THE EVOLUTION OF LOUDSPEAKERS, PAGE 7

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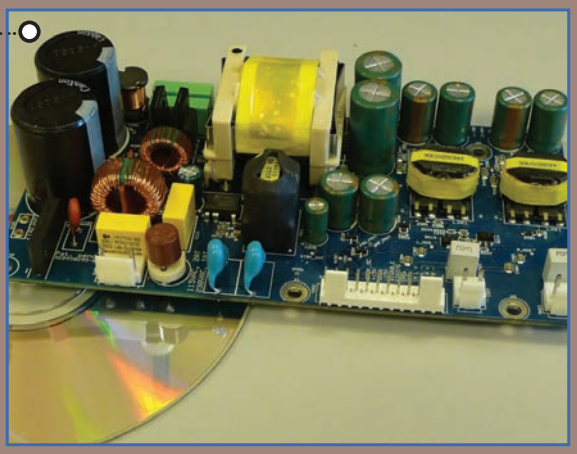

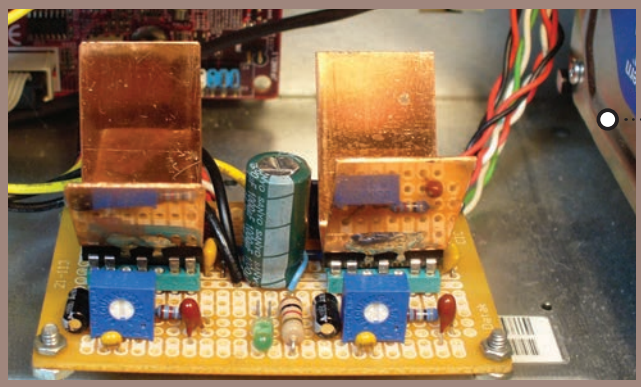

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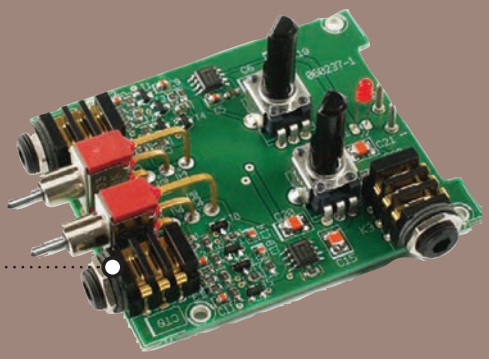

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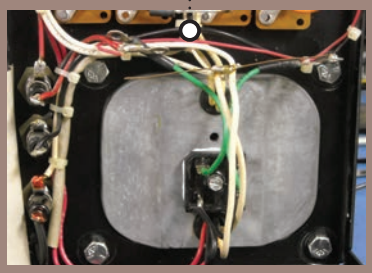

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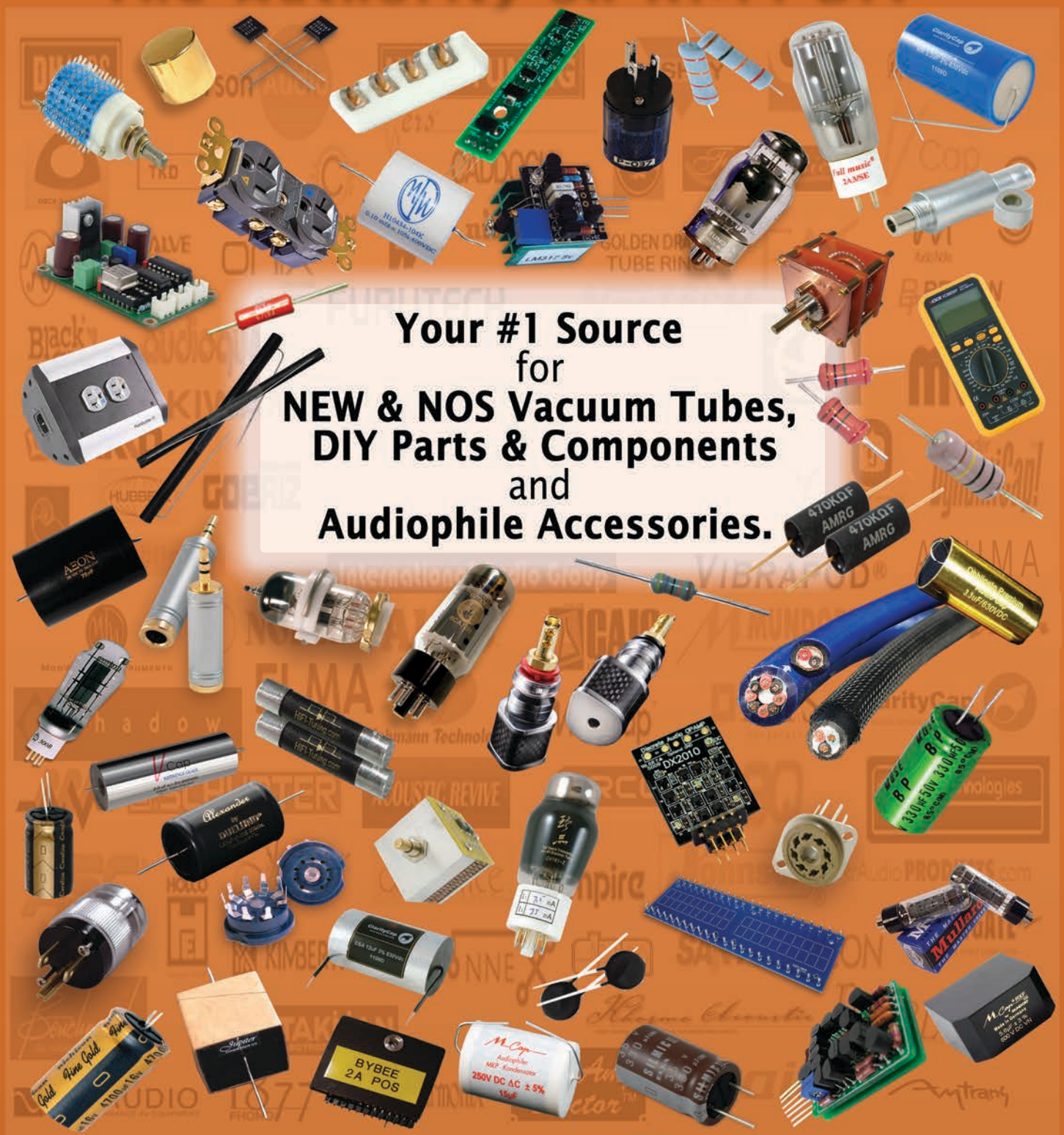

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CONTENTS

VOLUME 42 NUMBER 11 NOVEMBER 2011

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Speakers, Amps, Acoustic Theory, and More

If you were to poll 10 editors about the usefulness of columnists, five would likely say they make periodicals better and five would say they're more trouble than they're worth. Those who oppose using columnists typically make the same arguments: (a) you can alienate readers who disagree with a columnist's point of view and (b) you tend to get bad articles when a columnist is contracted to write an article but has writer's block. Sure, these arguments are logical when you're dealing with certain news magazines. But I think it's prudent to consider the value of columnists on a periodical-by-periodical basis. Columnists make sense for *audioXpress,* which is a detail-oriented magazine covering complex topics such as speaker design, power amps, tubes, and acoustic theory. Here's why.

You simply can't ask a random audio enthusiast to teach readers how to design a tube amp or bench test a silk dome (and provide accurate impedance, waterfall, and surface intensity plots). Skill sets vary from person to person. That's why columnists are important. A columnist for *audioXpress* brings all of his expertise to bear on real-world audio projects and problems, and then makes the information accessible to readers of all levels. For instance, a consummate professional like Vance Dickason ("Test Bench," p. 15) is an industry luminary with years of experience building and testing loudspeakers. With Vance on our team, we can provide go-to information on speaker testing each month. You don't need to wonder if you'll get a dose of speaker analysis next month; you know it is coming.

This month, I'm pleased to announce a new monthly column by Mike Klasco titled, "Parts is Parts." A well-known audio engineer, Mike will endeavor to cover a new audio engineering topic in each issue. Over the course of the next few months, you'll learn about a variety topics such as the evolution of loudspeakers (p. 7), speaker design techniques, test instrumentation, and signal processing.

I trust you'll find Vance's and Mike's columns useful additions to *audioXpress*. If you have other topics you'd like to see covered on a monthly basis, feel free to e-mail me your ideas.

Regards,

C. J. Abate
editor@audioxpress.com

By Mike Klasco (United States)

Loudspeakers Revealed

A Look at the Bits and Pieces of a Loudspeaker

An insider's take on of the evolution of loudspeakers

As you may have read, *audioXpress* has a new publisher who has exciting plans to expand the scope and content of this magazine. Leading the charge is our new editor, C. J. Abate, and one of the additions to the magazine's content will be "Parts is Parts," a monthly column discussing what goes into a loudspeaker—its component parts. Older readers might remember the "Parts is Parts" line from the Wendy's fast food commercials in the mid-1980s. Some of you may also know of my articles on speakers from our sister publication, *Voice Coil*, but the last time I wrote for *audioXpress* was 15 years ago when it was still a separate DIY speaker magazine (*Speaker Builder*).

My business is consulting for speaker companies and materials suppliers in the audio industry, which I've been doing now for over 25 years. Our work ranges from forming techniques for new diaphragm materials; bonding, or otherwise fastening and assembling speaker component parts together; reinforcing additives for speaker cones,and more precise dispensing of coatings, adhesives, and ferrofluids to increase the power handling of voice coils; tightening production and assembly procedures for more consistent products; and similar stuff.

Writing about speaker technology projects is my hobby, and I'm happy to be sharing some of this "wisdom and witchcraft" with you. To introduce this series, let's start with a little loudspeaker history and then take a quick look at the topics for the remainder of 2011.

BRIEF HISTORY OF LOUDSPEAKERS

A transducer is a device that changes one form of energy into another form. A microphone changes acoustical signals to electrical signals. A loudspeaker is also a transducer, which produces an acoustic replica of the electrical signal. Early efforts at transducers for sound reproduction took place in the late 1870s and focused on telephony. Other efforts on sound reproduction soon followed with the development of the phonograph. Phonograph horns (see **Photo 1**) were actually acoustic transformers, better matching the acoustical impedance of the movement of the needle vibrating to the groove modulations into the room—as such, the horn itself is not a transducer. Today, horns are typically coupled to compression drivers, which is the transducer in this type of system.

Photo 1: Little Nipper checking out an acoustic transformer (the horn), but his speaker engineering background is in woofers. [Source: Wikipedia; Artist: Francis Barraud (1856-1924)]

Moving coil loudspeakers were conceived around 1874 when Ernest W. Seimens' s United States patent first described the moving coil loudspeaker. Later that decade, Bell was granted a patent for the telephone (the first patent describing audible transmission) and Seimens was granted the first patents for the acoustical horn. The early part of the 20th century saw continued work on loudspeakers and sound reproduction including John Stroh's British patent describing the cone and surround (1901), Anton Pollak's U.S. patent for the spider (1909), Bell Lab's initiative to improve sound recording and reproduction (1915), and the "Phonetron" (see **Photo 2**) is the first direct radiator loudspeaker sold in the United States (1923).

However, it was the work of Rice and Kellogg in the mid-1920s which was really the start of the loudspeaker as we now know it. Along with the growing demand for the type of entertainment provided by radios and "talking pictures," the commercial development of the loudspeaker and related components was one of the driving forces behind the formative years of our consumer electronics industry.

Up until World War II, speakers borrowed power from the amplifier power supply to energize their electromagnet, but during the war, practical and cost-effective permanent magnets were developed and became common place.

Many soldiers came home from the war (WWII) with radio training and

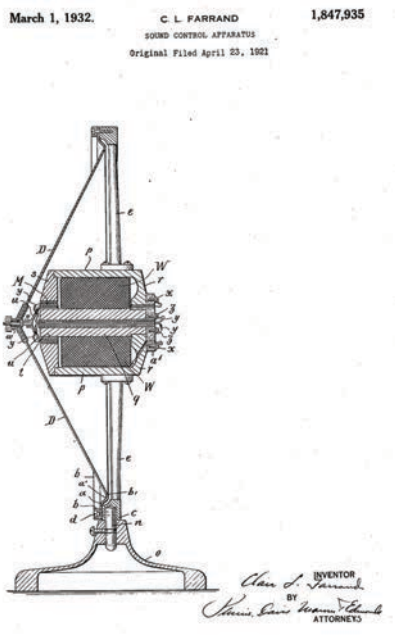

Photo 2: The "Phonetron" (Image courtesy of Professor Steven Schoenherrr)

became electronics hobbyists. Some veterans even started their own electrical appliance repair or manufacturing businesses. Early audiophile gear was tube-based and the speaker systems were huge. By the late 1940s, companies such as Klipsch and Altec-Lansing were manufacturing large horn-loaded systems. The acoustic suspension woofer was launched in the early 1950s and greatly impacted popular acceptance of hi-fi and stereo. The early AR products were quite low in efficiency and the tube amplifiers of the day were really pressed to get them to play to their full capability. The wife-acceptance factor of speakers improved in the 1960s with the introduction of "stereophonic sound," which forced the downsizing of speaker enclosures.

By the early 1970s, amplifier power was increasing thanks to transistor amplifiers and stereo receivers. This further enabled manufacturers to downsize their speaker systems. More efficient acoustic suspension designs (but at the expense of deep bass) came from KLH followed by many others. Before the end of the 1970s, high-quality audio had gone mainstream. The speaker industry in the United States was flourishing, with the West coast sound from JBL and Altec Lansing, followed by Cerwin Vega, and the East Coast contingent from AR and KLH soon to be followed by Advent, ADS, Polk and Boston Acoustics. Back then, all of these outfits made their own drivers (woofers, tweeters, and midranges), rather than sourcing from Asia.

Tweeters were the first to be outsourced and came first from Japan in the mid-1970s. In the 1980s, all types of speakers started to be outsourced to Taiwan and Korea, later joined by Malaysia, and eventually the black hole of manufacturing, China, sucked up most of the world's speaker production. Presently, most speakers are made in China, with fragmented production in Vietnam, Indonesia, Philippines, Malaysia, as well as higher-end specialists in Europe, and elsewhere.

Today, billions of speakers are produced annually, and they surround us in our everyday lives—from cell phones, earphones, intercoms, telephones, laptops, docking stations, TVs, radios, automobiles, public address, and commercial sound applications, home theater, and whole house audio, etc.

Yet even when audio hobbyist magazines focus on speakers, the articles are most often on system design rather than the actual nitty gritty of speaker-driver construction. Sadly, it seems the only place where speaker-driver engineering is explored is in the advertisements, not in the magazine articles. In this continuing series, our focus will be on the function of each speaker part, how it is made, what it's made from, and what defines the good, the bad, and the ugly.

For starters, we will focus on the cone speakers, leaving dome tweeters, compression drivers, coaxes, and more esoteric transducers like ribbon and electrostatic planars, piezo, and air motion drivers for future discussions.

Here are the topics we have lined up for upcoming installments of "Parts is Parts."

THE CONE AND DUST CAP

The cone is the most critical component of the speaker and has a significant contribution to its sound quality (see **Photo 3**). When mounted in an enclosure, it is the only visible component to the end-user. The design engineer spends much of his precious bill of materials budget on a flat wire coil—and no one will know—but spend a few extra bucks on a woven Kevlar cone and no one will miss the yellow high-tech look (see **Photo 4**). Most speaker cones are fabricated from a recipe of wood (cellulose) fibers in a paper-making process, but polypropylene, woven glass, carbon fiber, Kevlar composites, and metal cones are also popular choices. Poly cones may be thermoformed (like melted cheese) or injection molded. Metal cones can use aluminun, magnesium-aluminum, or even beryllium. Metal cones may be formed by punching or spinning or even vapor deposition. Woven cones are typically formed in a hydraulic press.

COMPLIANCE–THE SURROUND AND SPIDER

The speaker cone reproduces sound through back and forth movement in a mechanical analogy to the electrical signal. The compliance(s) guide the cone, keep the voice coil centered in the magnetic gap, provide restoring force, and limit excessive excursion. The various types of compliances will be discussed, from shape, materials, and fabrication. Surrounds can be half roll, M and W profiles, along with a number of other unusual topologies. Materials range from the basic woven cotton and foam, to Nomex and Conex aramids and thermoplastic elastomers such as Santoprene or butyl rubber. Spider fabrics

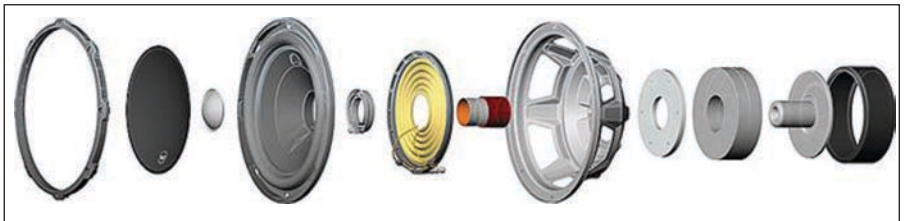

Photo 3: An exploded image of a high-quality loudspeaker (Image courtesy of JL Audio)

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typically include cotton, poly-cotton, and aramids.

VOICE COIL ASSEMBLY—BOBBIN, VOICE COIL WIRE/WINDING, COLLAR, LEAD-OUT WIRES

The voice coil is the heart and motor of the speaker. Essentially a coil of wire wound on a form (bobbin) with the beginning and ends of the coil leading out to the speaker terminals. Wire gauge, voice coil winding length, type of the wire conductors, including materials and shape, fabrication techniques, and many other parameters will be explored. Voice coil bobbin materials include polyimides (such as Kapton), aluminum and other metal foils, and woven fabrics with resin coatings. Voice coil wire (see **Photo 5**) can have a round, square, a rectangular cross-section and the conductor can be copper, aluminum, or copper-clad aluminum. Even the "dress" of how the voice coil wire lead-out are run and the flex wire-woven composition is a significant consideration. How all these factors influence efficiency, power handling, distortion, and cost will be discussed.

HARD PARTS—MAGNETIC STRUCTURE AND FRAME

The speaker components can be separated into the hard parts and the soft parts. (It's the soft parts that move.) Hard parts include the magnet, the magnet return structure, and the frame to which all the speaker parts are assembled. Frames can be stamped steel, cast aluminum, or plastic. Each of these can have an impact on distortion, power handling, and other factors. Recent pricing trends are forcing a move away from the compact high-energy Neodymium magnets back to ferrite.

The physical topologies of the various magnetic structures have an impact on sound quality, including venting of air pressure. The magnetic structure design also has an impact on sound quality, from design issues such as saturation of the top plate, aluminun bobbin eddy currents creating intermodulation distortion, and remedial techniques such as copper cap on the pole piece and shorting rings, etc.

A side bar discussion will cover both recent and past politics on speaker magnetic structure design.

ASSEMBLY—INCLUDING A LOOK AT PRODUCTION LINES, ADHESIVES, COATINGS, AND TREATMENTS

The adhesive joints throughout the speaker have an impact on the sound from the speaker, as well as power handling and product consistency. Coatings can enable weather resistance and moisture stability, to smoother response. We will also take a look at ferrofluids and their application to speakers for lower distortion, increased power handling, and less field failures.

Additionally there will be some exploration of assembly jigs and fixtures, and factors that contribute to product consistency. Finally, a look at speaker production lines, from manual, "islands of automation," semi-automatic, and full "lights-out" automation (see **Photo 6**).

Next month, "Parts is Parts" will explore the world of speaker cones. *aX*

Photo 4: B&W woven Kevlar cone (Photo courtesy of Dai-Ichi Electronics, the Phillipines)

Photo 5: High-performance, flat wire coils from Precision Econowind (Source: Precision Econowind)

Photo 6: Speakers on a semi-automatic production line in the Phillipines. (Photo courtesy of Dai-Ichi Electronics)

By Carl Casper (United States)

The Casper PC Audio Amplifier

Here's an easy-to-build amplifier for your computer speakers

As computers have evolved, their functionality has increased. Faster processors, more storage and more memory are changes that have benefited us all. However, at some point, the soundcard (or mainboard sound section) lost its ability to drive small speakers. Many of us have found that the pair of speakers we had carefully chosen for this purpose is now collecting dust after one of those upgrades. There are numerous types of amplified speakers you can buy at various prices, but many of them are made of plastic and may not have the sound quality you would like.

