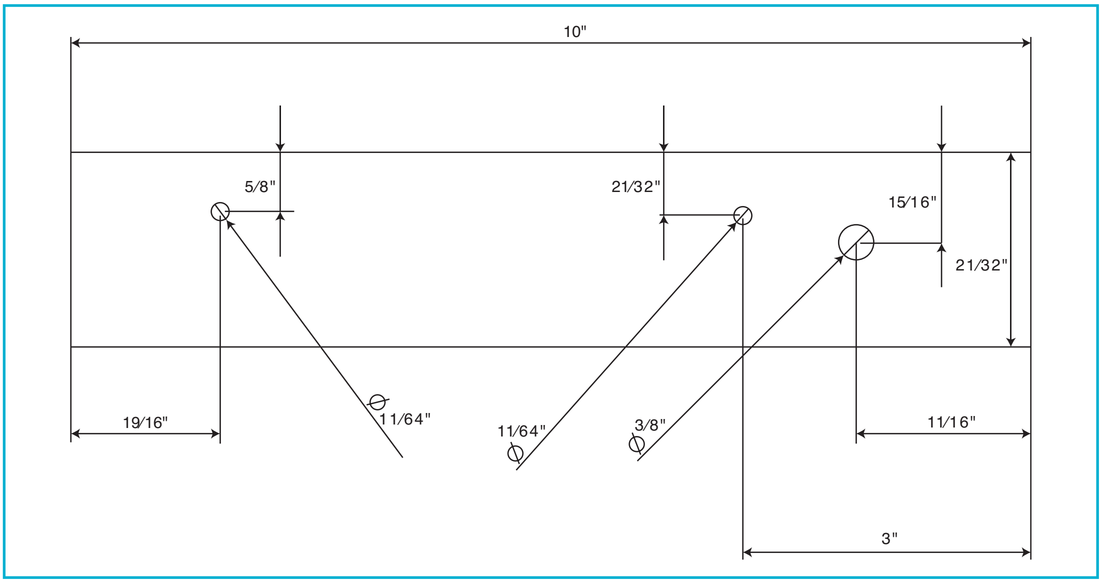


Figure 4: The template shows where to drill the holes when assembling the Budgie's chassis.

keys pressed at the same time will generate unique voltages and be processed as special functions.

Power Supply

The external 24-V desktop supply must be a Class-I grounded type and should be rated above 1 A. Some supplies may have a tendency to trip their current limit at initial power-up due to the heavy inrush from the cold tube filaments. The relays and tubes operate at 24 V without needing further regulation to eliminate wasted power and heatsinks on the Budgie PCB. The 24-V supply also powers two lower current 5-V supplies. A polarity diode and then a 12-V power Zener diode drop the incoming 24 V to around 11 V so the linear regulators don't dissipate too much heat from the voltage differential (see **Figure 2**). The 7805 regulator on the PCB (U2) is solely for the motorized Alps volume potentiometer. Meanwhile, the 5-V regulator on-board the Nano powers all other 5-V circuits including the display backlight.

Audio Circuit Description

The 12B4 triode was originally designed to be used in televisions as a vertical deflection amplifier. New-old-stock (NOS) 12B4s still exist. They can be purchased from most US tube resellers. However, a European equivalent doesn't exist. The 12B4 works well in preamplifiers as a one-tube solution, having both high input impedance and low output impedance, without need for an output transformer. An audio circuit can then be distilled down to a simple circuit with few parts consisting of a volume potentiometer and a grounded cathode gain stage.

The 12B4 has about 23-dB gain, which is more than is needed. This extra gain is used as feedback to the grid, in what is often referred to as an anode follower circuit. The noise, distortion, and output impedance are reduced (see **Figure 3**). Using relays controlled by the Arduino enables switching between two feedback amounts for adjustable gain. For this preamplifier, I chose 0- and 6-dB overall gain. A second relay enables a bass boost with a series capacitor.

You only need a lightweight 15-to-20-V plate voltage to operate the 12B4s at 5 mA. Linearity is very good due to the small signal levels involved, as rarely will the output be greater than 2 V_{PP}. A constant current source (CCS) active load is used with the 12B4s instead of a traditional plate resistor. This maximizes the possible output voltage swing before clipping. For example, a 12B4 biased at 5-mA plate current with a 20-kΩ

Resources

Future Technology Devices International (FTDI), Ltd., www.ftdichip.com.