If you are like me, and still want to use those unamplified speakers, here is a low-power stereo amplifier that will drive them (see **Photo 1**).

It fits into some of the vacant space within your desktop enclosure and uses a small amount of power from the computer's 12-V output power supply. It taps off of the sound card "Line Output" and loudness varies using the computer's taskbar volume icon.

CIRCUITRY

The National Semiconductor LM380 (or the newer LM384) was selected for the purpose. It is a single-supply audio amplifier; inexpensive and is easy to work with. There are other integrated circuits out there with more impressive specs, but many of them are only available in surface-mount-type packages. I chose a chip that a person with average construction skills could use successfully (see **Figure 1**).

The LM384 has a higher supply voltage rating but we won't be taking advantage of it here (see **Figure 2**).

With only +12 VDC to power the amplifier, you won't be able to rattle the china in the cabinet, but the sound reproduction seems quite good. The integrated circuit has an internally fixed gain of 34 db, which is divided down by an adjustable voltage divider using pot R1. The non-inverting input is used and a 1000-pF capacitor is used across the amplifier input to attenuate the high-frequency response slightly. It gives a –3-db point around 15 kHz. (You may want to reduce this value if more high-frequency response is needed) The inverting input is grounded, giving better output symmetry and less possibility of noise pickup.

The output uses a Zobel network (2.7 Ω and 0.1 μF). There is a 560 Ω resistor in place across the output that acts as a light load to keep the output capacitor charged if no speaker is connected. A resettable fuse (more commonly known as a Polyswitch) is used on the 12-VDC circuit to safely limit the current in the event there is a malfunction. The green LED indicates the 12 VDC is on.

CONSTRUCTION

This project is fairly straightforward to construct on a general-purpose construction board. The (Radio Shack 276–150) lends itself nicely to this project (see **Photo 2a** and **2 b**). The two lengthwise traces are strapped together at various places to form a sturdy ground bus.

The integrated circuits could be soldered directly to the board, but I prefer sockets. Turret-type sockets are preferable for good integrated circuit (IC) retention (five of the 14 IC pins will not be in the socket) Place and solder the sockets and solder grounds to pins 6, 7, and 11 using #22 AWG bare wire. Install the components next, starting with resistors, then capacitors, and remaining parts.

A PC floppy adapter cable (with the floppy connector cutoff) is used to bring +12 VDC to the amplifier. The yellow wire to +12 and two black wires to the circuit board ground. The red wire is unused, so cut it short and insulate the end.

A length of two-wire shielded cable connects to the inputs of each amplifier input circuit and ground; it should be long enough to reach the line level output jack on the rear of the computer or the line source points on the mainboard

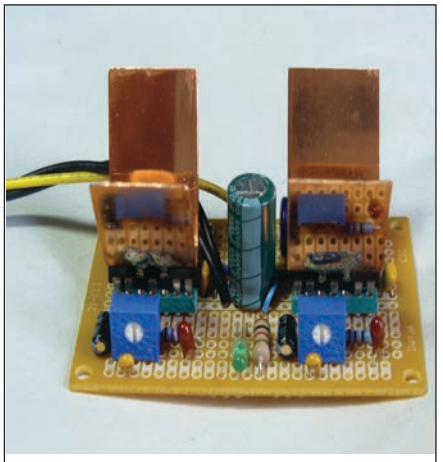

Photo 1: Completed stereo amplifier card

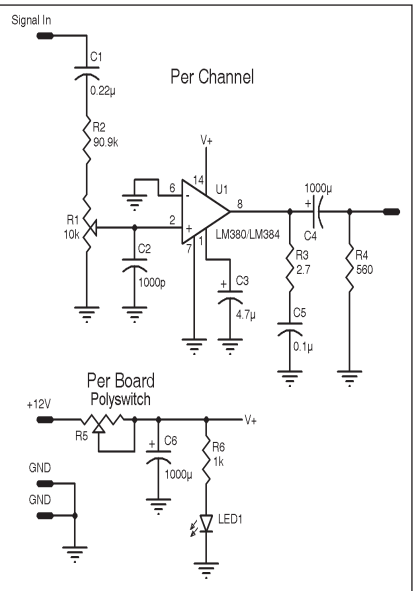

Figure 1: A schematic diagram showing the amplifier circuitry

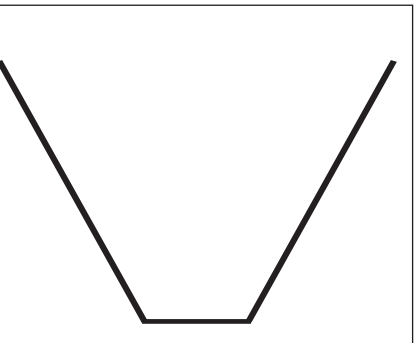

Figure 2: Bending guide

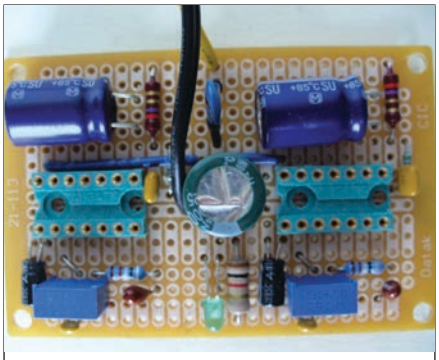

Photo 2a: Top view of the construction board

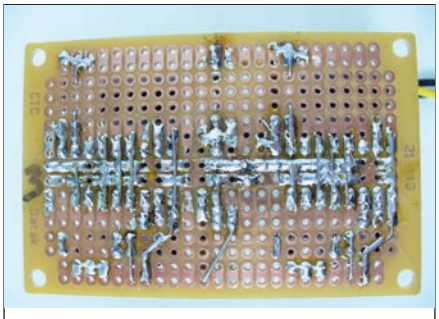

Photo 2b: Bottom view of the construction board

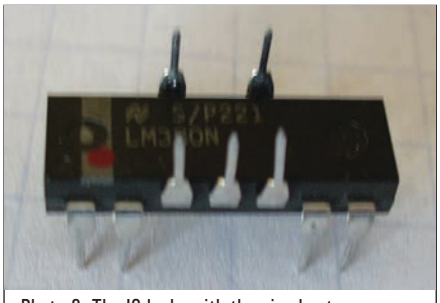

Photo 3: The IC body with the pins bent

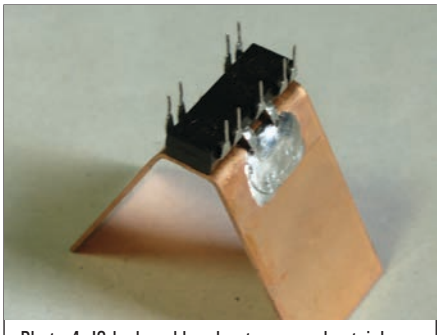

Photo 4: IC body soldered onto copper heatsink

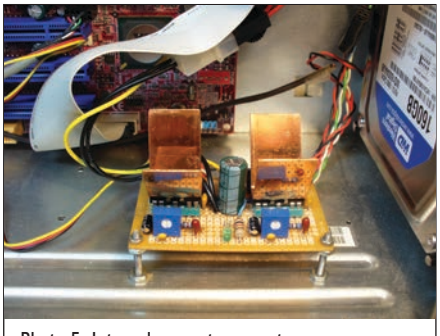

Photo 5: Internal computer mount

or soundcard.

IC CHIP PREP

The LM 380 has to have Pins 3, 4, 5, 10, and 12 bent outward and upward to contact the external heatsink. This must be done slowly and gingerly by grasping one pin at a time with needle nose pliers approximately 1/16" away from the IC body (see **Photo 3**). Too much force and repeated movement may cause the lead to break from metal fatigue. If you break off more than one lead, it would be best to use another chip.

HEATSINKS

Cut 0.75" × 2.5" strips of .023" copper. (Try your local hobby store). Bend using **Figure 2** as a guide. The contact area on the heatsink that the IC pins will touch should be clean and shiny such that soldering will go quickly. If not, polish with fine emery paper. Use Tin-lead solder if possible, its melting point is lower than non-lead solders. Position the IC squarely on the heatsink, making sure that the five pins are touching it (see **Photo 4**). You can secure it prior to soldering, with a drop of five-minute epoxy. Solder with a medium-high heat. The copper will suck up the heat and you want to get the job done quickly. Place the soldered assembly on top of a wet sponge to aid cooling.

BENCH TESTING

After assembly, I recommend bench testing this project before connecting it to your computer: An accidental short circuit to a computer power supply may damage it.

With no integrated circuits in the board, connect a 12 V, 1-A-rated DC power supply. The green LED should light. Turn off the power. Insert one LM380 and heatsink assembly. Turn on power. Measure pin 9 on the IC with a voltmeter, taking care not to short it. It should be close to one-half the supply voltage. If not, turn off the power and check for solder shorts on the board near the IC. If it is okay, repeat the above instructions for the second LM380.

After several minutes of operation (with no input or output) touch the heatsinks with your finger. There should be no more than slight heat. If either IC is hot, stop and check for shorts or connection errors.

Having made it to this point, attach an 8-Ω, 10-W resistor (or test speakers) to each output. Connect a sine-wave generator, set to 1000 Hz, to amplifier inputs. Set gain trimmer pot on each amplifier full clockwise. Monitor first one output, then the other with an oscilloscope. While increasing the generator output, note that the waveform is symmetrical and undistorted up to about 8 to 9 V_{PP} where it should start clipping more or less symmetrically.

If you don't have a signal generator and oscilloscope available, with the test speaker connected, touch your finger to the open input and listen for a 60-Hz hum. (Living 1,500' away from a 50 kW AM radio station, I hear music, too!)

MOUNTING/INSTALLATION

You have a choice of two mounting schemes. With the first method, the amplifier is built into a PC using holes drilled from the outside (bottom) of the chassis using 4-40 screws set up as standoffs (see **Photo 5**). This method works well when the chassis is empty. You must be extremely scrupulous when drilling into a chassis that is complete because any metal scraps could cause havoc with the motherboard or disc drives. Masking tape on the inside of the panel can be used behind the points at which the drill bit will break through. The open side of the chassis can also be oriented downward when drilling and the inside of the chassis vacuumed when done. The 1/8" speaker jack is mounted in a hole drilled in a blank expansion-board panel which can be fitted into the chassis when done.

The second method involves attaching the amplifier board as well as the speaker jack to the blank expansion-board panel using small L brackets (see **Photo 6**). Lay out and drill the holes for each L bracket in the expansion board panel to match the corner holes of the L brackets attached to the circuit board. Now drill two holes in the expansion panel opposite each trimpot to facilitate later adjustments. This assembly can be mounted inside the computer chassis where there is an unpopulated slot.

AUDIO INPUT

Consider an external plug-in. The simplest arrangement brings the shielded cable through a hole in the rear of the

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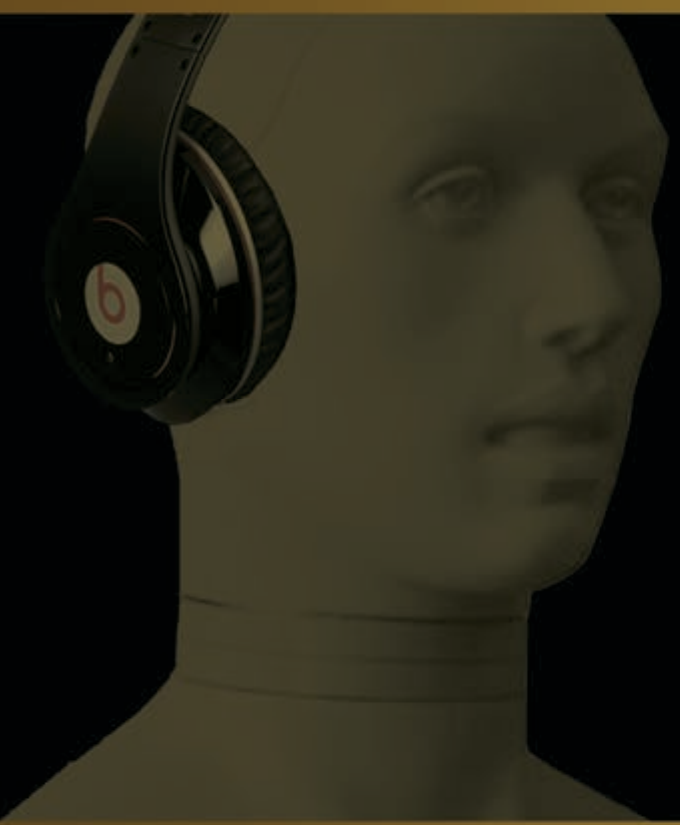

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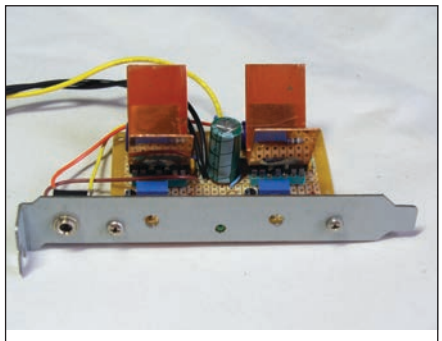

Photo 6: Panel mount

Photo 7: Input cable

chassis (or a blank expansion board panel with a 0.25" grommeted hole) and has a 1/8" stereo plug to plug into the line output jack (see **Photo 7**).

Now, consider an internal connection. Attaching the amplifier input wires to the line output inside the box results in a neater installation and the line output jack will be available to plug other gear into at the same time. (The amplifier board has an input resistance of 100 kΩ which presents a negligible load.)

This is a more demanding installation. If you are not familiar with the innards of a computer, it may be best to consult with someone who has more computer experience. The two common arrangements are as follows:

For a plug-in sound card, locate the line output jack. Unplug the card (with power unplugged) and solder the wires from the input cable to the rear of the respective jack. Solder the red and black wires to the right and left terminals. Solder the ground wire to nearby ground trace. Ground wire from the shielded cable may need a short length of insulated tubing to make it short-proof. Reinsert the card.

If the computer doesn't have a plug-in sound card, the audio is integrated into the main board and you have to get at the solder side of the line output jack. The connection procedure is the same.

The main board will have to be systematically unplugged, unscrewed, and lifted out to get at the solder side of the line output jack. Re-assemble.

FINAL ADJUSTMENT

With amplifier and speakers connected, try various sound sources. Adjust the two trimpots for satisfactory loudness when the speaker icon on the screen is set to maximum.

The output power will be limited by the supply voltage applied. Since we are using the +12-VDC supply from the computer, the largest output swing that is available is 9 to 10 V_{PP}, which translates to 3.2 W across a 4-Ω load per channel. *aX*

AMPLIFIER PARTS LIST

Per Channel (duplicate for stereo)

- 1 - 1000 µF @ 10 V radial electrolytic capacitor
- 1 - 4.7 µF @ 25 V radial electrolytic
- 1 - 0.22 µF @ 50 V film capacitor
- 1 - 0.1 µF @ 50 V ceramic capacitor
- 1 -1000 pF @ 50 V ceramic
- 1 - 90.9 kΩ 0.25 W 1% resistor
- 1 - 560 Ω 0.25 W 5% resistor
- 1 - 2.7 Ω 0.50 W 5% resistor
- 1 - 1 turn, 10 kΩ trimpot: Bourns 3386-w-1-103LF
- 1 - 14-pin turret IC socket
- 1 - LM380 (or LM384) integrated circuit
- 1 - heatsink (0.075" × 2/5" piece of copper)

Per Board

- 1 -1000 µF @ 25 V radial electrolytic capacitor
- 1 -1000 Ω 0.50 W resistor
- 1 - 0.1 µF @ 50 V ceramic capacitor
- 1 - Green 3 mm LED
- 1 - PTC resettable fuse; Polyswitch # RXEF090
- 1 - Grid-Style PC board: Radio Shack # 276-150
- 1 – 3' length 2-wire, 22-gauge shielded cable

Per Amplifier

- 1 - 1/8" stereo jack
- 1 - PC chassis slot fill panel
- 1 - adapter cable: power supply to floppy; MPJA 7206 CB
- 2 - right angle brackets; 4-40 threaded; Keystone Mouser PN # 534-621

Miscellaneous

- 440 screws, lockwashers, and nuts

Scan-Speak Discover Drivers

A test of two high-end home audio drivers

This month Scan-Speak sent a couple of high-end home audio drivers from their relatively new Discovery lineup: a 1" wide surround silk dome tweeter, the D2604/833000, and a 10" aluminum cone subwoofer, the 26W/4558T00.

The D2604/833000 Scan-Speak has been back on its own for a couple of years now and doing well as an independent business entity now that it is no longer part of DST or Tymphany. Founded in 1970, Scan-Speak is still working from the same address in Videbaek, Denmark, and with the same "no compromise" philosophy that was always part of the company's mission.

One of the latest efforts for Scan-Speak was to develop a new high-quality line of more competitively priced OEM products, dubbed Discovery. The D2604/833000 is the latest tweeter offering from the Discovery line and features a large surround 1"-diameter silk dome, a low 475-Hz resonance, CRT shielding, a separate rear chamber, a ferrite magnet motor, an injection-molded faceplate with a five-point mounting configuration, voice coil wound with copper-clad aluminum wire, and gold-plated terminals.

I began testing the D2604/833000 silk dome (see **Photo 1**) by generating a stepped sine wave impedance plot using the LinearX LMS analyzer. The result of the LMS 300-point impedance sine wave sweep is given in **Figure 1**. The tweeter resonance is 426 Hz, somewhat lower than the factory specification. Minimum impedance for this tweeter is 3.2 Ω at 2.4 kHz with a measured Re = 3.0 Ω.

After completing the impedance measurements, I recess mounted the Scan-Speak tweeter in a small enclosure that had a baffle area of about 12" × 7" and measured the on- and off-axis frequency response at 2.83 V/1 m. **Figure 2** depicts the on-axis response. Frequency response for the D2604 dome is a very flat ±1.75 from 890 Hz to 16 kHz with the response extending to 25 kHz. **Figure 3** gives the on- and off-axis response for the Scan-Speak wide surround silk dome tweeter. Off-axis the device is -4.5 dB down at 10 kHz from the on-axis response with respect to the 30° off-axis curve and -7.2 dB at 45° off-axis, again with respect to the on-axis response. **Figure 4** illustrates the normalized version of **Figure 3**. In terms of production consistency, the two-sample SPL comparison is depicted in **Figure 5,** indicating the two samples were well matched with some minor variation in the 3 to 7.5 kHz region.

Next, I used the Listen, Inc. SoundCheck analyzer and SCM microphone to measure the impulse response with the tweeter recess-mounted. Importing this data into the Listen, Inc. SoundMap software produced the cumulative spectral decay plot (waterfall) shown in **Figure 6**. While there are no major resonances indicated in this plot, it is very difficult to correlate long decay resonances with subjective performance.

Figure 7 gives the Short Time Fourier Transform (STFT) displayed as a surface plot. Last, I set the 1 m SPL to 94 dB (5.8 V),

Photo 1: The D2604833000 silk dome

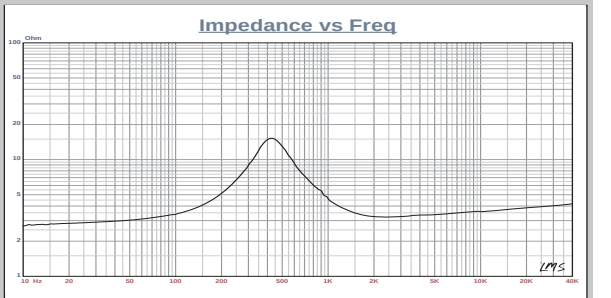

Figure 1: Scan-Speak D2604/833000 free-air impedance plot

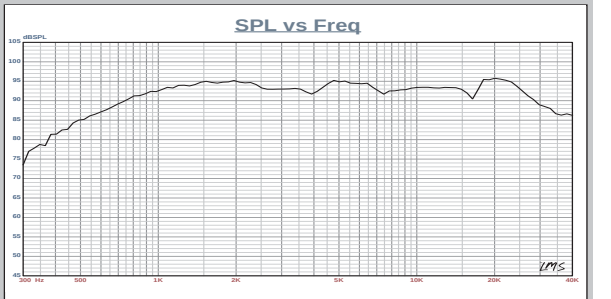

Figure 2: Scan-Speak D2604/833000 on-axis response

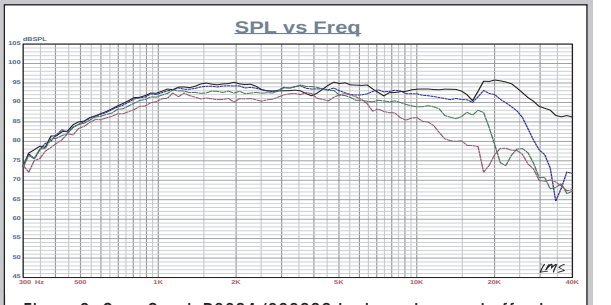

Figure 3: Scan-Speak D2604/833000 horizontal on- and off-axis frequency response (A = 0°; B = 15°; C = 30°; D = 45°)

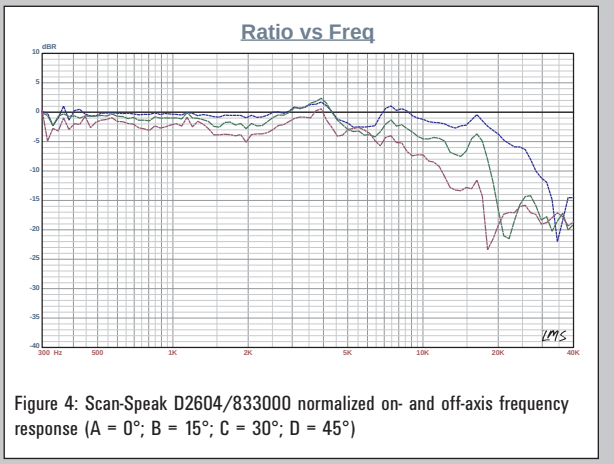

Figure 4: Scan-Speak D2604/833000 normalized on- and off-axis frequency response (A = 0°; B = 15°; C = 30°; D = 45°)

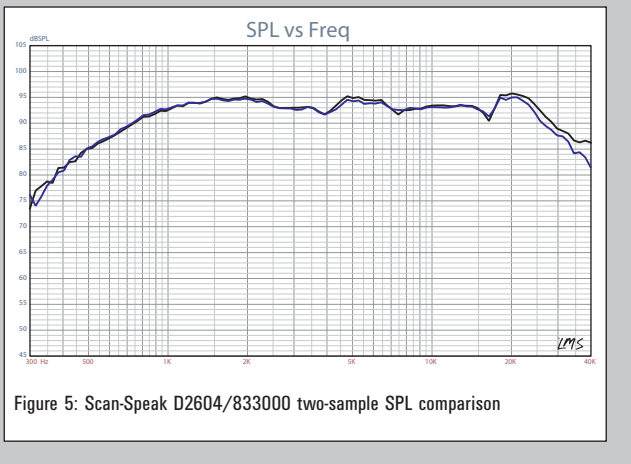

Figure 5: Scan-Speak D2604/833000 two-sample SPL comparison

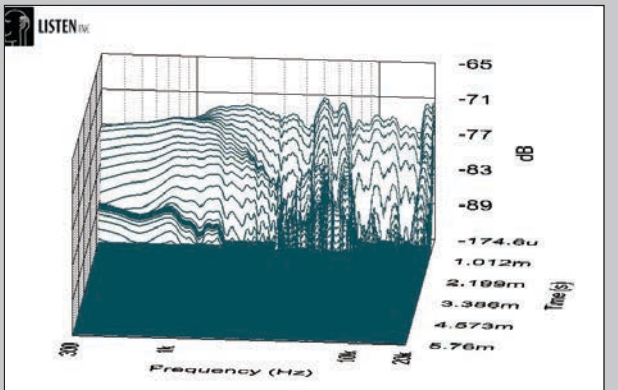

Figure 6: Scan-Speak D2604/833000 SoundCheck CSD waterfall plot

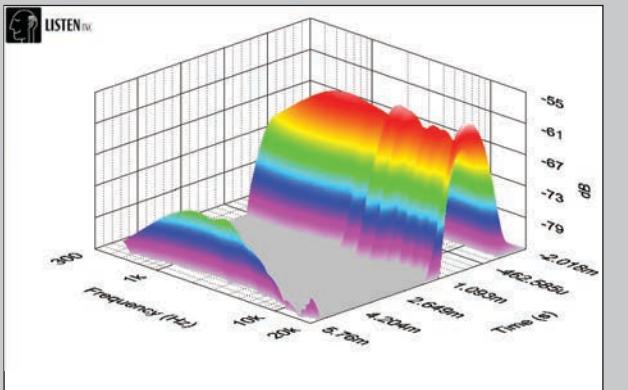

Figure 7: Scan-Speak D2604/833000 SoundCheck STFT surface intensity plot

and the sweep range to 2 kHz to 20 kHz and measured the 2nd and 3rd harmonic distortion at 10 cm, depicted in **Figure 8**. This shows the relationship between the 2nd and 3rd harmonic distortion; however, correlation to subjective preference based on THD is not well established.

For those of you who have not used this format tweeter in a system, it's a great design and one of my favorites. All of the various incarnations I have designed into systems have sounded very good, and given the quality and craftsmanship, always exhibited by Scan-Speak, this is a great addition to their new Discovery line.

The new Scan-Speak 26W/4558T00 subwoofer (see **Photo 2**) is built on an eight-spoke proprietary cast aluminum frame, which includes eight 35 mm × 20 mm "windows" below the spider mounting shelf for cooling. Scan-Speak obviously spent a lot of time designing the cone for this woofer. The material

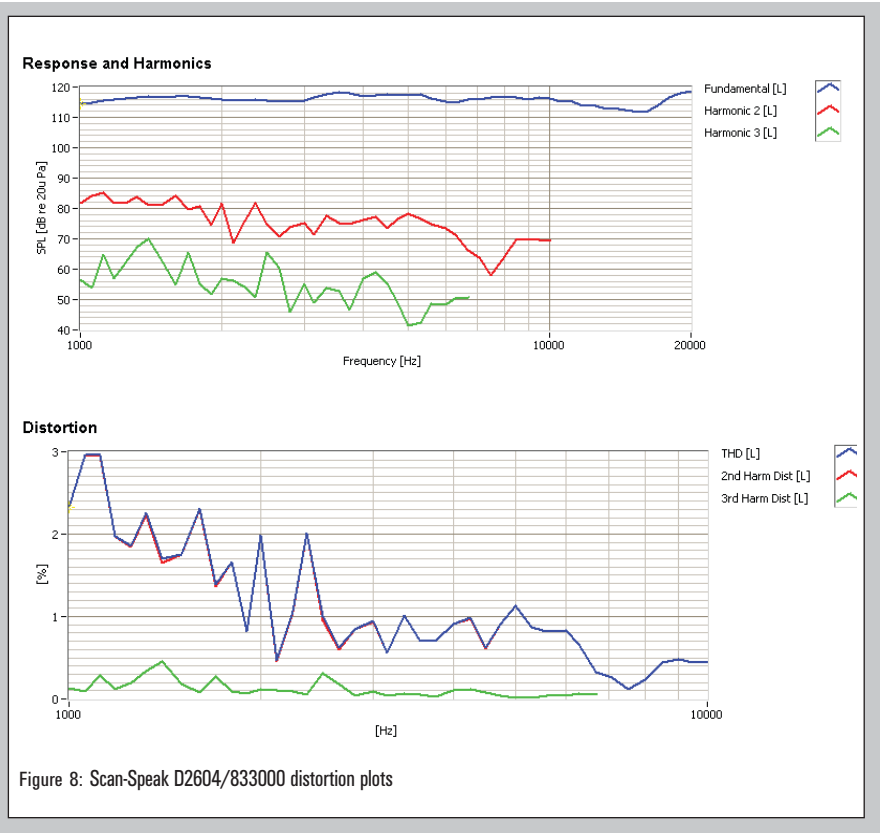

Figure 8: Scan-Speak D2604/833000 distortion plots

Photo 2: The Scan-Speak 26W/4558T00 subwoofer

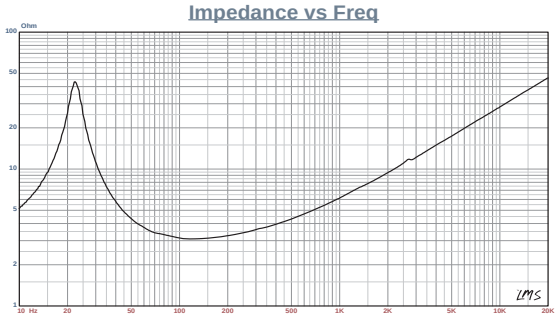

Figure 9: Scan-Speak 26W/4558T00 free-air impedance plot

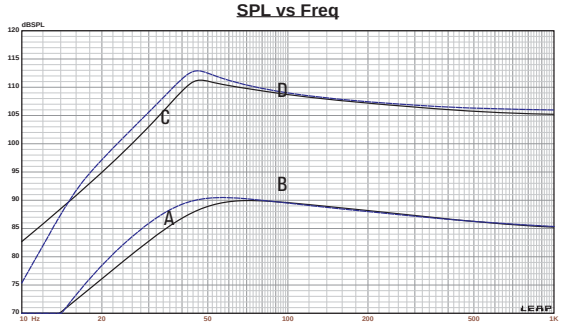

Figure 10: Scan-Speak 26W/4558T00 computer box simulations (A = sealed at 2.83 V; B = vented at 2.83 V; C = sealed at 36 V; D = vented at 40 V)

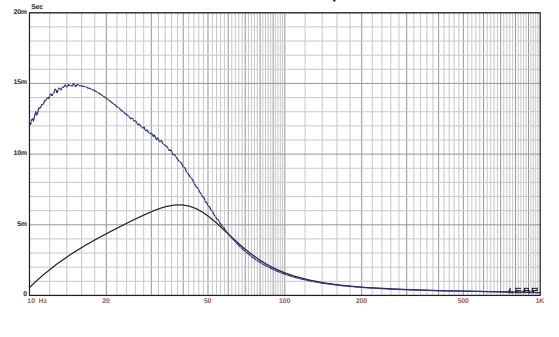

Figure 11: Group delay curves for the 2.83 V curves in Figure 10

used for this cone is a thick, black flat profile anodized aluminum that has the outside perimeter of the cone turned down to strengthen the edge to prevent flexing. However, even more interesting is that the cone has a 5" articulation that forms a 1/16" rib that is in just about the same place that the dust cap attaches on the front side of the cone, which should add considerable stiffness. There are also eight slightly oblong 0.25 × 3/8" holes intended to vent air out of the area beneath the dust cap. The dust cap is a 5" diameter convex-type made from a black-coated fiberglass/paper sandwich.

Providing compliance for this woofer is a fairly wide (30 mm) black NBR surround and a black 6" diameter flat Nomex spider. It also includes flat Litz wire sewn into the spider that terminates to a pair of gold-plated terminals.

While not connected to a terminal (this is a single voice coil subwoofer), there is another pair of Litz wires sewn into the opposite side of the spider.

The motor system for the 26W is composed of a stacked pair of 147-mm diameter and 18-mm thick ferrite magnets. These are sandwiched between an 8-mm front plate and a 9-mm back plate with a 10-mm bump out.

I commenced testing the Scan-Speak 10" 26W/4558T00 subwoofer using the LinearX LMS analyzer and VIBox to produce both voltage and admittance (current) curves with the driver clamped to a rigid test fixture in free-air at 1 V, 3 V, 6 V, 10 V, 15 V, and 20 V. The 12 550-point stepped sine wave sweeps for each 26W sample were post-processed and the voltage curves divided by the current curves (admittance) to produce impedance curves, phase added using LMS calculation method, and along with the accompanying voltage curves, uploaded to the LEAP 5 Enclosure Shop software.

Besides the LEAP 5 LTD model results, I additionally produced a LEAP 4 TSL model set of parameters using just the 1-V free-air curves. The final data, which includes the multiple voltage impedance curves for the LTD model (see **Figure 9** for the 1-V free-air impedance curve) and the 1-V impedance curve for the TSL model, were selected and the parameters created in order to perform the computer box simulations. **Table 1** compares the LEAP 5 LTD and TSL data and factory parameters for both 26W samples.

LEAP parameter calculation results for the 26W were reasonably close to the factory data, although my data showed a lower sensitivity. Following my usual protocol, I set up computer enclosure simulations using the LEAP LTD parameters for

Table 1: Scan-Speak 26W/4558T00 subwoofer

	TSL model		LTD model		Factory
	sample 1	sample 2	sample 1	sample 2	
F_S	22.1 Hz	21.9 Hz	21.2 Hz	21.1 Hz	20 Hz
R_{EVC}	2.51	2.51	2.51	2.51	2.5
Sd	0.0340	0.0340	0.0340	0.0340	0.0352
Q_{MS}	6.17	5.99	5.95	6.13	5.59
Q_{ES}	0.38	0.38	0.35	0.38	0.29
Q_{TS}	0.36	0.36	0.35	0.36	0.28
V_{AS}	83.3 ltr	85.2 ltr	94.7 ltr	92.2 ltr	104 ltr
SPL 2.83V	85.7 dB	85.6 dB	85.3 dB	85.4 dB	89 dB
X_{MAX}	12.5 mm	12.5 mm	12.5 mm	12.5 mm	12.5 mm

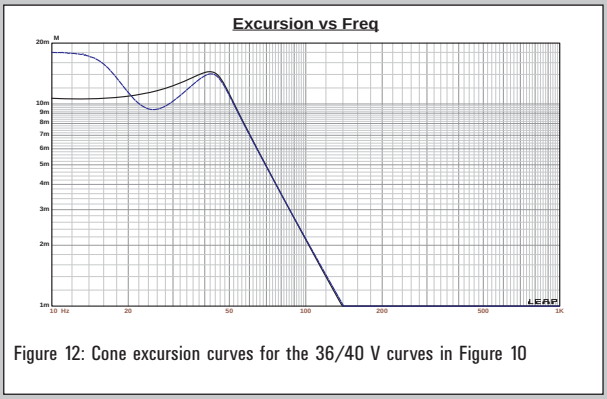

Figure 12: Cone excursion curves for the 36/40 V curves in Figure 10

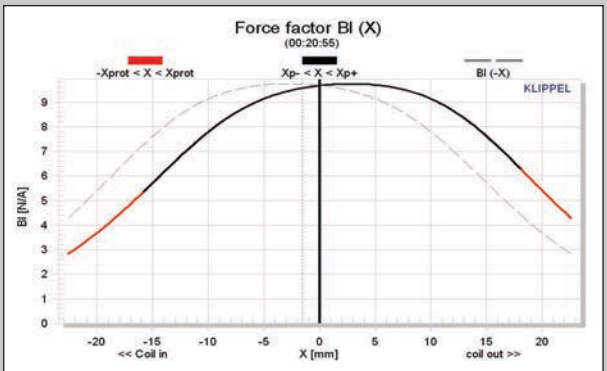

Figure 13: Klippel Analyzer Bl (X) curve for the Scan-Speak 26W/4558T00

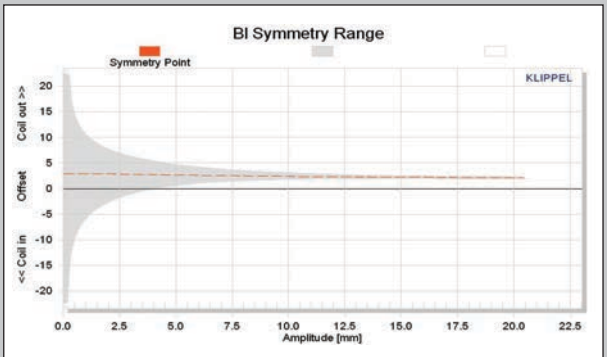

Figure14: Klippel Analyzer Bl symmetry range curve for the Scan-Speak 26W/4558T00

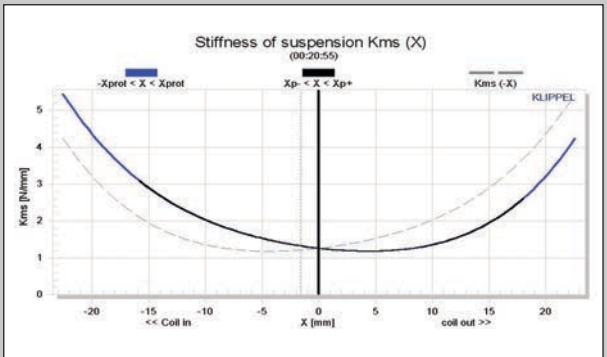

Figure 15: Klippel Analyzer mechanical stiffness of suspension Kms (X) curve for the Scan-Speak 26W/4558T00

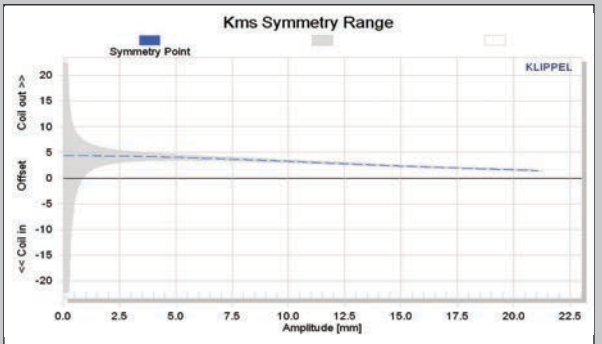

Figure 16: Klippel Analyzer Kms symmetry range curve for the Scan-Speak 26W/4558T00

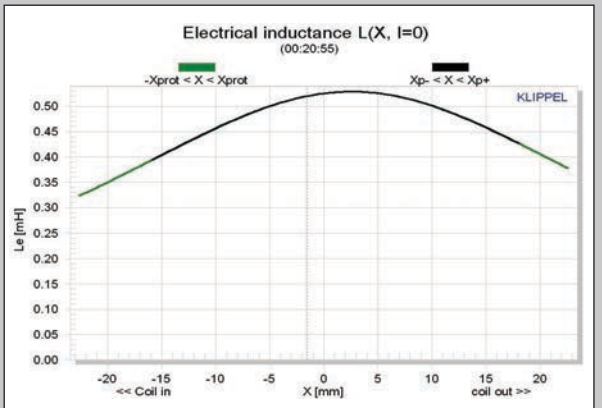

Figure 17: Klippel Analyzer L(X) curve for the Scan-Speak 26W/4558T00

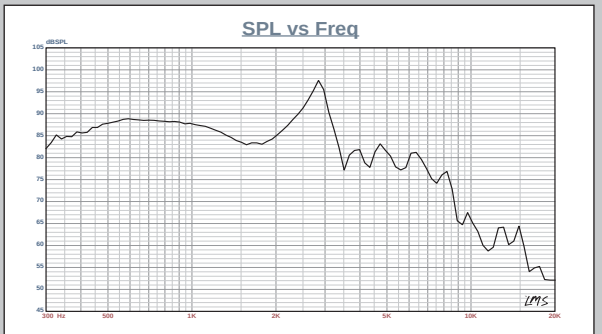

Figure 18: Scan-Speak 26W/4558T00 on-axis frequency response

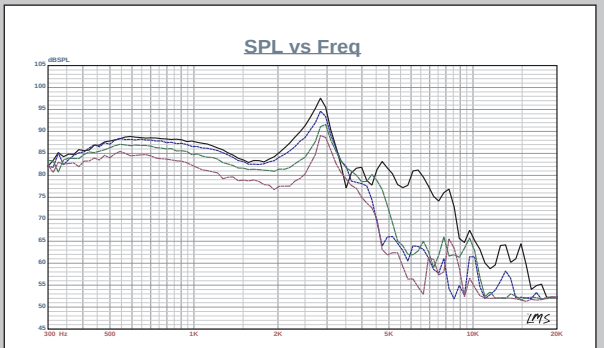

Figure 19: Scan-Speak 26W/4558T00 on- and off-axis frequency response

Sample 1. I set up two box simulations, one sealed and one vented. For the closed box simulation I used a 1.0 ft^3 enclosure with 50% fiberglass fill material, and for the vented box, a 1.7 ft^3 QB3-type vented alignment with 15% fiberglass fill material and tuned to 24 Hz.