Parks Audio, LLC, www.parksaudiollc.com/arduino.

Sources

RK168 Series volume potentiometers
Alps Electric Co., Ltd. | www.alps.com

Arduino Nano
Arduino | http://arduino.cc/en/main/software

ATmega328 Microcontroller
Atmel, Corp. | www.atmel.com

TPIC6C595 Shift register and SN754410NE IC
Texas Instruments, Inc. | www.ti.com

plate resistor would drop 100 V and would then require a 120-V supply voltage or higher. Conversely, the CCS will only drop about 2 V. Its naturally high impedance also improves the tube's gain and linearity while providing high levels of power supply noise rejection. A single resistor from the supply to the emitter sets the current, which can be calculated with the following formula:

$$\frac{(\text{voltage drop of red LED} - V_{BE})}{R_{SET}} = \frac{(1.7\text{ V} - 0.65\text{ V})}{210\ \Omega} = 5\text{ mA}$$

The adjustable cathode resistor is initially set to 300 Ω for good performance, but the lowest distortion can accomplished by adjusting the trim potentiometer to achieve maximum output signal. This tweak can be done with a free smartphone app such as Signal Generator outputting a 500-mV 1-kHz sine wave. Use any digital multimeter (DMM) set to AC to measure the output level and adjust the cathode potentiometer for maximum output. Note that the DMM doesn't need to be TrueRMS as the measurement is relative, not absolute. Also, don't forget to adjust the volume potentiometer all the way clockwise.

Chassis Assembly

The Budgie chassis uses an inverted Hammond powder-coated 6" × 10" × 2" box. I can provide a front panel express file for a ready-made top plate via my website (see Resources). Most of the chassis holes are on this one panel, but three holes must be drilled into the front of the chassis for the power LED, the IR sensor, and the volume control. A template with detailed dimensions facilitates drilling these holes (see **Figure 4**).

Compiling and Loading Arduino Code

The first time you connect the Arduino Nano to your computer with a USB cable, the device drivers for the Future Technology Devices International (FTDI) virtual com port will be prompted to load. Once you successfully install the drivers, start the Arduino software. First make sure you have your COM port set correctly in the Arduino software (Tools> Serial Port…). Then configure the software for the Nano: "Tools>Board> Arduino Nano w/ ATmega328." Then go to "Sketch>Import Library…>Add Library…" and select the IR remote library from the Budgie files folder, which decodes all the raw IR data. Clicking the "Upload" icon will first compile and then upload the sketch to your Nano's flash memory. After you've uploaded your sketch, disconnect the USB connection before you power the Budgie with 24 V to prevent reverse current into your computer's USB port and possible damage.

Completed Budgie

The entire project can be assembled for approximately $250 (see **Parts List**). As an introduction to the world of Arduino microcontrollers, it's a comprehensive starter kit. Elements of the design (e.g., the IR-controlled volume potentiometer) could be used as a stand-alone project. Once you experience the relative ease of microcontroller programming with the Arduino, you'll find no shortage of creative applications.

Linearity in Audio Preamplifiers

In last month's article, we discussed the design of a low-distortion audio preamplifier stage. One of the first steps in such a design is to plot the load line, which is the locus of all possible combinations of plate voltage and plate current, given the specific B+ voltage and plate resistor that are used in the circuit.

By
Richard Honeycutt
(United States)

Figure 1 shows the plate characteristic curves for one triode of an ECC83 vacuum tube, using a 300-V B+ and a 100-kΩ plate resistor. The stage's control grid is biased at a particular voltage (when no signal is applied), resulting in a particular plate voltage and current: the "quiescent" or Q point. As control-grid voltage is varied in a positive or negative direction by the input signal, the plate voltage and current will be changed, but the combination of plate voltage and plate current will always fall on the load line. If you graph the plate voltage vs. the control-grid voltage, as shown in **Figure 2**, you can see that at the most negative grid voltages, the graphed lines curve. Lines are plotted for three values of plate resistor: 100, 200, and 300 kΩ.