Figure 10 illustrates the results for the Scan-Speak 26W subwoofer in the sealed and vented boxes at 2.83 V and at a voltage level high enough to increase cone excursion to Xmax + 15% (14.4 mm for the 26W). This yielded a F3 = 40 Hz with a box/driver Qtc of 0.67 for the 1.0 ft^3 sealed enclosure and –3dB = 33.5 Hz for the 1.7 ft^3 vented simulation.

Increasing the voltage input to the simulations until the maximum linear cone excursion was reached resulted in 111.2 dB at 40 V for the sealed enclosure simulation and 106 dB with an 16-V input level for the larger vented box (see **Figures 11** and **12** for the 2.83 V group delay curves and the 36/40 V excursion curves).

Klippel analysis for the Scan-Speak 10" subwoofer produced the Bl(X), Kms(X) and Bl and Kms symmetry range plots given in **Figures 13** to **16**. The Bl(X) curve for the 26W (see **Figure 13**) is very broad and reasonably symmetrical, typical of high Xmax drivers. Looking at the Bl Symmetry plot (see **Figure 14**), this curve shows a coil forward (coil out) offset that goes from 2.8 mm at rest to about 2.3 mm of excursion at the physical Xmax position. Because it stays fairly constant throughout the operating range, this is likely a physical offset. **Figures 15** and **16** give the Kms(X) and Kms symmetry range curves for the Scan-Speak 26W. The Kms(X) curve (see **Figure 15**) is also as moderately symmetrical as the Bl curve, and likewise has a coil-out offset. In the Kms symmetry range curve in **Figure 16**, the coil-out offset is 4.4 mm at rest decreasing to about 3 mm at the physical Xmax position. From the displacement limiting numbers calculated by the Klippel analyzer for the 26W, the XBl @ 70% is Bl = 12.6 mm and for XC @ 50% Cms, the minimum was 13.2 mm that means that for the 26W woofer, Bl was the limiting factor for the prescribed 20% distortion limit. This occurs at just about the physical Xmax of the driver, so not bad.

Figure 17 gives the inductance curves L(X) for the 26W. The curve illustrates a decreasing inductance as the coil moves inward, the result of the aluminum shorting sleeve in this motor system. Voice coil inductance varies only about 0.16 mH from Xmax in to Xmax out (0.43 mH to 0.59 mH), which is very good given the amount of wire in the voice coil.

Even though the 26W is a subwoofer and could likely be used with a plate amp limiting its upward extension to 100 Hz, this 10" driver could also find application in three-way designs. Given that, I mounted the 26W/4558T00 subwoofer in an enclosure which had a 17" × 8" baffle and was filled with damping material (foam) and then measured the DUT on- and off-axis from 300 Hz to 20 kHz frequency response at 2.83 V/1 m using a 100-point gated sine wave sweep. **Figure 18** gives the 26W's on-axis response, displaying a very smooth rising response to about 600 Hz, followedby a 6 dB drop in SPL to about 1.8 kHz, followed by the start of the aluminum breakup mode centered on about 2.8 kHz.

Figure 19 has the on- and off-axis frequency response at 0°, 15°, 30°, and 45°. With respect to the on-axis curve, –3 dB at 30° occurs at 1.0 kHz, so a cross point anywhere below that frequency should be fine. It certainly

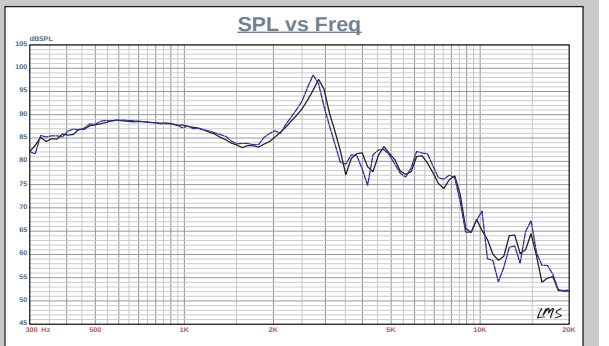

Figure 20: Scan-Speak 26W/4558T00 two-sample SPL comparison

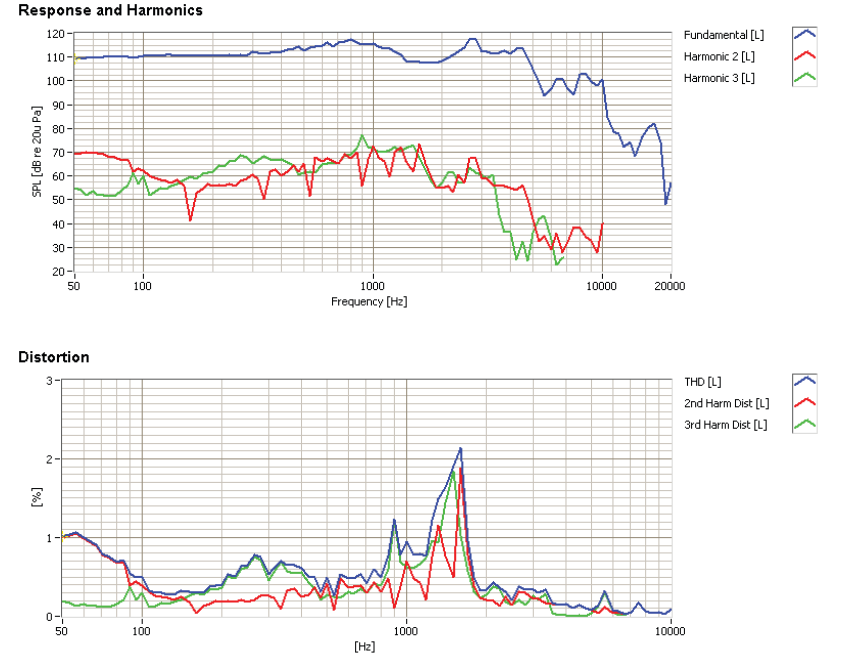

Figure 21: Scan-Speak D2604/833000 SoundCheck distortion plots

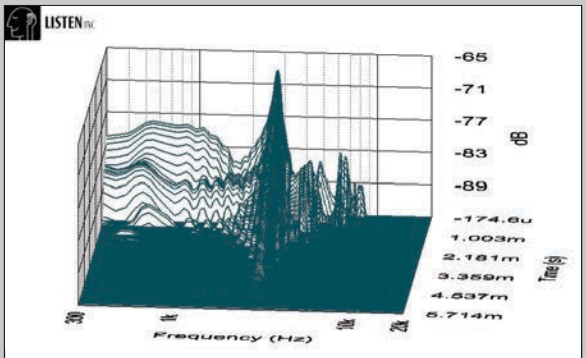

Figure 22: Scan-Speak 26W/4558T00 SoundCheck CSD waterfall plot

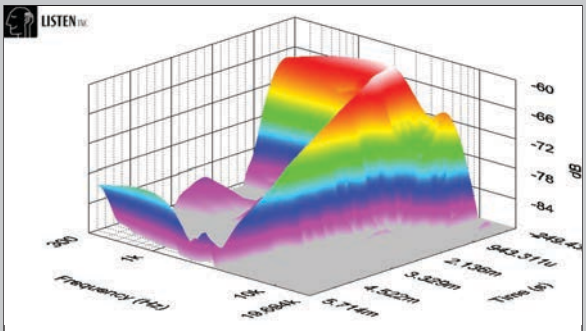

Figure 23: Scan-Speak 26W/4558T00 SoundCheck Wigner-Ville plot

would be possible to match it to a ribbon driver in a two-way at that frequency, but a more likely configuration would be combining the 26W with a 5.25" midrange and a 1" dome. The last SPL measurement is shown in **Figure 20** and gives the two-sample SPL comparisons for the 10" Scan-Speak driver, showing a close matchup within the relevant operating range of the woofer.

For the last group of tests, I employed the Listen, Inc. SoundCheck analyzer (courtesy of Listen, Inc.) and SCM microphone to measure distortion and generate time frequency plots. Setting up for the distortion measurement consisted of mounting the woofer rigidly in free-air, setting the SPL set to 94 dB at 1 m using a noise stimulus (SoundCheck has a software generator and SPL meter as two of its utilities), and then measuring the distortion with the Listen, Inc. SCM microphone placed 10 cm from the dust cap. This produced the distortion curves shown in **Figure 21**. Last, I employed the SoundCheck analyzer to get a 2.83 V/1 m impulse response for this driver and imported the data into Listen, Inc.'s SoundMap Time/Frequency software. The resulting CSD waterfall plot is given in **Figure 22** and the Wigner-Ville logarithmic surface map (for its better low-frequency performance) plot in **Figure 23**.

Looking over all the data presented, this is a very well designed driver, and another good one from Scan-Speak. For information on these and the other Scan-Speak drivers, visit the Scan-Speak website at www.scan-speak.dk. *aX*

100% Audio

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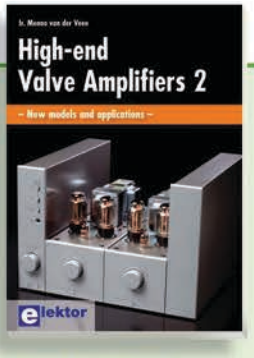

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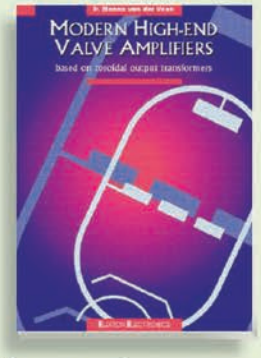

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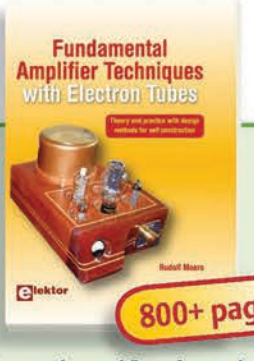

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DVD-ROMS

By Wim de Haan (The Netherlands)

The Abletec Class D Amplifier

Class D technology for home use

Some of you might recall the Sinclair X-10 PULS Width Modulation (PMW) Class D amplifier kit as sold in 1964 (see **Photo 1**). Well, the market for Class D amplifier technology has substantially increased over the last few years. Respected audio manufacturers such as Rotel, Onkyo, Pioneer, and T+A are using Class D technology with success in their designs.

High-tech companies such as Cary Audio, Bel Canto, and Jeff Rowland are using Class D for serious listening. B&O's IcePower ASC200 module found its way into subwoofers used by Canton, KEF, Adam Audio, Martin Logan, and B&W. And, UcD Class D technology by Hypex found its way into Bryston, Meridian, and Micromega products. An extraordinary Class D solution is offered by Lyngdorf (TACT), Flying Mole, NuForce, Marten, Halcro (Lyrus), and Devialet.

Class D for DIY purposes might have some disadvantages—board layout and component choices can be critical. For optimum results, use of SMD is almost mandatory. But some companies do offer complete working amplifier boards. Some use Tripath technology, some are using the Philips TDA892X/TDA8950 family, some Hypex UcD modules, and some are using IR's IRS2092. Ti's Purepath TAS5630 Class D technology is offered by Classdamp.com. Some Class D DIY kit suppliers include: 41hz.com, Connexelectronic.com, Hifimediy.com, Hypex.nl, CADaudio.dk, Classdamp.com, and Audiopower.it.

The UK-based Profusion company is the former reseller of Tripath products in the UK and Europe. Tripath has been out of business for a few years now. But, on a visit to Profusion's website in early 2011, I noticed the new Class D amplifier range by Abletec and paid attention because the available amplifier boards are based on a patent dated from 2010. Patrik Boström from Anaview AB applied for this patent; the patent reference number is WO2010126440A1. (Refer to the sidebar for an interview with Patrik Boström.)

ABLETEC AND ANAVIEW AB

Norway-based Abletec is an established manufacturer of high-quality customized power supplies to the demanding international electronics market. Anaview was founded in 1980 under the name Bostrom Industry Consulting AB by Christer Boström, who worked as a mechanical designer. In 2001, Christer's son, Patrik Boström, became the owner and started working with analog electronics. Today, Anaview is situated in Helsingborg in the south west of Sweden. Anaview consists of dedicated analog and digital designers with much experience and a higher education. Anaview delivers standard and customized OEM products to the audio industry. In special cases, contracting missions are also performed.

The two companies have enjoyed a strategic partnership since 2008, and in 2010, Abletec acquired Anaview AB. Together, the two companies represent a well-built player within audio amplification. They provide customers with numerous advantages, including an extensive product portfolio, skilled and stable R&D department, keen sales organization, and a well-developed manufacturing and supply chain. "This merger enhances the technology we provide in step with the development of this market, and both the employees and I am very satisfied with this choice," said Patrik Boström, entrepreneur and majority owner of Anaview AB. Under the agreement, Boström was named chief technology officer (CTO) of Abletec, bringing valuable knowledge, experience, and technology management to supplement the R&D department.

CLASS D POWER AMP BOARDS

The Audio Line Combination (ALC) Series combines Class D power amplifiers with Abletec's Zero Voltage Switching (ZVS) switching power supplies for greater than 85% efficiency. With Energy Star compliance, regulatory approvals, and compact open-frame form factors, the ALC Series is ready to unleash creative design potential, accelerating development of next-generation consumer electronics, musical performance, and professional audiovisual systems.

Photo 1: The Sinclair X-10 PULS PMW Class D amplifier kit as sold in 1964.
(Photo courtesy of Planet Sinclair/Chris Owen)

The amplifier technology in this module range is a globally-modulated, self-oscillating topology which controls the signal directly at the output. The flat, load-independent frequency response, high bandwidth, low distortion, low noise, and perfect transient response makes it suitable for the most demanding applications. Also patent-pending Adaptive Pole Control (APC) is used to deliver Class AB audio performance with the efficiencies of Class D. Overcurrent, overvoltage, and overtemperature protection is built in and balanced analog inputs are offered. With damping factors of 4,000 and higher, ALC Series amplifiers excel when powering highly reactive woofers and subwoofers.

A powerful switch mode power supply is incorporated giving 180-W RMS power with 360-W peak output power for short periods of time to cover the reality of music. This is specific for ALC0180 (see **Photo 2a**, **2b**, and **2c**). The power supply is inherently designed for ultra-low noise, making it well suited to the demands of audio and consumer electronic products. It also delivers ± supplies to an external PCB. For example, a crossover board or preamplifier board in a dual mono integrated amplifier.

The built-in overcurrent protection has two modes. During transients into the current limit threshold, it allows constant output current; but if this condition persists, it goes into burst mode. This guarantees that the amplifier does not produce audio holes during heavy listening but still effectively protects it during short circuiting.

Profusion offers four different ampli-

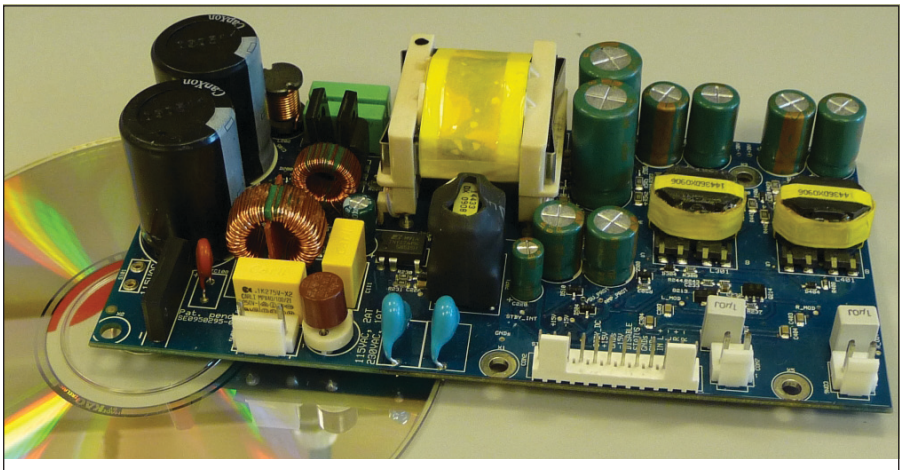

Photo 2a: Abletec ALC0180-2300 stereo amplifier module with integrated power supply

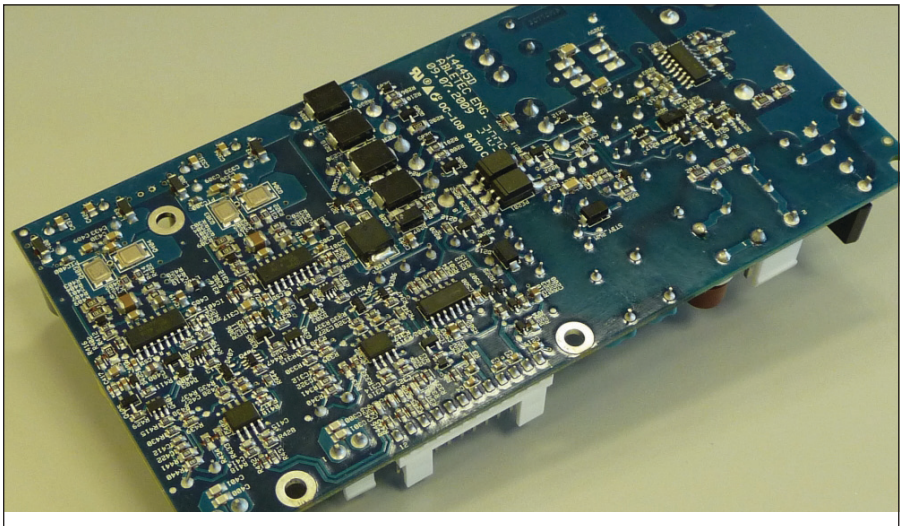

Photo 2b: Solder side of the ALC0180-2300 board using SMD throughout

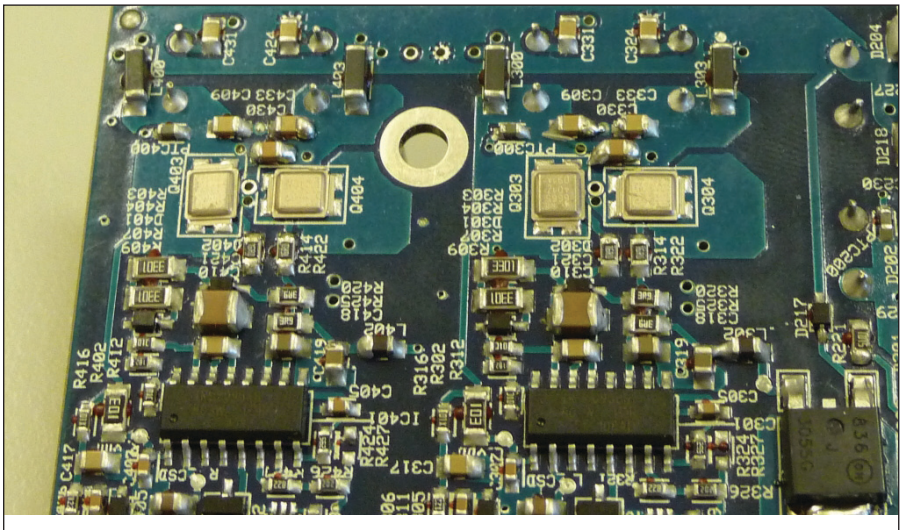

Photo 2c: Output stage section ALC0180-2300 board using no heatsinking whatsoever

Overview Abletec amplifier modules

Module	ALC0180-2300	ALC0240-2300	ALC0300-1300	ALC1000-1300
Power in 8 Ω	2 × 50 W	2 × 65 W	1 × 150	1 × 600 W
Power in 4 Ω	2 × 90 W	2 × 120 W	1 × 320	1 × 1000 W
Bridge mode	1 × 180 W/8 Ω	1 × 200 W/8 Ω	-	-
Channels	2	2	1	1
THD+N	<0.05%	<0.05%	<0.05%	<0.05%
Freq.range ±0.5dB	20 Hz – 35 kHz	20 Hz – 35 kHz	20 Hz – 35 kHz	-
Freq. –3 dB	60 kHz	60 kHz	60 kHz	50 kHz
Integrated PSU	yes	yes	yes	yes
115 & 230 V use	yes	yes	yes	yes
Dimensions mm	150 × 75 × 38	-	154 ×120 × 58	180 × 166 × 63
Price at Profusion ex VAT + shipping	$102	$150	$205	$409

Table 1: This is an overview of the Abletec amplifier modules. Note: The ALC0100 (2 × 5 W), the ALC0200 (2 × 40 W + 1 × 120 W), and the ALC0600 (2 × 300 W stereo) are under development.

fier boards, ranging from 50 W to 600 W (see **Table 1**). These boards are obtainable for anyone who has interest, so you don't need to be a manufacturer or company to buy them.

PRACTICAL REALIZATION

An ALC0180-2300 module was chosen for this project. The 2 × 50 W into 8 Ω or 2 × 90 W into 4 Ω was found perfect for "living room" use. The amplifier module comes as a complete working module; no soldering or adjustments have to be made. Only input source, speakers, and mains have to be connected.

The first thing you will notice is that the module does not use any kind of heatsinking, not even a small one. The module uses SMD throughout, with some custom ICs, and comes complete with datasheet and documentation.

The PCB looks very professional and is only 150 mm × 75 mm. The height is 38 mm. And remember this board includes the power supply, so you don't have to buy a mains transformer or mains capacitors. Even with its 2 × 90 W into 4 Ω at 1% distortion, the amplifier does not use any heatsink. The 4 IR DirectFet MosFets are shown in one of the figures, including the custom Anaview.com IC (see **Photo 3**). This IC is based on IR's IRS20955/6/7 family.

Mains, loudspeaker, and input use standard connectors that are available at companies like Digikey or Farnell. Profusion does sell a dedicated Abletec cable set (stock no. ALC-MP133-1) for just $2.73.

The board is made in such way that the ground of it will automatically connect to the chassis ground using a M3 bold; the upper side is connected to ground (see **Figure 1**). The module does offer differential inputs, but if used with asymmetrical inputs (unbalanced inputs), the input has to be connected with the signal ground of the preamplifier. The author used isolated Neutrik cinch input sockets and balanced cable. The two internal wires are connected to the inputs of

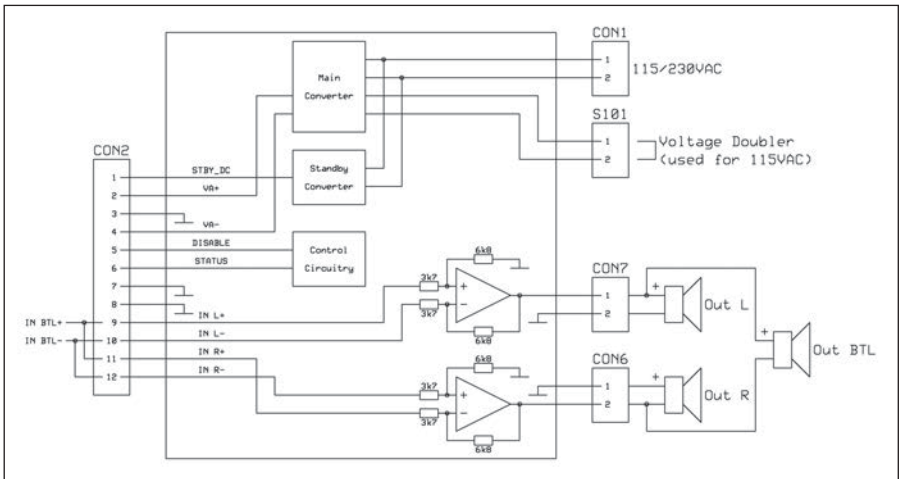

Figure 1: The bridge mode connection diagram for the ALC0180

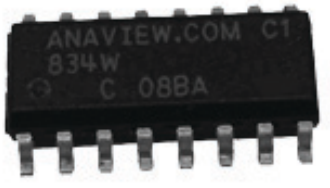

Photo 3: Custom-made Anaview.com IC manufactured by IR. Not available for DIY purposes.

Photo 4a: Front overview Abletec class D amplifier

Photo 4b: Back overview Abletec class D amplifier

Photo 4c: Inside overview Abletec class D amplifier

the board, this to L– and L+ and to R– and R+. The two shields are connected together as the two GND's of the input connector (this at one point of one the inputsocket GND). These GND's are connected together with one wire to point 7 or 8 of the board. The shields of the internal cable are left open at the board side.

In the datasheet, it is clearly stated how this module works with 230 VAC or 115 VAC. Do take notice of this.

An enclosure by www.hifi2000.it is used. Some might feel it is a little big for such a small board, but the author had three of these enclosures on hand and did not want to buy an additional one at extra cost.

UP AND RUNNING

This is one of the most easy to build amplifiers. Only a few connections have to be made (see **Photos 4a–c**). No difficulties were encountered during the building process. The amplifier was running in no time, and for the last few months, the author has been using this Class D amplifier for daily use with no problems.

Figures 2a, **2b**, and **2c** show the ALCO180-2300's test results. The module offers a professional-made amplifier board, high performance specs, and sound at a very attractive price. In fact, it's a steal. You will get a patented power supply and a patented two-channel amplifier all-in-one—with no heatsinks, no fancy looking power supply capacitors and no toroidal mains transformer—simply, because you don't need them. The product is highly recommended! *aX*

ALC0180-2300 module test results

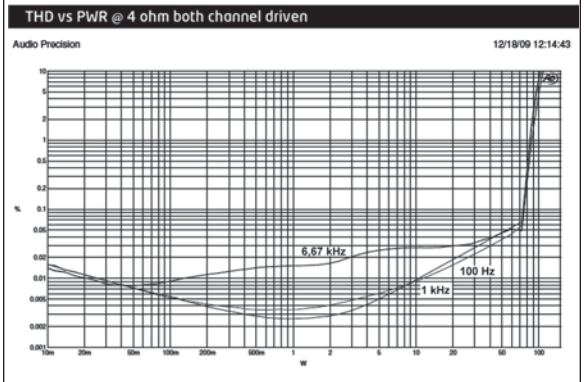

Figure 2a: ALC0180-2300 module THD vs PWR at 4 Ω with both channels driven

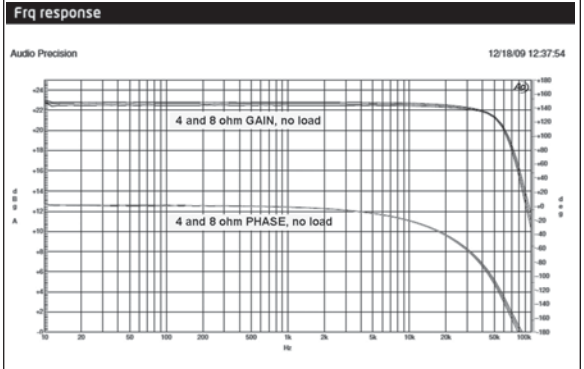

Figure 2b: ALC0180-2300 module frequency response

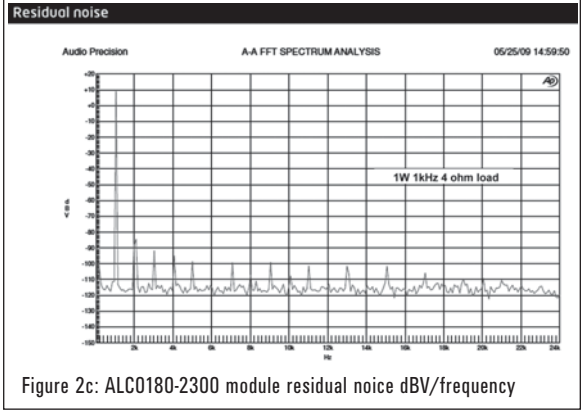

Figure 2c: ALC0180-2300 module residual noice dBV/frequency

Editor's note: Wim de Haan created a table titled "Class D Technology for Home Use." You can find it at www.audioamateur.com.

SOURCES

ALC0180-2300 Module
Abletec
www.abletec.com

Custom Anaview.com IC
Anaview
www.anaview.com

Ablatec amplifier modules
Profusion
www.profusionplc.com

RESOURCE

WIPO
Patent: WO/2010/126440A1
Link patent: http://www.sumobrain.com/patents/wipo/Power-amplifier/WO2010126440A1.pdf

An Interview with Patrik Boström

By Wim de Haan

Wim: Where did it all start for you and audio?

Patrik: I just love music and I have always been the type of person who wants to build everything myself, even if things are readily available. When I was younger I built my own bicycles and skateboards. When I was 14, I ripped apart a pair of SONY speakers that came along with a low-cost rack system and put them in a home-built ¼ wave horn for which I found the blue prints in a very old Philips book. It became a big success and at least four of my friends built similar ones. When I was 16, I built my first amplifier, a big 150 W Class AB design with Sankens BJTs.

Wim: Is there a person or company in the audio industry that inspired you to enter the audio business?

Patrik: I have always been inspired by "no compromise" designs like Krell, Mark Levinson, etc. Also, Pioneer's big mono blocks from the '90s were really beautiful. A close friend of mine, Juha Välitalo who lived in Halmstad where I studied at the university, inspired me in the way that he was really interested in everything I built and he always invited me to listen to all the equipment that went in and out of his apartment (which was a lot).

Wim: How and when was your company, Anaview AB, formed?

Patrik: After university studies, I started working as a power supply designer at Ericsson in Stockholm. I learned a lot about power electronics and met a lot of talented people who really knew analog electronics. When we moved from Stockholm, I started my own contracting business (using my father's company) and continued to develop power supplies, mainly for the military industry but also for some consumer and telecom applications.

In 2004, I eventually gave in to the idea of pulse width modulating the audio signal in order to improve efficiency. It was actually Audio Pro in Helsingborg that called me up and asked if I knew how to design Class D amplifiers. I did not at that point but it triggered me to look into it. One can say that Anaview, as it was when we merged it with Abletec, was formed in 2004.

Wim: Can you tell us more about the topology of your Class D design?

Patrik: The core technology in all of our amplifiers is an analog self-oscillating modulator which modulates directly on the output of the amplifier. The Adaptive Pole Control (APC) patent, which is used in the ALC line, is actually not a Class D patent but a patent around an error amplifier in a negative feedback topology. However, the advantages of it are especially important in a Class D amplifier.

Every time an amplifier goes into hard clipping, current limit, or just when it starts up, the error amplifier (if it has a lot of gain that is) will go into a positive or negative end state. Once the error becomes smaller again, the error amplifier will take some time to get away from the end state and go into normal operation. Not only does this sound terrible, but in a self-oscillating Class D amplifier, it might also cause the amplifier to go into a destructive low-frequency oscillation mode.

The APC patent is a circuit that turns the error amplifier into a follower during those situations and thereby shortens the recovery time significantly. This has been done in many ways throughout the history of analog electronics and the most common way is to use back-to-back zener diodes. In a power supply this is okay, but not in an audio amplifier as the leakage of these diodes causes significant nonlinearities. Our APC circuit realizes ideal zener diodes.

Wim: Can you explain, in short, the working of the amplifier?

Patrik: Above is a block diagram (see **Figure 1)** from the APC patent application. It shows a globally modulated Class D stage with negative feedback around it. It also shows a small box around the error amplifier, which is the APC circuit.

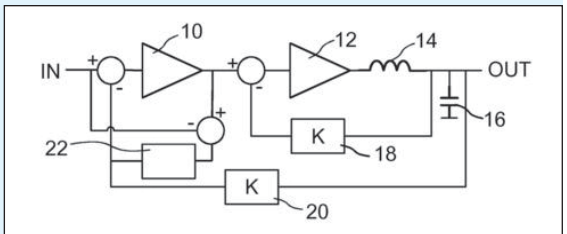

Figure 1: A block diagram from the APC patent application

Wim: What is unique about Abletec Class D module as mentioned in the article?

Patrik: There are a few Class D makers in the world that have good performance but most of them are missing one or two characteristics. If the distortion is low, it might have a crooked-frequency response and if the frequency response is straight it might have poor efficiency or high distortion.

Our technology has been refined over some years now and does not really have any weak points. We have also combined our knowledge about clean switch mode power-supply design with the amplifier topology to provide a complete unit, which makes it very easy for companies to put it in their products and ship it. I still think we are the only ones who ship modules that are Energy Star compliant. Not that this is hard to achieve, but it shows that there are not many complete alternatives out there.

Wim: Anaview uses custom-made integrated circuits. Can you tell us something about this?

Patrik: It is very expensive to make your own ICs but all the ICs available in the market miss some characteristics. IRs own gate drivers are very good in general but only had over current shut down which creates audio holes. This means that the OCP threshold has to be avoided all together, as this state sounds really bad. Since we have a long and positive relation with IR, we simply purchased our own version. Into it, we designed a double mode OCP, which allows the amp to run at the OCP level for a while before going into shut down mode. The periods are fully adjustable, using passive parts around it. Thanks to this we get a lot of flexibility when we design for extreme situations (like car subwoofer amplifiers) as

we can actually run at the OCP level forever if we want.

Wim: Your testing method, using the Prism audio analyzer, does not just seem to concentrate on THD+N. Can you tell us more about this?

Patrik: THD tells a lot about an amplifier but there are some measurements that can be very useful as complement. For example, looking at the distortion residual can give you a lot of clues to what actually causes the distortion. Also, measuring IMD at 18+19 kHz tells you a lot about how much loop gain the amplifier has left in the top end of the audio band.

Wim: What is your idea about switching power supplies vs. conventional transformer diode capacitor-based power supplies?

Patrik: The Abletec ZVS topology we use actually behaves like a traditional power supply as it is non regulated (the rail voltages vary with the mains voltage). The reason why this works fine is that audio amplifiers usually have enough PSRR to reject the resulting ripple as it is at low frequencies. This is not the case for all Class D amplifiers, but it is for our Class D technology.

I can honestly say that the more stable your supply voltage is the better and the less noise it radiates the better. Therefore, batteries would be the best alternative when it comes to sound but we must make modules that are sellable.

Wim: Should musicality be the prime goal of audio designer?

Patrik: It depends on the budget. I always try to create the best audio performance I can with the money I have available. The ALC line from Abletec has been designed to be a good compromise of usability, size, audio performance, and cost. But when I build amps where cost is not an issue, of course, they are quite different.

Wim: Can you tell us more about the Marten M-Amp Class D amplifier?

Patrik: Marten is a highly-awarded speaker brand which has been around for a while. They received very good reviews of their Coltrane speaker and, in their projects, cost is not an issue. When I filed the patent application for the new Adaptive Modulation Servo (AMS) technology, I immediately started thinking about how to introduce it to the market. AMS is our next generation technology. It further improves the performance and actually reduces the complexity of the circuit although it is very simple as it is. With AMS you get a first-order system (pure delay line) that has basically identically low distortion at all the frequencies in the audio band. These are the main differences compared to APC.

The gentlemen at Marten and I spoke about this and agreed that it would be very interesting to do something together, and that is how it started. Marten said that it is very important to have a lot of power margin as they had seen most amps in the market run out of voltage when driving their biggest speakers. The power of 500 W/8 Ω is what you can achieve using a non-regulated power supply and 150-V devices in a balanced configuration. When you move up in voltage, you need to start using devices that are not optimal for Class D design.

Wim: What do you think about Class AB vs. Class D?

Patrik: The performance of a well-designed Class AB amplifier is the goal for our Class D designs. The day we can achieve that is the day we retire. But, that is when looking at measurements. Sound, as you know, is very subjective and I get a lot of e-mails from people who test our modules and claim that the dynamics and purity of them is amazing, among the best they have heard. It is difficult being a designer in those situations as it is not always straight forward to explain their experiences in technical terms. I wish that some day I have enough time to do research of my own in this field.

Why do we hear what we hear in amplifiers? As it is now, I strive to make amplifiers as simple as I can and with as good a performance as possible. My personal favorite quote is, "A designer knows he has achieved perfection, not when there is nothing more to add but when there is nothing more to take away." (Antoine de Saint Exupéry). I love this quote so much it is scary.

Wim: Do you have a dream product or is it realized already?

Patrik: Wow, well, the first thing that comes to mind is a "cost is not an issue" active speaker system. I am really interested in the work and findings of Earl Geddes and am right now building an active system based on his theories (not the "cost is no issue" version though). I would complement the front speakers with four active subwoofers. The amplification would be 400 W/4 Ω AMS amplifiers with a big power supply, consisting of a PFC stage and a synchronously rectified DC/DC-converter. Ideally, the DACs would be placed in the speakers, meaning the DSP would have digital outputs and volume control in the digital domain.

Wim: What do you listen to at end of the day, when work is over?

Patrik: It depends on the mood I am in. When I am blue, I listen to blue music. When I am happy, I listen to happy music. I love Patricia Barbers' latest records, and I consider her to be a true artist. The first two records of Stina Nordenstam are also timeless, in my opinion. Even though they are 20 years old and I have heard them a thousand times, I can still listen to them with the same feeling as the first time. The band I have the most records of is the Yellowjackets. They play fusion jazz as if they are engineers. Each tone has its place and its use, and they simply rock. The record "Like a River" is my favorite Yellowjackets record.

Wim: What are your future plans?

Patrik: I am still not warm in my clothes as CTO for Abletec. There are a lot of things to take care of because of the merger. When things settle down, I will start to focus on further developing our technology to maintain the role we have today. We also have some new modules coming out soon that fill needs that we cannot quite fill today.

Wim: Any last thoughts for the readers of *audioXpress*?

Patrik: If you feel like working for Abletec as an audio designer, feel free to send us your resume. We know that it is not always possible to get people to move to Norway or Sweden so do not consider this a requirement. We just need crazy audio enthusiasts who love electronics and we will try our best to adapt to wherever they are. *aX*

By Marcel Cremmel (The Netherlands)

MP3 Pre-Stereo Microphone

A mic pre-amp for MP3 players

There are still a lot of hobbyists who enjoy making high-quality live audio recordings of concerts, interviews, radio broadcasts, natural sounds, and so on. Professional equipment such as Nagra and the like is simply too expensive. We can thus be thankful that many MP3 players are equipped with an analogue input and a high-performance MP3 or WMA encoder.

But, you need portable equipment if you want to make recordings in the field. Professional equipment for this purpose is very expensive, and many affordable alternatives (such as MP3 players) cannot handle microphone signals.

Microphones require some sort of assistance to supply the high signal levels that must be provided on the line inputs. Although some players have built-in microphones, the quality is often barely adequate for use as a dictating machine. This means you need a microphone pre-amplifier to adapt the microphone output to the line input.

The characteristics listed in the "MP3 pre-amplifier features" inset can be obtained without using any special components. Transistors and opamps that you have tucked away in a drawer somewhere should do just fine. Aside from the potentiometers and connectors, all the components are SMD types. But there's no need to be afraid—even beginners can assemble the circuit board if they are blessed with a grain of patience, an ounce of dexterity, and a pinch of precision.

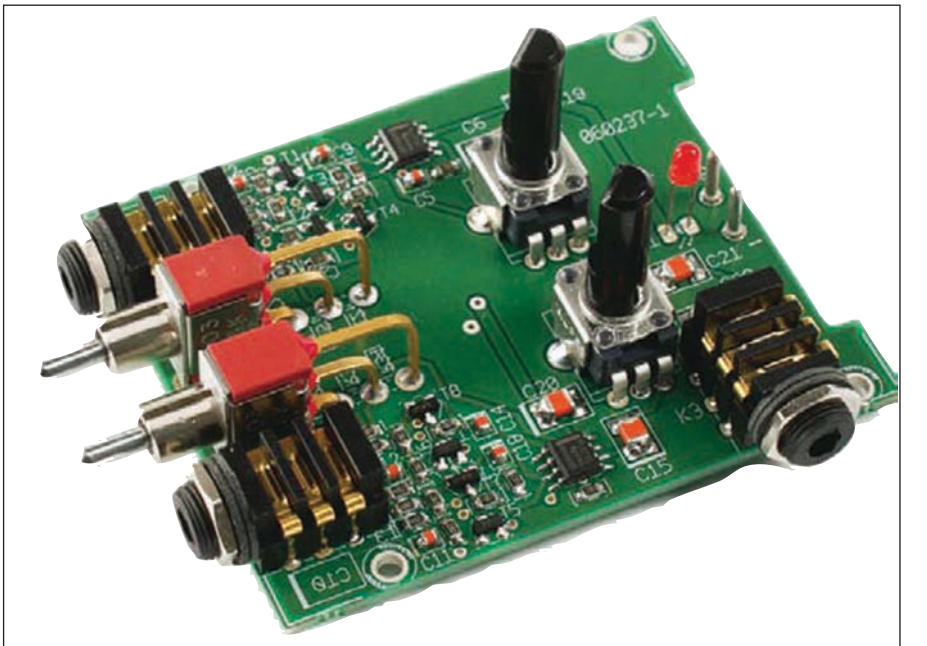

Photo 1: The preamplifier described here is optimized for a dynamic microphone.