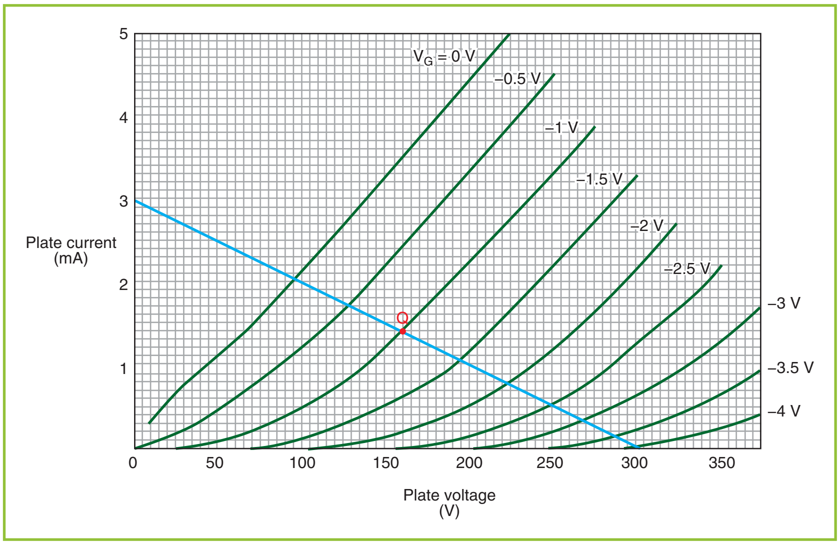


Figure 1: The load line of an amplifier stage shows all possible combinations of plate current and plate voltage.

The slope of each line, $\Delta V_P/\Delta V_G$, is the voltage gain. When the line is curved, the gain is changing as the control-grid voltage changes. Notice that smaller plate resistors are associated with more curvature, and hence, more distortion.

The reason for the curvature of the lines at more negative grid voltages is that the tube is approaching a cut-off condition, where the control-grid voltage is so negative that the plate current is completely stopped. The actual negative control-grid voltage necessary to produce cutoff is not shown on the graph, but it depends on the B+ voltage: the higher the B+, the more negative the $V_{GK(cutoff)}$. When the plate current is cut off, the output waveform is clipped at the top (See **Figure 3**). The plate signal voltage is drawn inverted in the figure, for easier comparison.

Large input signals can "try" to push the control grid positive. When this happens, the control grid draws electrons that would otherwise have gone to the plate, thus reducing the gain. Under these conditions, the grid and cathode act like a diode's plate and cathode, clamping the grid to a

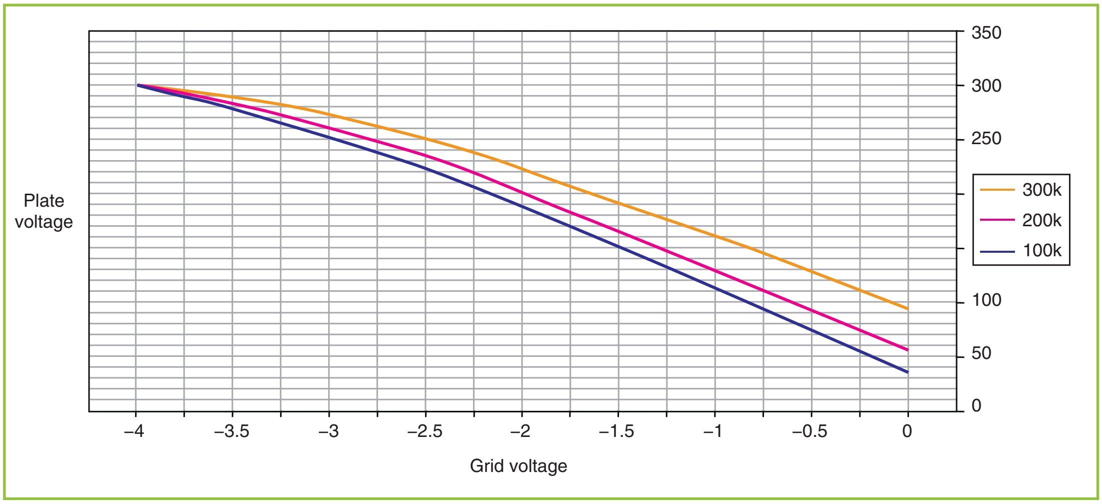


Figure 2: Any deviation from a straight line in the grid-voltage/plate-voltage curves causes distortion.

maximum of 0 V. When the control grid is biased so that it no longer can cause plate current to increase, the stage is said to be "saturated." Even when the stage is not saturated, if the positive half-cycles of the input signal cause grid current, a portion of the input signal voltage will be dropped across the source resistance, clipping the top off the input wave. **Figure 4** shows the resulting output wave, which is of course clipped on the bottom due to the signal inversion in the common-cathode amplifier stage.