MICROPHONES

Even the best pre-amplifier won't perform well with a mediocre microphone or a poor cable. A wide variety of systems are available, with prices ranging from a few dollars to several hundred. The pre-amplifier described here is optimized for dynamic microphones.

The characteristics of the microphone (see the "Principal characteristics of a microphone" inset) are a major factor in selecting a microphone.

The Sennheiser MD21 is a genuine reporter's microphone that has been on the market for several decades already. It has a sensitivity of 1.8 mV/Pa and generates an output signal of 0.36 mV in a quiet room, or 3.6 mV if you aim it toward a pneumatic hammer.

It has a very large dynamic range: 3.6 mV, 0.36 mV is a healthy 80 dB. Based on this information, we can specify the following properties of an ideal pre-amplifier:

- An input noise level that is significantly less than 0.36 mV, in order to obtain a satisfactory signal-to noise-ratio (SNR).
- No overdrive or signal distortion at 3.6 mV.

A good SNR can be achieved by using a well-considered design and choice of components for the pre-amplifier. The second property requires using a potentiometer so the output level can be adjusted to suit the maximum acoustic pressure.

The distance between the microphone and the pre-amplifier is often more than 10 m. The cable used for this link is thus exposed to interference from magnetic fields, in particular AC mains fields, at a frequency of 50 Hz. The signal carried

Features

- Stereo
- Compatible with dynamic and electric microphones
- Maximum input sensitivity: 50, 60, or 70 dB, selectable
- Level control using two potentiometers
- Bandwidth: 50 Hz to 50 kHz at maximum gain (70 dB)
- Distortion: 0.04% at –10 dBV (316m V_{eff}) 0.12% at 0 dBV 91 V_{eff}
- Input Noise: –24 dBV (0.63 μV_{eff})
- Power source: 9-V battery
- Current consumption: 3 mA per channel
- Designed to fit in a commercially available enclosure with a battery compartment

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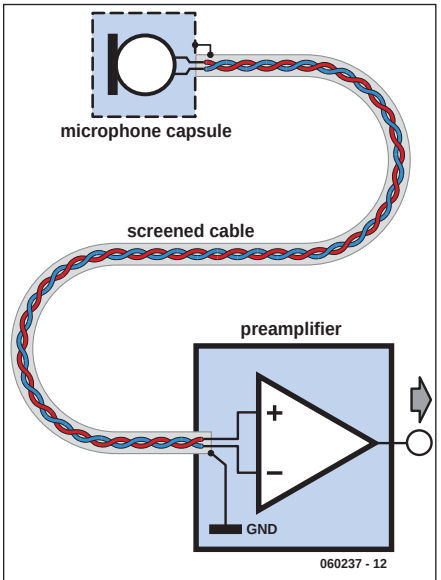


Figure 1: Construction of the screened cable link between the microphone and the pre-amplifier

by the cable has a low amplitude and must be protected against this interference. Otherwise, the interference will be audible as the well-known "mains hum." This protection is provided by the screen of a screened cable.

For proper operation, the screen must be connected to ground, and it cannot be used to carry the signal.

Despite this protection, some interference signals still manage to reach the active conductors. If these conductors have the same impedance relative to ground, the induced signals will also be the same. Their effect can then be neutralized by the differential input stage of the pre-amplifier. In addition, the low impedance of a dynamic microphone reduces the effect of external interference.

In short, use a screened cable with two conductors and solder the cable together as shown in **Figure 1**.

SCHEMATIC DIAGRAM

The schematic diagram consist of two identical amplifier channels. Here we describe one of the two channels.

The circuit is powered by a 9-V battery. An LED wired in series indicates the pre-amplifier is switched on. The voltage drop of the LED (1.6 V to 2 V for a red LED) does not affect the performance of the preamp.

The supply voltage is switched via the contacts of the audio socket. In the quiescent state, two contacts prevent T9 and T10 from conduct-

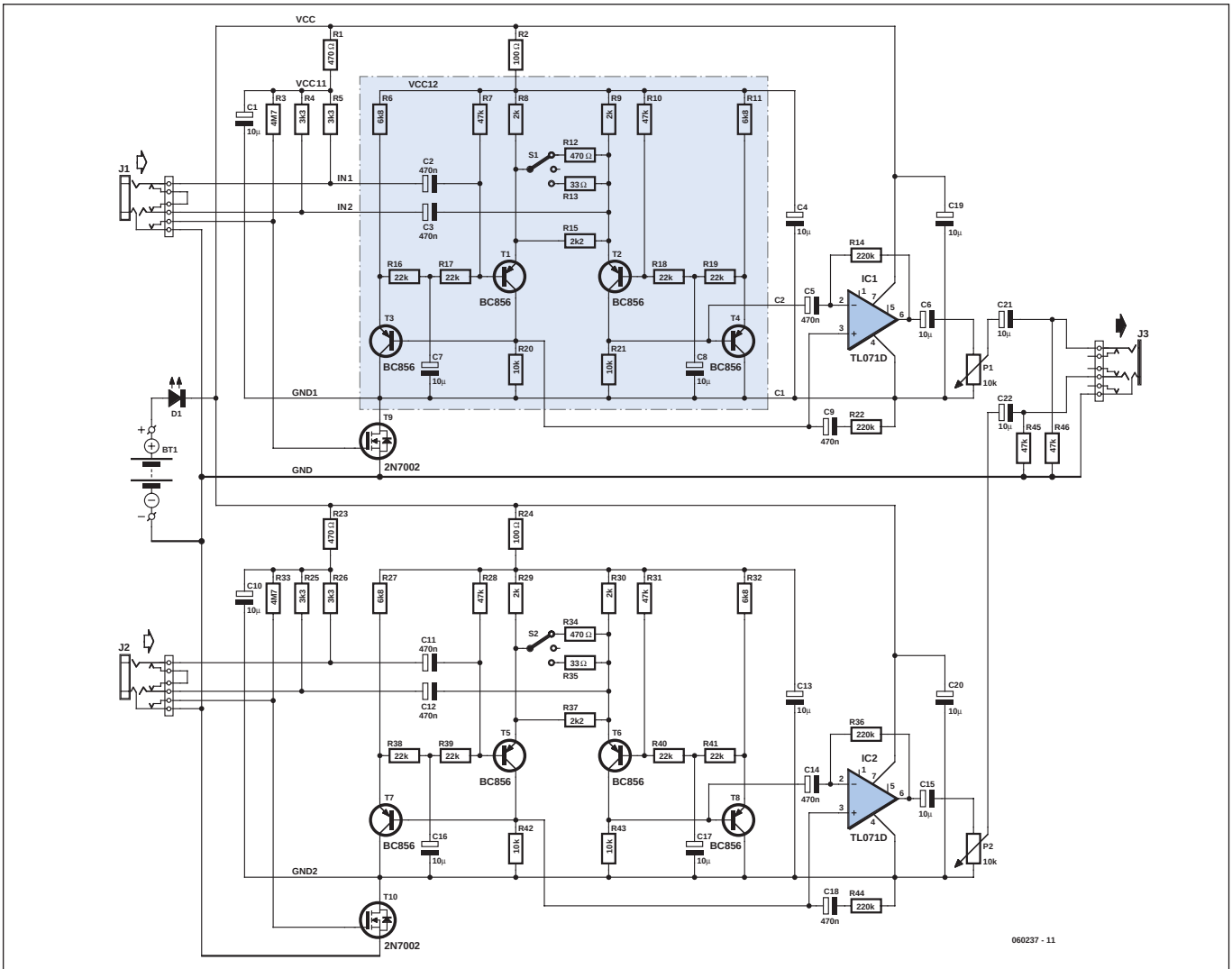


Figure 2: Full schematic diagram of the microphone pre-amplifier for MP3 players

Mic characteristics

- Sensitivity in mV/Pa
- Output impedance
- Bandwidth
- Directional sensitivity
- Maximum acoustic pressure

An electronics specialist is interested in the first three characteristics, while a recording specialist is interested in the last three.

The signal voltage provided by a microphone is proportional to the acoustic pressure and the sensitivity of the microphone:

- $V_{mic} = S \times P$
- S: sensitivity in V/Pa
- P: acoustic pressure deviation in Pa

ing. The TMOS transistor is thus cut off, so no voltage is applied to the other components. When a microphone cable is plugged in, the contact in the socket is opened. The supply voltage is now applied to the gate of T9 or T10 via R3 or R33. The TMOS transistor is switched on and provides power to the amplifier channel.

The high resistance of R3 and R33 limits the current consumption to 1.6 mA per channel in the "off" state.

AMPLIFIER DESIGN

Why is it necessary to amplify the signal? The line input of an MP3 player requires a peak signal level of approximately 1 V, which means 2 V peak-to-peak. This range is converted to 16-bit digital values in the device, which provides a resolution of 30 mV. However, the signal produced by the MD21 at a normal conversation level is only 3.6 mV, which is much too low for the analogue-to-digital converter (ADC) of the MP3 player. If you connect the microphone directly to the player, you will make a recording without any sound.

For a satisfactory SNR, the level of the signal applied to the input of the player must be at least 100 times as large as the value of the least significant bit, which means at least 3 mV. The signal from the microphone must be amplified by a factor of 833 to achieve this level. This is the job of the pre-amplifier.

The pre-amplifier described here provides a choice of three gains: 315 (50 dB), 1000 (60 dB), or 3150 (70 dB). Such high gains cannot be obtained in a single stage in combination with an acceptable bandwidth, so this task is split over two stages. The first stage is built using discrete transistors, while the second stage is implemented using an operational amplifier.

In the first stage (shaded grey in **Figure 2**), you can see the classic structure of a differential transistor amplifier (T1 and T2). The microphone signals arrive via IN1 and IN2, with the capacitors providing DC blocking to avoid upsetting the DC bias setting. The output of the first stage is taken across the collectors of T1 and T2.

The gain is determined by the equivalent resistance between the two emitters:

– 2.2 kΩ: 13.6 × (23 dB)
– 2.2 kΩ || 470 Ω: 43 × (33 dB)
– 2.2 kΩ || 33 Ω: 135 × (43 dB)

Transistors T3 and T4 stabilize the operating point of the stage, which makes it unnecessary to use a matched pair for T1 and T2.

WHY USE TRANSISTORS?

Could the first stage be built using an opamp and a few passive components instead of transistors? The answer is yes with regard to the gain, but no with regard to noise. This can be shown using a simple comparison. We compared the behavior of two amplifier designs with the same input sensitivity: the transistor stage described above (with S1 set to R12 for a gain of 33 dB) and an amplifier built using a TL071, which is known for its low noise level.

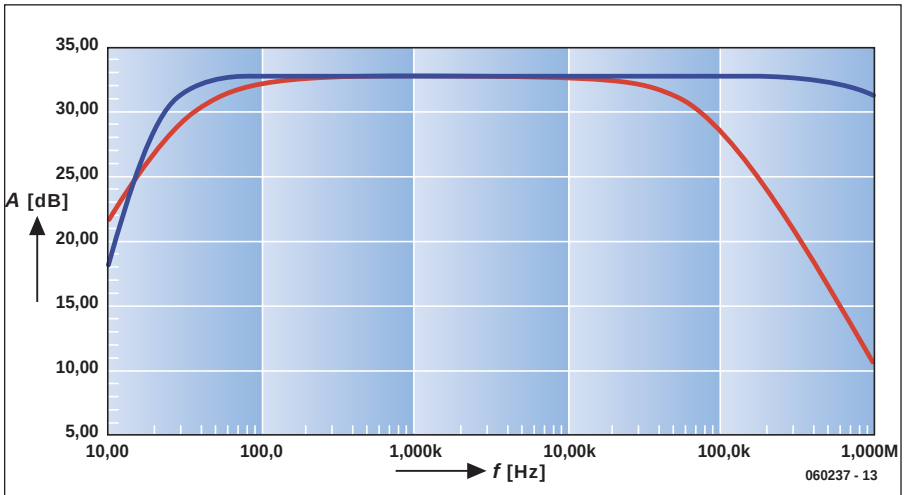


Figure 3: Bandwidth of the first stage: blue = transistor version; red = TL071 version

The frequency characteristics are shown in **Figure 3**. Both circuits meet the requirements, but the transistor stage does it better.

We can use a PSPICE simulation to determine the effective input noise voltage. This calculation takes all noise sources into account. It yields the following results:

- Design using a TL071: noise level $500 \times 10^{-18} V^2/Hz$ over the

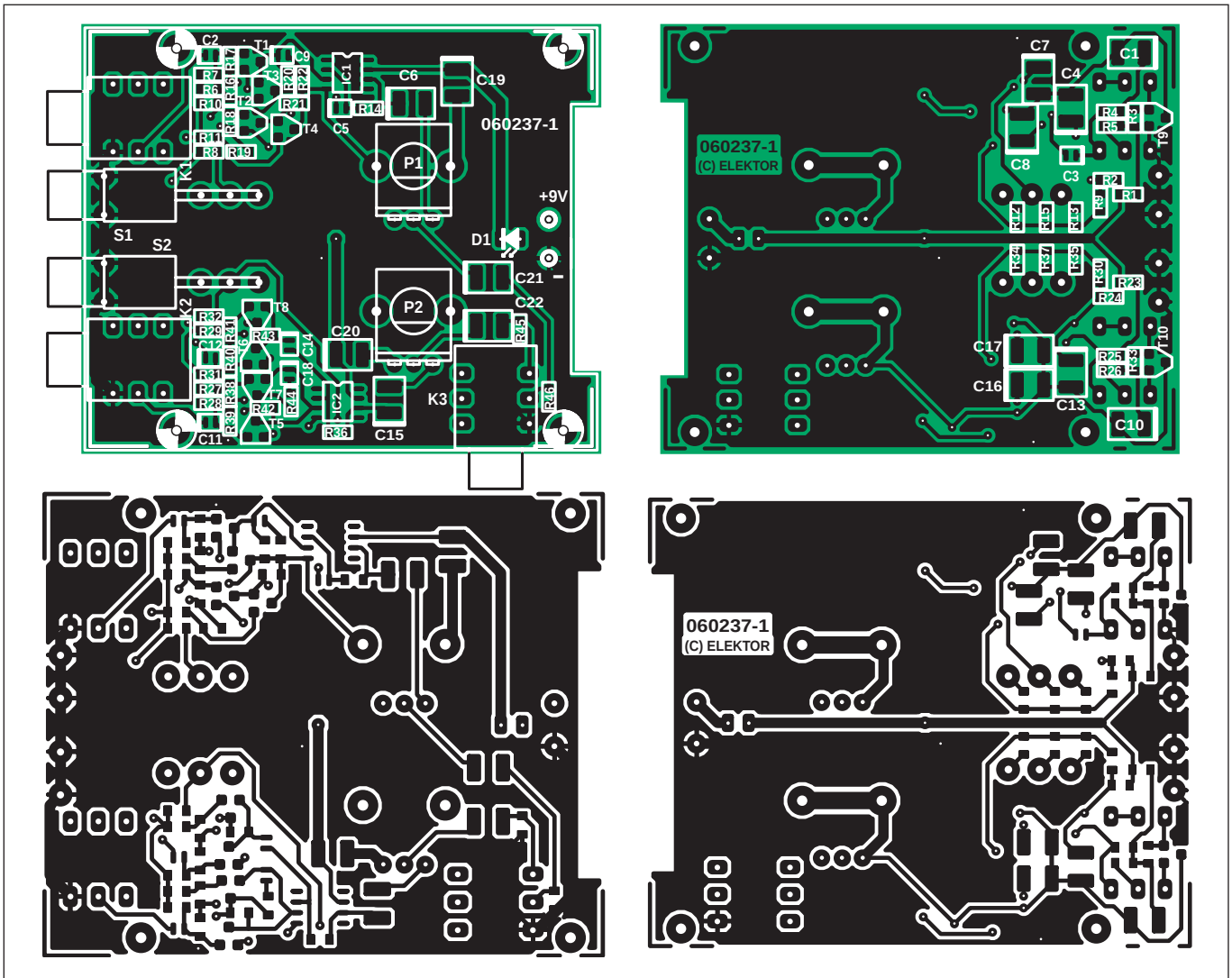


Figure 4: Layout of the double-sided PCB designed for this project

frequency range of 100 Hz to 20 kHz (white noise). This yields:

$$V_{eff} = \sqrt{[(500 \times 10^{-18}) \times (20{,}000)]} = 3.2\ \mu V$$

- Design using transistors: noise level $11.5 \times 10^{-18} V^2/Hz$ over the frequency range of 100 Hz to 20 kHz. This yields:

$$V_{eff} = \sqrt{[(11.5 \times 10^{-18}) \times (20{,}000)]} = 0.48\ \mu V$$

The noise level is a factor of 6.6 lower, so the transistor stage wins hands down. This noise level is of the same order as the signal produced by the microphone in a quiet room. This means you can hear a fly buzzing around in the room.

Strictly speaking, you also have to take into account the thermal noise generated by the internal resistance of the microphone. The MD21 has an internal resistance of 200 Ω, which yields an effective noise voltage of $8.6 \times 10^{-9} \times \sqrt{R} = 0.12\ \mu V$ (at 25 °C and a bandwidth of 20 kHz).

Due to the gain of the first stage, an opamp can be used for the second stage with satisfactory results. A TL081 can also be used in place of the TL071 without any appreciable degradation of performance.

The differential structure is maintained to avoid losing 6 dB in input sensitivity. The gain is fixed at 22 (27 dB) as determined by resistors R14, R20, R21 and R22 for the one channel or R42, R43, R36 and R44 for the other channel.

Potentiometers P1 and P2 can be used to independently adjust the output levels as desired. Capacitors C6 and C15 block the DC component to prevent crackling from the potentiometers.

CONSTRUCTION

All components except the battery are fitted to a printed circuit board. This is a double-sided, through-hole plated PCB. If you make the board yourself, you will have to make the through-hole connections by hand. The shape of the board is specifically designed to fit an enclosure distributed by Farnell with a compartment for a 9-V battery. We selected PCB-mount chassis connectors for the audio sockets to ensure optimum contact. Note: these are not standard versions.

We advise using a soldering iron with a very fine tip. Start by soldering the SMD parts, since they will be difficult to reach later due to the other components. Pay attention to the polarity of the electrolytic capacitors, which are present in generous numbers on the bottom of the board.

The SMDs are distributed over both sides. Handle the MOS transistors carefully to avoid exposing them to electrostatic charges. Be careful when soldering the resistors for switches S1 and S2 (R12, R13, R15, R34, R35, and R37): solder only the ends of these resistors that are not connected to the corresponding switches.

The other ends will be soldered when you fit the switches to the board. If you solder them before you fit the switches, the holes may be filled with solder.

Be sure to do a neat job of soldering, since many of the solder pads are close together. If you etch your own PCB, be sure to make the through connections between the top and bottom sides carefully and avoid making any shorts, which would be difficult to find later on.

ENCLOSURE

Once the circuit board is finished and you have checked that it works properly, you can fit it in the enclosure with the battery compartment. Make holes in the bottom part of

Acoustic pressures

Condition	P [Pa]	P [dBSPL]
Auditory threshold	20 µ	0
Quiet room	200 µ	20
Conversation	2 m	40
Lively conversation	20 m	60
Busy street	0.2	80
Pneumatic hammer	2	100
Pain threshold	20	120

the enclosure for the input and output connectors and the two switches. The top part must have holes for the two potentiometers and the LED, which adds a bit of flair to the top of the enclosure (a square or rectangular LED is a good choice here, with the leads trimmed to the right length).

PUTTING IT TO USE

After you're satisfied that the preamplifier is working properly, you can start using it. All you have to do is connect two microphones to the input connectors (K1 and K2) and connect the output to the line input of your MP3 player, and you're ready to go. Use a length of screened stereo cable for the connection to the MP3 player.