Grid-Current Limiting

A comparison of **Figure 3** and **Figure 4** shows that grid-current limiting produces gentler clipping than does cutoff. The reason is the onset of grid current is gradual. An ECC83 vacuum tube will have a grid current of about 0.3 μA when the grid is 0.9 V less positive than the cathode. This value may be stated on datasheets as $V_{GK(MAX)}$. The ECC82/12AX7 has a slightly lower $V_{GK(MAX)}$ than the ECC83, and that of the ECC81/12AT7 is lower still. Grid-current limiting is reduced by higher B+ voltage, since a more positive plate can more easily "out-attract" electrons compared to the control grid. Very low source resistance can make it possible to drive the grid slightly positive, producing much gentler clipping that resembles soft compression. But the output impedance of guitar pickups, as well as other considerations, make it difficult to use this effect in a practical amplifier.

Determining Distortion

Designing an input stage for a guitar amplifier to produce a certain amount and type of distortion requires you to identify some preferences. In addition to how much distortion (%HD) you want at a particular signal level, you need to decide whether you like "soft" distortion and some added sustain that would sound good on chords, or a more aggressive, "hard" distortion for "metal" leads. Also, do you want symmetrical distortion—a lot of odd harmonics—or asymmetrical distortion that is composed mainly of even-order harmonics?

To design a preamplifier that produces soft distortion implies that you want the distortion to come from grid-current limiting, and not from cutoff. Even-order distortion is also perceived as more gentle than odd-order so we will bias the grid so that only a very high-amplitude signal would drive it to cutoff. A low B+ voltage will help achieve soft grid-current limiting.

Since grid-current limiting occurs because of changes in the waveform at the grid rather than because of changes in amplification by the triode, we can calculate the grid bias that will drive the stage to the desired distortion level. Let's aim at 25% distortion with a 0.4-V_P input level, which is a fairly typical level for a lead guitar playing one note at a time (no harmony), with two humbucking pickups, and the tone and volume controls on the guitar at maximum. The equation for approximate percent second harmonic distortion is:

$$\%2HD = \frac{\Delta V_{larger} - \Delta V_{smaller}}{2\left(\Delta V_{larger} + \Delta V_{smaller}\right)} \times 100$$

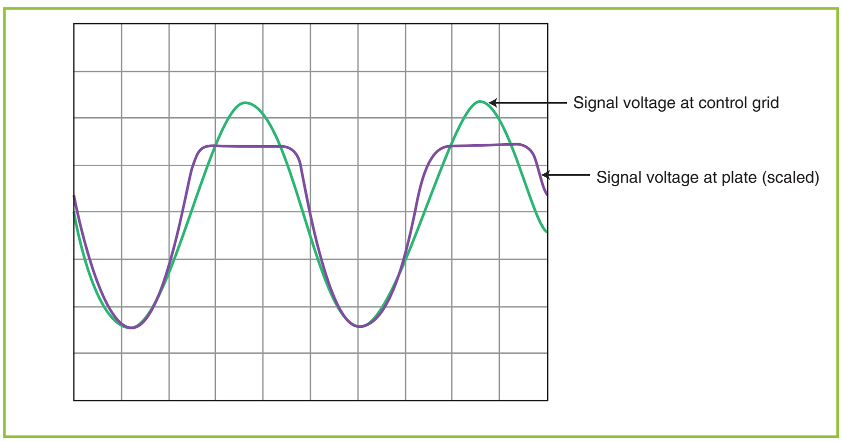
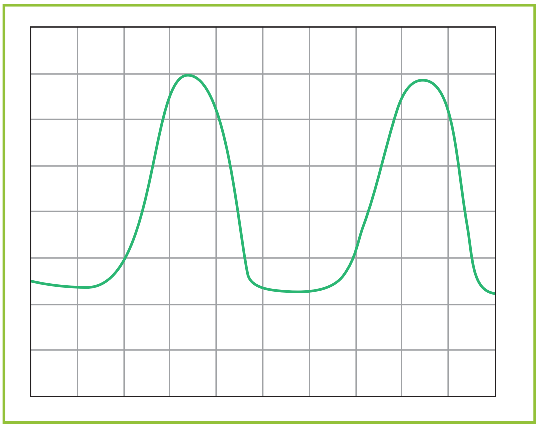


Figure 3: Signal voltages that drive the grid to cutoff cause clipping of the positive half-cycle of output wave.

Figure 4: Grid-current limiting produces clipping of the negative half-cycle of the output wave.



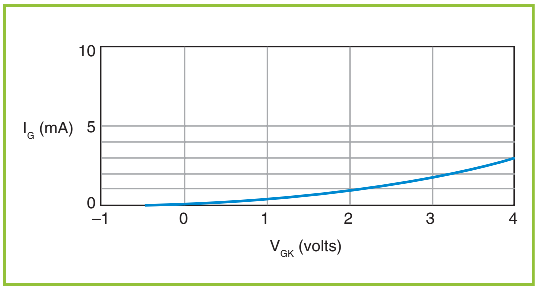


Figure 5: Positive grid voltage in an ECC83 causes grid current.

where, in this case, ΔV_{larger} and $\Delta V_{smaller}$ are the peak voltages of the negative and positive half-cycles of the input voltage at the grid. Since the negative half-cycle is not affected by grid-current limiting, ΔV_{larger} = 0.4 V. Solving for $\Delta V_{smaller}$, the positive peak at maximum distortion will be 0.133 V. This means that when the pickup would be producing 0.4 V into an open circuit, enough grid current will flow to drop 0.267 V (i.e., 0.4 V-0.133 V) across the internal impedance of the guitar pickup. **Figure 5** shows the grid current for an ECC83, plotted against grid voltage. The traces depicted in **Figure 4** were made using a 300-V B+ and 47-kΩ plate resistor.

Figure 6 shows the output impedance Z_O of a pair of Guild humbucking pickups with tone and volume controls and a master volume control, and all controls set at maximum. At 440 Hz (5th-fret "A" on the first string), the impedance is about 7.5 kΩ. The grid current needed to drop 0.267 V across this impedance is:

$$i_G = \frac{V_G}{Z_O} = \frac{0.167\ \text{V}}{7.5\ \text{k}\Omega} = 0.036\ \text{mA}$$

It's difficult to tell from **Figure 5**, but we'll be pretty close to our target if we bias the triode at zero volts. This would give us a quiescent current of about 3.3 mA if we used a 300-V B+ and a 47 kΩ plate resistor. For softer limiting, choose a lower B+ voltage of 100 V. With a plate resistor of 100 kΩ, our load line will be as shown in **Figure 7**.

The inverse of the slope of the "0-V" curve at the Q point is the r_P for the triode at the selected Q point. This works out to:

$$\frac{40\ \text{V}}{0.7\ \text{mA}} = 57{,}142\ \Omega$$

Now we'll calculate the stage gain:

$$A_V = \frac{\mu r_L}{r_P + r_L}$$

where μ = the triode's amplification factor, which is 100 for an ECC83, r_L = the AC load seen at the triode's plate, and r_P = the plate resistance, which is approximately 57 kΩ.

The AC load seen at the plate is the parallel combination of the plate resistor and the input resistance of the following amplifier stage. Using 100 kΩ as the input resistance of the next stage, r_L is 100 kΩ‖100 kΩ = 50 kΩ. The voltage gain is:

$$A_V = \frac{100 \times 50{,}000}{(57{,}000 + 50{,}000)} = 46.73$$

If a harder distortion sound is preferred, we could bias the triode nearer to cutoff. We can use the same load line, meaning B+ and R_P can remain 100 V and 100 kΩ, respectively. We'll just need to change the grid bias so the 0.4-V peak signal will drive the triode to cutoff.

Being precise while using a graphical approach and working this close to cutoff is difficult. **Figure 7** shows the $V_{GK\ (cutoff)}$ is about –1.7 V when the B+ and R_P have the chosen values. Using the earlier calculation and assuming that the negative half-cycle needs to cross the $V_{GK(cutoff)}$ point when the signal voltage has an instantaneous value of –0.133 V, we need to bias the grid at –1.833 V [i.e., –(1.7 V + 0.133 V)]. Setting the grid bias this precisely is a challenge, because