Here's to good recordings! *aX*

Parts & Components

Capacitors (all SMD)
- C1,C4,C6,C7,C8,C10,C13,C15,C16, C17, C19-C22 = 10 µF 16 V tantalum, Vishay Sprague 595D106X9016B2T (Farnell # 392-8895)
- C2,C3,C5,C9,C11,C12,C14,C18 = 470 nF/25 V tantalum, Vishay Sprague 595D474X9025T2T (Farnell # 116-6814)

Miscellaneous
- K1,K2,K3 = 3.5 mm jack socket with 3 contacts, Schurter R36310000 (Farnell # 149-933)
- S1,S2 = SPST switch, angled pins, PCB mount, height = 12.7 mm, C&K 7103MD9 AV2BE (Farnell # 957-5510)
- BT1 = 9-V battery with clip-on lead
- K4 = 2-way PCB pinheader
- Case with battery compartment, e.g. BC2, Farnell # 223-554
- PCB, ref. 060237-I from ThePCBShop

Resistors
- R1,R12,R23,R34 = 470 Ω
- R2,R24 = 100 Ω
- R3,R33 = 4 MΩ7
- R4,R5,R25,R26 = 3 kΩ3
- R6,R11,R27,R32 = 6 kΩ8
- R7,R10,R28,R31,R45,R46 = 47 kΩ
- R8,R9,R29,R30 = 2 kΩ
- R13,R35 = 33 Ω
- R14,R22,R36,R44 = 220 kΩ
- R15,R37 = 2 kΩ2
- R16-R19,R38-R41 = 22 Ω
- R20,R21,R42,R43 = 10k Ω
- P1,P2 = potentiometer 10 kΩ, B, Alps type RK09K11310KB (Farnell # 119-1725)

Semiconductors
- D1 = LED, 3 mm, red
- T1-T8 = BC856ALT1G, SMD, SOT-23, ON Semiconductor (Farnell # 114-7782)
- T9,T10 = 2N7002, MOSFET, SMD, SOIC, Texas Instruments (Farnell # 102-1754)

IC1,IC2 = TL071CD, SMD, SOIC, Texas Instruments (Farnell # 958-9449)

MEASURED RESULTS

Due to the very low input levels and the fact that it was not possible to produce the necessary levels directly using the available signal generators, we made an attenuator cable as shown in **Figure 5.**

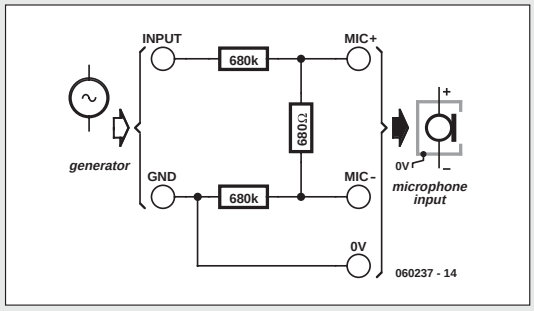


Figure 5: Electrical diagram of the attenuator cable

Characteristics
Attenuation: 2000 (66 dB)
Output impedance: 680 Ω (nearly the same as a dynamic microphone)
Preamplifier operating in differential mode

Results
The three curves correspond to the three settings of switch S1. Measured results:

Setting 1: Maximum gain = 1 dB + 66 dB = 67 dB. This is nearly the design value of 70 dB. Lower –3-dB point: 50 Hz
Setting 2: Maximum gain = –8 dB + 66 dB = 58 dB. This is nearly the design value of 60 dB. Lower –3-dB point: 50 Hz
Setting 3: Maximum gain = –18 dB + 66 dB = 48 dB. This is nearly the design value of 50 dB. Lower –3 dB point: 50 Hz
The upper –3 dB point is above 100 kHz (–2 dB at 100 kHz).

Noise and distortion
The noise and distortion figures were measured by the author. The results we obtained with our own analyser (Audio Precision) are nearly the same.

Results

Switch S1	S_{gen} [dBV] at 1 kHz	S_{out} [dBV]	Measured gain	S/N	THD	Output noise level	Input noise level	
Gain max	-1	0	67 dB	57 dB	0.12%	-57 dB	-124 dBV	0.63 μV_{eff}
Gain moyen	+8	0	58 dB	65 dB	0.12%	-65 dB	-123 dBV	0.71 μV_{eff}
Gain mini	+18	0	48 dB	72 dB	0.12%	-72 dB	-120 dBV	1 μV_{eff}

Distortion
0.12% at V_{out} = 0 dBV0.04% at V_{out} = 10 dBV

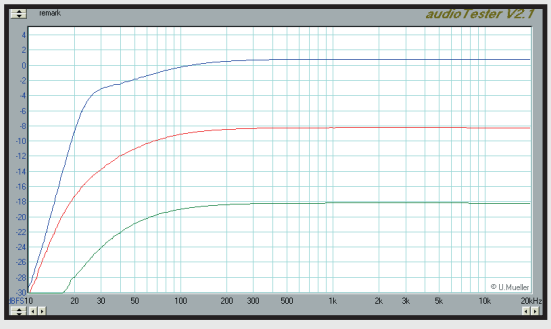


Figure 6: The results measured by the author are very close to our measured results.

By Ed Simon (United States)

Is There DC on Your AC?

The possiblility of receiving DC on an AC power line is explored

There are some audio design issues that, at first look, just don't seem to be reasonable. For example, is it possible to get DC on an AC power line? I decided to try a simple circuit shown in **Figure 1** to see.

The way it works is that the positive going peak of the incoming AC power line will charge just one of the capacitors. The negative going peak will charge the other one. If both peaks are the same, there will be no difference between the output terminals. If for some reason, one peak is not the same then there will be a voltage! I used a leftover Microtran PSS4-48 transformer for this.

Of course, any transformer with dual secondary windings will work. Calibration is easy. Plug it in and read the difference between the outputs. Flip the AC plug over and adjust the trimmer resistor until it reads the same absolute value both ways. You might want to play with the values of both the fixed and trimmer resistors to optimize this circuit for your exact transformer.

My small transformer, as lightly loaded as it is, read 60 VAC out so a scale correct factor of two times the meter reading would be the equivalent DC asymmetry on the power line.

Some may wonder how such a problem could occur. Refer to **Figure 2**. Here is a typical branch circuit fed from a transformer and feeding three light bulbs. The third light bulb has an old-fashioned dimmer switch. In one position, the lamp is directly connected to the AC line; in the other, a diode is placed in the series to dim the lamp. Although this is an extreme case, it does actually occur. More often there are devices where the load is not perfectly symmetric, these could be regular light dimmers where the active semiconductor does not fire at exactly the same place on both the positive and negative slopes of the AC power line or even a sum of small loads that do convert AC to DC with unequal loads.

To find out what can happen from DC on AC, I deliberately set up an extension cord with a 60-W light bulb on a diode dimmer. It turned out there was an offset of 0.88 V on the AC power line.

Would 0.88 V of "DC" actually have an effect on anything? There was an easy way to find out. I hooked up a similar power transformer but with each winding loaded by 1,000-Ω resistors. I powered it from my not so good AC line and measured the output voltage.

On the first day I measured a bit more often so the results are shown in **Table 1**.

As you can see, there is a small but steady loss of transformer efficiency until the end. The end resulted from a storm knocking out power for a day. The transformer recovered much of the lost efficiency but not all of it.

So the conclusion is that surprisingly enough you can get "DC" on an AC line, and it will slowly have an effect on the equipment powered by that line! *aX*

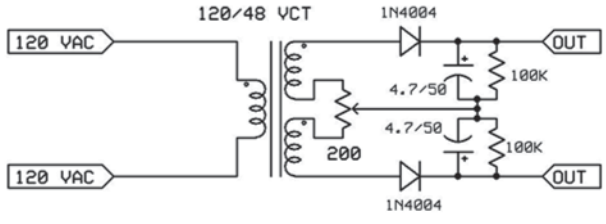


Figure 1: Circuit to check for DC on AC lines

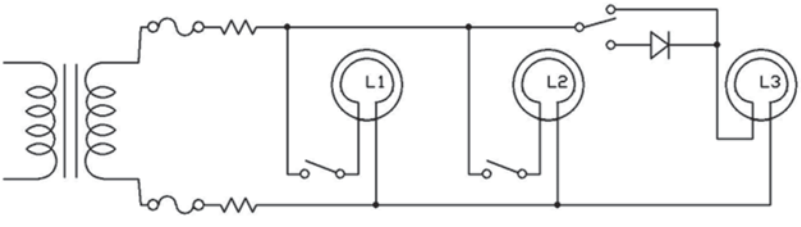


Figure 2: Model of AC power branch circuit

AC LINE	OUT	RATIO	LOSS	% LOSS
125.2	60.0	0.4792	0.0000	0.00
125.5	60.0	0.4781	0.0011	0.24
126.6	60.5	0.4779	0.0014	0.28
125.6	60.0	0.4777	0.0015	0.32
123.9	59.1	0.4770	0.0022	0.47
125.2	59.6	0.4760	0.0032	0.67
125.1	59.7	0.4772	0.0020	0.42
125.4	60.0	0.4785	0.0008	0.16

Table 1: Transformer efficiency with DC on AC

By Patrick Brunner (United States)

Phase Linear 400 Amp Upgrade

A guide for PL 400 owners to repair/upgrade their units

In a previous *audioXpress* article, ("Upgrading the Phase Linear 200 Amp," October 2011, p. 26), I detailed how to improve the performance of your PL 200. This article addresses its big brother, the Phase Linear 400.

I refer you to an excellent guide on the repair of power amplifiers at www.synthetizer-sche.chez-alice.fr/power%20ampli/phase%20linear/repair.htm. Please read this article carefully for many good pointers and safety precautions.

POWER SUPPLY

My amplifier had a shorted power supply capacitor, so the capacitors were obvious candidates for upgrade. The original 5900 µF/85 VDC capacitors are compact, measuring 1.75" diameter and 4.65" long. They are either hard to find, too expensive, or both. These caps were held in place with double-sided foam tape and rested in curved indentations in the cast alloy transformer end cap.

My choice for replacements were 10,000 µF/100 VDC with a 2" diameter, and 3.70" long (see **Photo 1**). They were a great find on eBay, except they hit the back of the left VU meter. The solution is to remove the transformer's 3.75" × 1/4-20 mounting bolts and replace them with 4" × 1/4-20 bolts inserted through the chassis first. Then, add four (± depending on your hardware) SR lock washers between the transformer and the chassis in order to raise the transformer 0.25" off the chassis. Now, use the original nuts and washers on the outer side of the transformer. This gives you maximum usable space inside the chassis (see **Photo 2**).

The bottom of the capacitor indentations should be nearly even with the inside of the chassis. Remove all of the foam tape remnants. Now there is enough space to mount the capacitors and clear the meter.

With this modification you can use 2" diameter capacitors up to about 4.65" long. Be sure to clean the capacitor indentations and capacitors with a solvent (acetone) to ensure maximum adhesion of the new double-sided foam tape (see **Photo 3**). I used longer-than-needed pieces and trimmed them with a razor knife to match the transformer end cap outline.

Note: Leave the protective paper on the foam tape before finally securing the capacitors so you can rotate the capacitors to accommodate the copper ground bar. Install the ground bar with the screws slightly loose to help ensure this and be very sure to observe polarity. When you get the rotation right, remove the protective paper and press the capacitors firmly into place. Then connect the wires as needed and fully tighten the screws.

Suggestion: Replace the 25 A 200 PIV bridge rectifier with a 35 A 600 PIV and add a little thermal grease under it to improve the heatsinking to the transformer end cap (see **Photo 4**). The bigger capacitors will have higher inrush currents.

DRIVER & OUTPUT TRANSISTORS

The old original devices were low gain, typically rated at 5 A 100 W, good for 40 years ago. Replacement parts are 16 A 250 W, such as the ON Semiconductor MJ12024, which is a good device, but you can do even better! The ST Microelectronics 2ST5949 has virtually identical SOA ratings, 1 A higher

Photo 1: Old and new caps

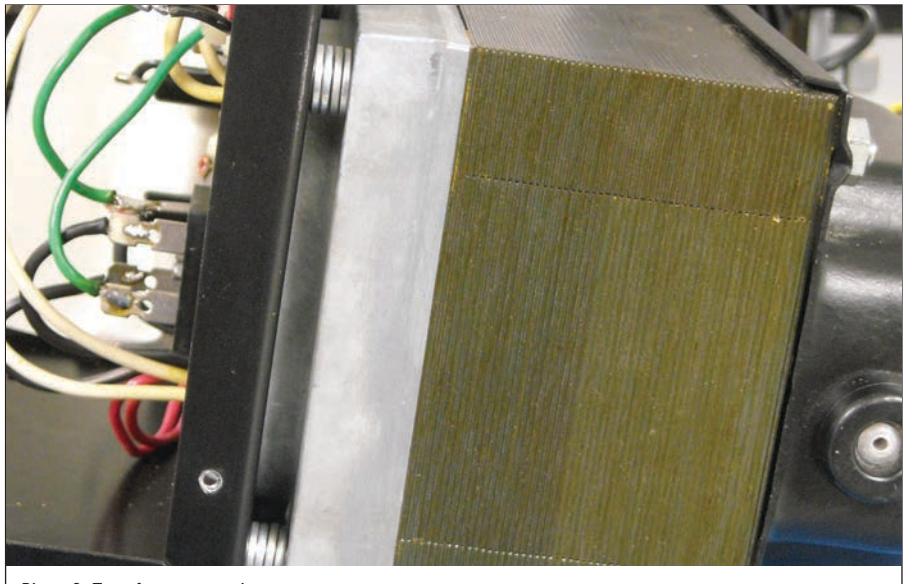


Photo 2: Transformer mounting

current rating, a Ft (bandwidth product or transition frequency) of 25 MHz instead of 4 MHz, and more important, a much higher and much more linear Hfe curve—(gain or Beta), 80 vs. 15 into the 7 to 8 A range. Therefore, the 2ST5949 also serves well as the driver. And, it is usually cheaper! The higher gain also lessens the demands made on the driver board.

Choose either the mica washer and grease or silicon rubber insulators, because there is debate as to which is better. This upgrade may require adjusting the value of C7 (on a PL14A board) for HF stability. You probably will also need to increase the fuse values, if you push the amplifier to its new limits, or use lower impedance speakers (which it can now handle easily). I went with 8 A on the DC supply fuses and 10 A on the AC line fuse. Note: There is a PNP complement (2ST2121) available if your amp needs it.

Make the heatsinks a little more efficient. Because you have removed the transistors in the previous step, it only requires removing two more screws to take the heatsinks off as well. When you reassemble the heatsinks, add heatsink compound between the chassis and heatsinks' surfaces to improve the overall thermal conduction (see **Photo 5**).

DRIVER BOARD

Unless your amplifier has been recently "gone through," replace all the electrolytic capacitors on the driver board. It is very cheap insurance when dealing with amps in this power class. Use high-grade 105°C components. If your driver board has transistor problems, upgrade the transistors, too.

I used ZTX694Bs for the input differential pair and ZTX796As for the second differential pair (deleted on PL400C boards). Buy at least 10 of each type (they are not very expensive) and match them as closely as possible in pairs of equal gain. (A cheap DMM with transistor test is sufficient.) This will help minimize the offset of the amp. I typically see much less than 10 mV of DC offset after this upgrade. Don't forget to measure the resistor pairs (R4, R7, R9, and R12 on a PL14A) in the differential amplifiers for a close match. If they are off by very much, replace them with 1% parts selected for very close matches.

Last, be sure to check the TO39 transistors' heatsinks for a tight fit. If loose, just pinch each of the fins with a pair of needle-nose pliers to tighten and reinstall them. Other possible transistor substitutions: Q3, Q10—MJE15032, Q7—MJE15033. These are TO220 parts and will require a clip-on heatsink. Note: Different numbered transistors may have different pinouts. Check before you solder.

Improve connection reliability on the thermal feedback transistors by securing the leads with either epoxy or adhesive mounted wire clamps. If these leads break, the amplifier will usually be damaged. If you already have broken leads, consider a 2SC6081 (or similar) as a substitute. This is a fully-insulated TO220 part. Solder and heat-shrink the leads to prevent lead breakage. A spot of heatsink compound between the device and chassis is advisable as well for either the old part or a new one.

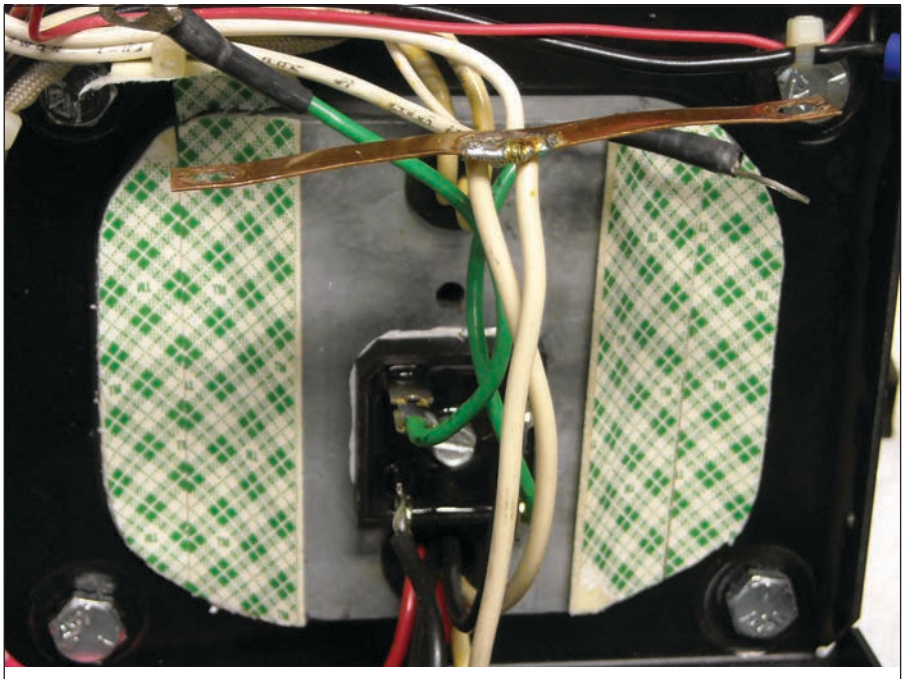


Photo 3: Mounting tape applied to transformer

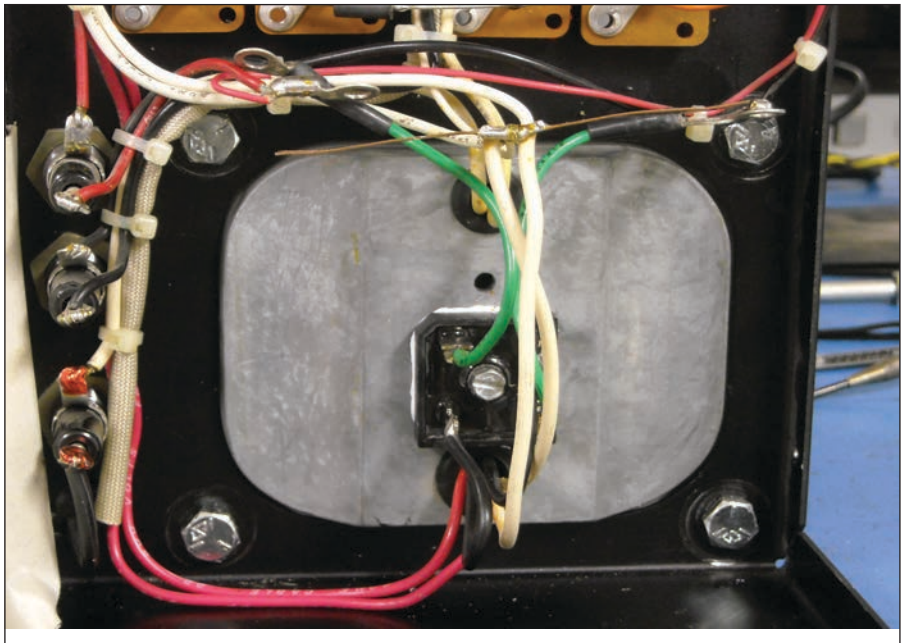


Photo 4: New bridge installed

Replace the old binding posts with new heavier duty gold-plated ones (eBay). In my case the required mounting holes are a bit larger, approximately 0.465" in diameter. I used a step drill to enlarge them to 7/16" and then a hand-tapered reamer to get them the exact size.