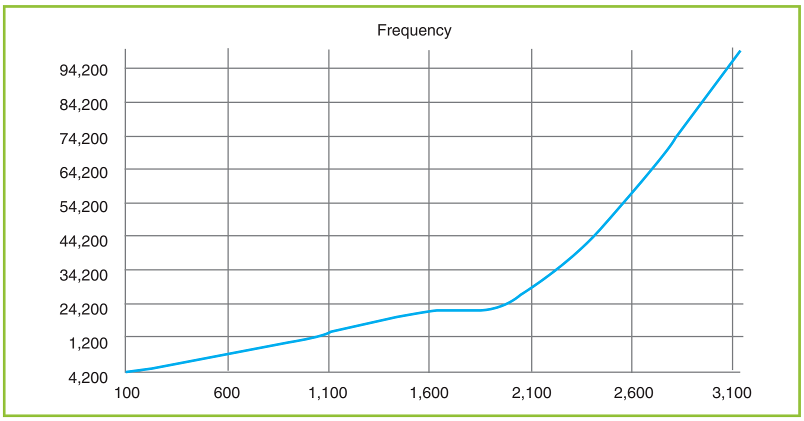


Figure 6: The output impedance of a pair of humbucking pickups varies greatly with frequency.

of variations among even brand-new tubes. Even if we did succeed in biasing the grid, the change in tube characteristics with aging would be a factor affecting the stability of performance: the distortion characteristics of the preamplifier would change over time.

The proper grid bias is important in setting the desired distortion characteristics of an amplifier, and this implies the need for the grid resistor to be fairly small to prevent grid leak effects from changing the bias voltage. A maximum R_G value of 1 MΩ is recommended.

One way to obtain the harder distortion sound of cutoff biasing is to use the first stage mainly for gain, designing it for low-distortion operation, then create the distortion in the second stage. At that point, the signal has much higher amplitude, so small variations in grid bias voltage make less difference in distortion performance.

A versatile tube guitar amplifier could be designed using a grid-current-limiting input stage followed by a second stage biased for cutoff distortion. The guitar's volume controls could be used to keep the input voltage low for a clean sound. With the guitar's volume controls more wide open, we'd get soft grid-current-limit distortion. A volume control between the first and second stages would enable you to add harder cutoff distortion in the second stage. By carefully designing for the proper range of signal level at the second-stage input, you could have a clean sound, soft distortion, or hard distortion all in the same amplifier. An additional volume control and a third gain stage would enable you to drive the phase splitter or output stage into distortion.

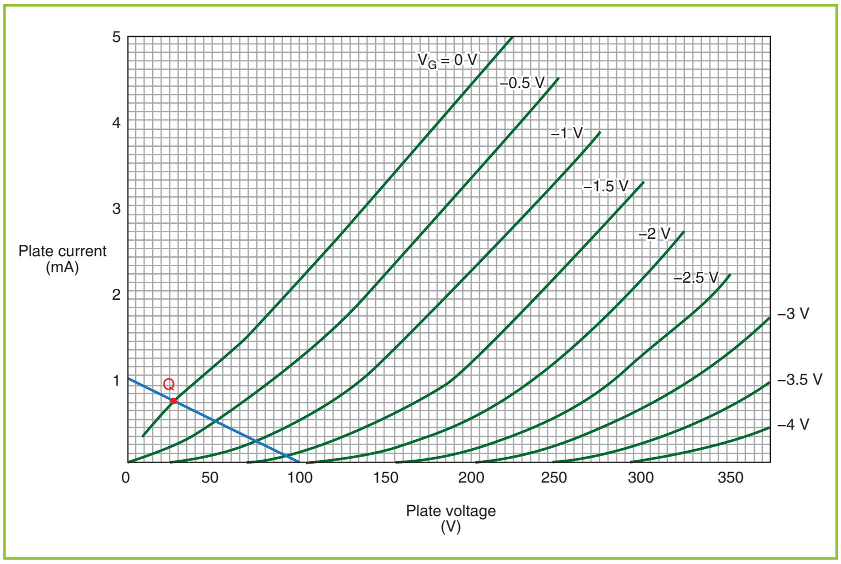


Figure 7: This is the final load line and Q point for our preamplifier designed for grid-current limiting.

Member Profile

Member Name:
Rick Silva

Location:
Studio City, CA

Education:
Musicians Institute, Los Angeles, CA—graduate of the Guitar, Audio Engineering, and Music Video and Television and Film Production programs

Occupation:
Rick is the vice president of production and product manager at Audionamix, and an audio engineering instructor for Musicians Institute (Console Operation and Practical Recording). He is also the music technology author and video artist for Hal Leonard and Alfred Publishing (e.g., *Power Tools for Pro Tools*, *Power Tools for Logic Pro*, and *Choosing the Right DAW for You*). Rick owns Mixed Emotions Productions, which he describes as his "humble audio cave for creativity and experimentation."

Member Status:
Rick subscribes to *audioXpress* because he says the magazine is a great source for reliable information about new and classic, relevant technology. He considers it an excellent "what's happening in all things audio" magazine. "It speaks to audio lovers from beginners, to seasoned professionals and everyone in between," he added.

Audio Organization Affiliations:
Rick is a current full member of the Audio Engineering Society (AES).

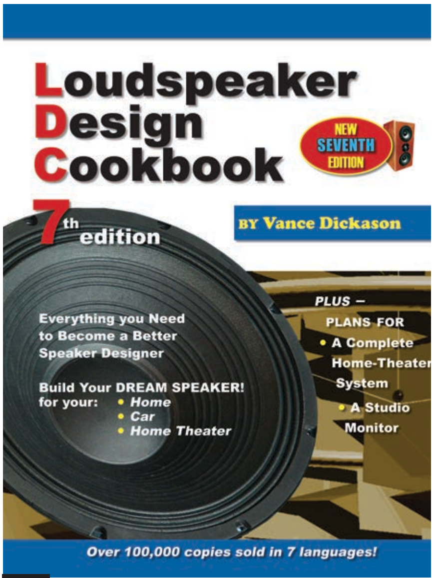
Audio Interests:
Rick plays guitar, writes songs, and enjoys music production. He has also become addicted to all aspects of audio engineering—recording, editing, mixing, mastering, and most recently applying ADX's audio source separation and spectral enhancement technology to the company's technical services projects and consumer-based Separate2Create software, ADX TRAX.

Most Recent Audio Purchase:
Rick listed a 27″ maxed out iMac among his most recent purchases.

Current Audio Projects:
Rick is working on a couple of high-profile, soon-to-be-announced projects for Universal Music Group. He wants to integrate spectral enhancements into ADX TRAX to refine the vocal isolation and instrumental creation software.

Dream System:
For the perfect setup, Rick would include an assortment of classic microphones, a great-sounding live room connected to a vintage Solid State Logic (SSL) or Neve console, Avid Pro Tools|HDX + HD I/O 16×16 interface, Waves SSL Mercury and Studio Classics Collection, ADX Technology, and some extremely talented and creative musicians.

AX Shop



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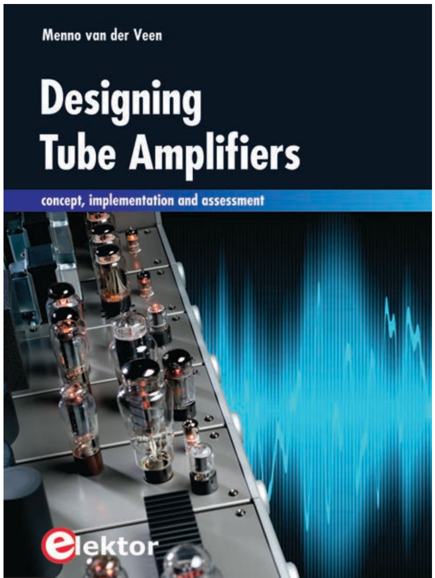

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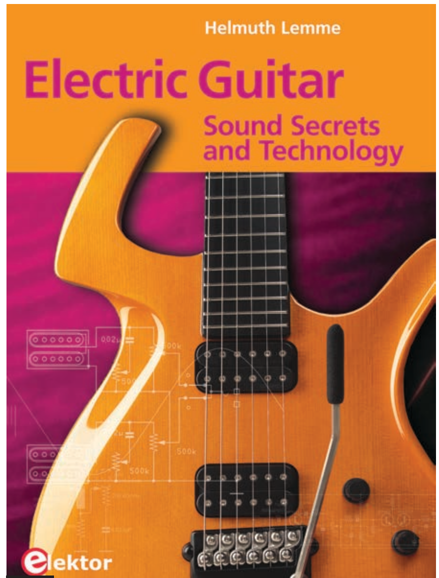

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1 LOUDSPEAKER DESIGN COOKBOOK

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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August 24–27, 2014 — SET Expo 2014

Expo Center Norte, Pavilhão Azul, São Paulo, Brazil
www.setexpo.com.br

In previous years, the Brazilian Television Engineering Society (SET) partnered with the Broadcast & Cable show to host a privately organized event that attracted more than 10,000 attendees. For 2014, the SET has created a new strategic partnership with the National Association of Broadcasters (NAB) and the NAB Show organizers to organize the SET Expo annual conference and trade show. The SET Expo, planned during the traditional SET congress (in August) could become the largest and most significant convention for broadcast and digital media professionals in Latin America.

The NAB Show organizers are developing several educational sessions, hosting an exhibit booth, and providing logistical support to the 2014 SET Expo. The US Consulate in São Paulo is working closely with both organizations as part of its efforts to encourage US-Brazilian business. "We are pleased to expand our relationship with the SET and to collaborate on the SET Expo, an event we see as NAB Show's counterpart in Latin America," says Chris Brown, NAB Executive Vice President of Conventions and Business Operations. "Both organizations are committed to promoting innovation and education in the broadcast and media industries throughout the Americas."

The SET Expo is expected to attract many visitors from Brazil, Latin America, and Africa. With more than 300 exhibitors already confirmed, the SET Expo will host world players involved with everything from creation to distribution across multiple platforms.

September 10–13, 2014 — CEDIA Expo 2014

Colorado Convention Center, Denver, CO
http://expo.cedia.net

The residential integration industry and home technology in general is in a near constant hyperchange and the Custom Electronic Design and Installation Association (CEDIA) Expo intends to address every aspect of the integrated home. For those focused on the residential application areas, the event is a great chance to explore hundreds of new products and meet with manufacturers dedicated to networking, security, energy management, entertainment, home health, and automation solutions. The annual CEDIA Expo is also the perfect opportunity for many professionals to update and/or receive new certifications and advanced training on systems integration and networking. The CEDIA Expo 2013 hosted 17,900 attendees representing 84 countries.

September 11–14, 2014 — NAMM Musikmesse Russia 2014 and Prolight+Sound NAMM 2014

IEC Crocus Expo, Moscow, Russia
www.namm-musikmesse.ru
www.prolight-namm.ru

Both the timing and the new, modern venue are being adjusted to best fit the growing needs of the music product and pro light and sound industries in Russia. The second annual NAMM Musikmesse Russia 2014 and Prolight+Sound NAMM 2014 events held in May 2013 received positive feedback and suggestions for further development from the participants. More than 130 producers and distributors from 15 countries met 9,381 unique visitors, which is a 49% increase compared to the show's premiere in 2012. The programs consisted of more than 90 events including professional development programming, seminars, conferences, and a music festival for the public.

New concepts introduced at the 2013 event—including opening the fairs to the public over the weekend, which gave music lovers the opportunity to take part in numerous master classes and to enjoy the musical festival on the open stage—will be continued in 2014.

Modern exhibition pavilions, conveniently located conference halls, direct metro station access, hotels at the exhibition center, free parking, numerous restaurants and cafés, and options for organizing outdoor festivals and demonstrations are some advantages of the new venue for show exhibitors and visitors.

The NAMM Musikmesse Russia 2014 and Prolight+Sound NAMM 2014 expos will include new elements for participants and will target higher participation of Russian and foreign exhibitors.

September 12–16, 2014 (Conference September 11–15) — IBC 2014

The Amsterdam RAI Exhibition and Congress Centre, Amsterdam, The Netherlands
www.ibc.org

Held every year in Amsterdam, the International Broadcasting Convention (IBC) 2014 conference and trade show is the global meeting place for those engaged in creating, managing, and delivering electronic media and entertainment technology and content. Featuring an influential conference and world-class exhibition, the IBC attracts more than 50,000 professionals from 170 countries to the RAI exhibition center every September. The IBC promotes several "themed areas" including the IBC Connected World to promote wired and wireless Internet delivery and devices, the IBC Workflow Solutions sessions and Workflow Solutions Village and Production Village hands-on exhibition areas, or the IBC Big Screen dedicated to the future of digital cinema. The IBC is also an important audio-oriented event for those interested in broadcast, post-production, and live production technologies. Attendees can expect to find an interesting mix of exhibitors, including all major microphone manufacturers, studio equipment, acoustics, and film and documentary production-oriented solutions.