Replace the old phono jacks with new gold-plated ones (eBay). Of course, these used smaller mounting holes. I used the supplied flat white insulating washer (Teflon, nylon?) on the outside and two 1/16" long pieces of heat shrink (one on top of the other) on the threaded shaft in order to build up diameter so I could use the original fiber shouldered washer on the inside of the case. I did this in order to keep the jacks centered in the too-large original holes. While you are at it, you may choose to replace the twisted wire input leads with a short length of shielded cable.

You may also consider replacing the two AGX fuse holders with AGC fuse holders so you can use the more commonly available AGC-size fuses. If you do, be sure to use AGC Fast Acting types for equal protection.

METERS

Install blue (or the color of your choice) LEDs for the VU meters (see **Photo 6**). First, make servicing easier by installing a five- or six-contact plug and jack pair so you can easily remove the front panel from the chassis for work. I like AMP MR types, but any type will do. By far the easiest LED option is to buy a $30 "White Oak Audio Design" LED kit on eBay. Another option is to modify the existing board for much less cost.

In either case, the stock LEDs project a narrow, focused beam. To get a more attractive, diffused beam, simply grind the lens end flat. If you want even more diffusion, paint the flat end with a thin coat of white spray paint or fingernail polish.

For the DIY route you will need:

- 8 - Blue (or the color of your choice) T1-3/4 (5 mm) LEDs, eBay or Digikey.com
- 8 – 180 Ω SMT 0603/0805/1206 or leaded resistors, Digikey.com
- 2 – 100 µF/25 VDC electrolytic

Photo 5: Grease applied to heatsinks

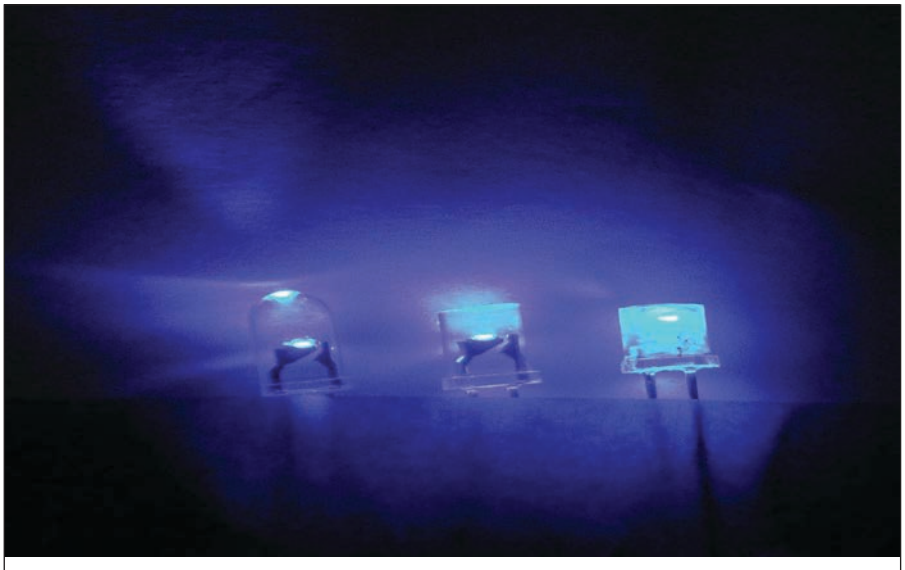


Photo 6: LEDs (left to right), standard, flat, flat plus paint

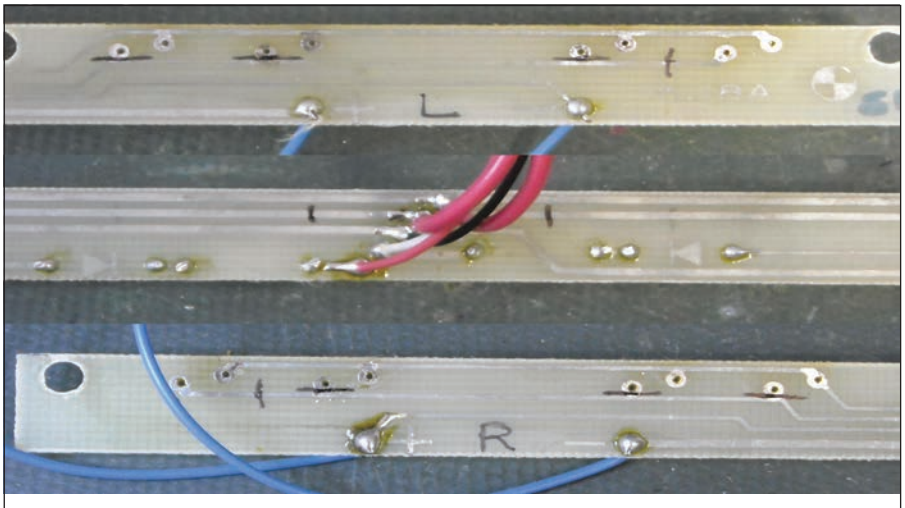


Photo 7: LED board cuts for resistors additions

capacitors, Digikey.com
- 2 – 1N4001-7 diodes – PIV not critical, Digikey.com

Unsolder the lamps, clean off excess solder and flux, and cut the traces with an X-acto knife or Dremel tool where indicated (see **Photo 7**). Cut only about a 0.020" wide gap in the traces, keeping as much pad and trace area as possible. First, solder the two diodes and two capacitors onto the traces. Then install one resistor and one LED to see whether the brightness is to your liking (see **Photo 8**). You may need to vary the resistors' value depending on the type (color, forward voltage drop) of LED. Then install all the remaining resistors (of the selected value) and LEDs. Make the LEDs' leads about 3/8" long, bent about 45°, and try the fit before soldering the LEDs and cutting the leads. See the schematic in **Figure 1**.

Note: Measure the voltage across the resistor and calculate the current so it is less than 20 mA. I thought 20 mA on the blue LEDs looked too bright.

Caution: The old single-sided board's traces tend to lift off very easily. Once you have all the parts mounted, cleaned, and tested, coat the circuit side of the board with five-minute epoxy to keep things from breaking in the future. If you have the meter lighting PCB off, you may wish to remove the meters, spray-paint the meter bezels, and clean the meters and the front panel, which would now be free of obstructions.

Note: The original too-long meter retaining screws usually get bent and "wander" on the back of the front panel as they are tightened. The solution is to use 1" or 1.25" long 6-32 screws with the ends ground to a sharp point. Position the meters for re-installation and see where the screws will hit the panel and mark it. Remove the meters and bezels and then either drill or punch a small shallow indentation for the screw point to set in. Take care not to scratch or drill through the front of the panel while doing this.

I have not found many of the aluminum front panel bolts in decent condition. The solution is to buy some 1/4-20 × 1/2" stainless steel bolts and nuts. File and then sand (use 240–400 grit silicon carbide paper) the heads of the bolts smooth and polish them with jeweler's rouge on a power buffer to a mirror finish. They look nice and won't corrode over time.

Last, bring the unit up with great care (heed the suggestions found in the second paragraph's link).

Warning: With these modifications and consequent increased power capability, a loose input cable or plugging in a cable while the amplifier is powered on can deliver a speaker-destroying transient. Expect 300 W into 8 Ω, and possibly more into 4 Ω. Add in line-speaker fuses to be safe. *aX*

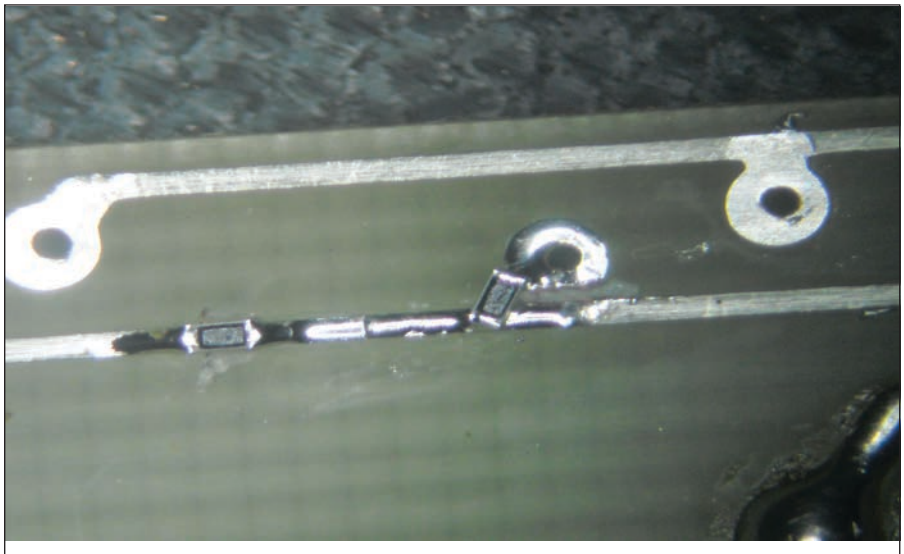


Photo 8: Installing the resistors

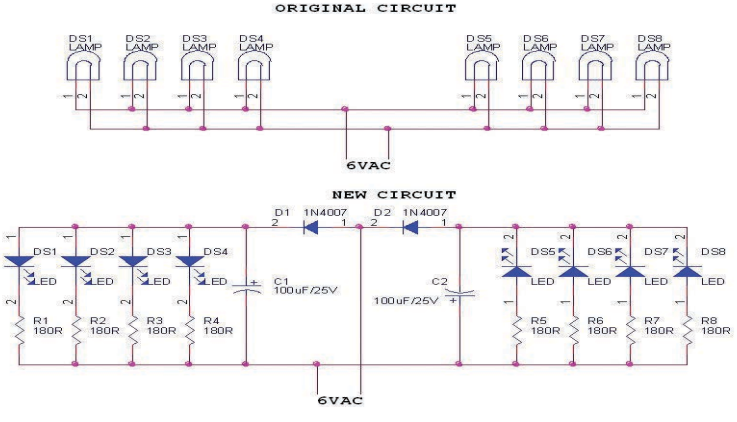
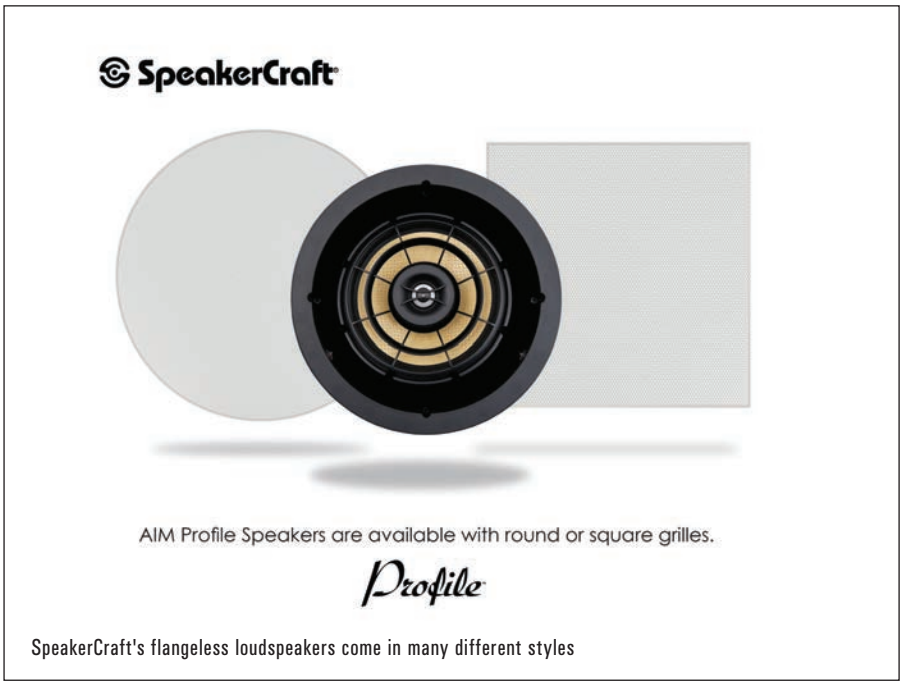


Figure 1: Old/new lamp/LED schematic



SpeakerCraft's flangeless loudspeakers come in many different styles

FLANGELESS LOUDSPEAKERS

SpeakerCraft's line of 43 new flangeless loudspeakers is meant for both in-wall and in-ceiling applications.

The Profile series takes advantage of audio technology made famous by the AIM line, allowing you to direct the sound toward the listening area. The new AIM Profile grilles come in both square and round types allowing for a round speaker to have the option of a square or round grille.

Using neodymium magnets to attach the grille, the speakers install in just seconds. There's no bezel or visible mounting flange. All you see is a grille, which can be painted to match the wall or ceiling. The system design allows dealers to reduce stocking multiple speaker shapes, fit, and finishes.

The Profile series saves a great amount of time in the painting process since the grille can be easily removed to paint and there is no flange to paint on the speaker itself.

Go to SpeakerCraft's website (www.speakercraft.com) for more information and pricing.

WIRELESS AUDIO STREAMER

The nRF2460 is a low-cost mono audio-streamer IC that delivers uncompressed 16-bit PCM digital audio, with audio latency as low as 22 ms, and a battery-operating, lifetime of up to 100 hours from two AAA prime cells.

The nRF2460 combines a class-leading ultra-low power (ULP) 2.4 GHz radio with an optimized adaptive frequency hopping (AFH) protocol and audio Quality of Service (QoS) engine. The audio-streamer is available in a compact 6 × 6 mm QFN package and can operate with a single low cost 16 MHz ±50 ppm external crystal. The I2S, SPI, and I2C interfaces provide connections with external audio ADCs, DACs, and microcontrollers.

The AFH protocol and audio QoS engine enable the nRF2460 to maintain robust wireless audio co-existence, even in challenging RF environments that include other common 2.4 GHz interference sources such as Wi-Fi and Bluetooth wireless technology. The nRF2460 utilizes smart retransmission of lost packets and the audible effects of lost data are minimal. The chip's 4 Mbps bandwidth provides ample overhead to support an auxiliary wireless data channel for user control functions such as mute and volume up/down.

The nRF2460 was developed in close collaboration with premier digital audio converter specialist AKM Semiconductor for cost-sensitive battery-powered mono (single-channel) wireless audio applications such as microphones, subwoofers, musical instruments, and gaming. Visit Nordic Semiconductor's website (www.nordicsemi.com) for more information.

ON-WALL LCR SPEAKER

Paradigm Electronics's Millenia LP on-wall LCR speaker has an ultra-thin (1.5") cabinet profile. It is slim enough to hang with the thinnest LED, OLED, LCD, or plasma flat-panel displays, even meshing with projector screens while reproducing music, soundtracks, and dialogue with accurate ease.

Lifestyle speakers are in demand and the Millenia LP caters to audio enthusiasts building a system piece-by-piece since each speaker can be used for left-, right-, or center-channel sound, just by configuring vertically or horizontally.

The Millenia LP LCR also suits those in search of non-architectural, minimalistic audio options since it requires no wall removal to fit an in-wall speaker. Simply cut small speaker wire holes and mount the speakers to the wall for a space saving LCR set up.

Built with Paradigm Reference technology, the Millenia LP packs convex S-PAL satin-anodized pure

The Millenia's ultra-thin cabinet profile from Paradigm Electronics

aluminum domes, curved mineral-filled polypropylene bass/midrange cones and other award-winning design elements into its slim frame. The infusion of mica adds stiffness and the injection-molding ensures critical shape so the cone stays true to form. Proprietary elastomer surrounds ensure a high-strength bond, even when the cone's being given a high-intensity workout.

How does a thin speaker handle bass? Dual high-velocity passive radiators are used instead of ports to attain the desired bass extension in the minimal internal volume design. Solid and stable, the mica-loaded polypropylene ensures clean, clear and articulate bass, free from audible vibrations and resonances. For bigger bass, three Reference Millenia LPs can be paired with the sonically-matched Millenia ADP surrounds and a MilleniaSub or Seismic 110 to create an acoustically immersive and aesthetically pleasing 5.1 surround sound set-up. For more information, go to www.paradigm.com.

MEAUREMENT INSTRUMENTS

Scantek, a distributor for multiple sound and vibration lines, offers two of the newest instruments from Soft dB, the "Piccolo" and the "Concerto." The former is a compact, data logging integrating sound level meter with up to 10 days of continuous measurement and many other features.

The introductory price is $345.00, which includes a windscreen, USB cable, battery, software, and case.

The "Concerto" is a sophisticated four-channel, Class 1, hand-held acoustic data-logging measurement system with up to 16 GB of internal memory. The four independent channels can do building acoustics, building vibrations, and soon, sound intensity.

The unit starts at about $10,000. Go to www.scantekinc.com for more information.

Scantek's "Concerto," a hand-held acoustic data-logging instrument

A multi-functional audio signal processor from Saelig

AUDIO SIGNAL PROCESSOR

Signal Wizard 3.0 is a powerful audio signal processor that features multichannel synchronous processing. It can mix, amplify, filter, delay, and adjust the phase of individual input signals, selected by using the included intuitive PC software.

Signal Wizard 3.0 features a 24-bit, 96-kHz codec with six analog input and eight analog output channels, and an internal DSP processing speed of 0.6 GMACs. Signal Wizard 3.0 also incorporates two digital audio (S/PDIF) inputs and outputs. Like its two channel equivalent Signal Wizard 2.5, the software requires no knowledge of mathematics or programming.

Signal Wizard 3.0 includes powerful mixer functions. Any channel can be blended with any or all of the other channels in any proportion, since the system incorporates mixer units at the input and output signal stages. Signal Wizard 3.0's unique filter design engine enables standard filter types to be created using the easy-to-use graphical software, but it also allows completely arbitrary frequency responses, both in amplitude and phase, to be realized via a simple text file import.

The Signal Wizard 3.0 design features six analog and two digital (S/PDIF) audio input channels, a first stage flexible 8 × 8 mixer unit, a multichannel gain stage, a multichannel filter/phase stage, a multichannel delay, a final stage flexible 8 × 8 mixer unit, and eight analog and two digital audio output channels. Sample rates are selectable from 96 kHz down to 3 kHz. Also included are USB, parallel, and JTAG interfaces, enabling Signal Wizard 3.0 to be programmed in native DSP assembly code or high level languages such as C++ using third-party compilers.

Performing up to 588 million multiplications and additions per second, Signal Wizard 3.0's high-quality analog signal conditioning with stereo 24-bit resolution is sufficient for the most demanding applications. Signal Wizard 3.0 brings the power of digital signal processing to any audio-bandwidth domain that requires electronic signal filtering.

Originally developed for teaching environments covering DSP principles and practical applications, Signal Wizard products are suitable for a wider range of engineering uses. Applications include: audio and sensor signal processing and conditioning, signal analysis, vibration analysis, education and research in electrical, electronic and other physical sciences, multichannel filtering, crossover networks, surround sound processing, adaptive filtering, inverse filters, echo, real-time gain control, mixing, phase delays, time delays, sound focusing and beam forming, real-time FFT and waveform capture, Hilbert transform, quadrature signal processing, envelope detection, etc.

Signal Wizard 3.0 is designed and developed by Signal Wizard Systems at the University of Manchester (UK) and is available through Saelig Company, Inc. (www.saelig.com).

SOUNDWARE XS SE

Boston Acoustics has a new addition to its SoundWare family—the SoundWare XS Special Edition Home Theater Speakers, which are available in both gloss-black and gloss-white finish. The package features the smallest loudspeakers Boston Acoustics has ever produced, which makes them ideal for consumers who want to enjoy powerful, room-filling sound in a stylish, ultra-compact system without the traditional box-type loudspeakers.

The system includes five ultra-compact SoundWare XS Special Edition satellites, each measuring less than 4", plus a compact 8" 100-W (250-W Peak) powered subwoofer. Notably, for users who want an even more immersive listening experience, Boston Acoustics is making optional SoundWare XS Special Edition satellite speakers available to create a 7.1 system. The system is designed for space-conscious and style-driven consumers who want to enjoy great sounding music or powerful home theater audio.

Compact satellites provide the rich audio power and performance of true two-way speakers, with 2.5" (65 mm) woofers and 0.5" (13 mm) tweeters. The result is astoundingly accurate and pure sounding music, plus room-filling home theater audio.

For more information, visit www.bostonacoustics.com.

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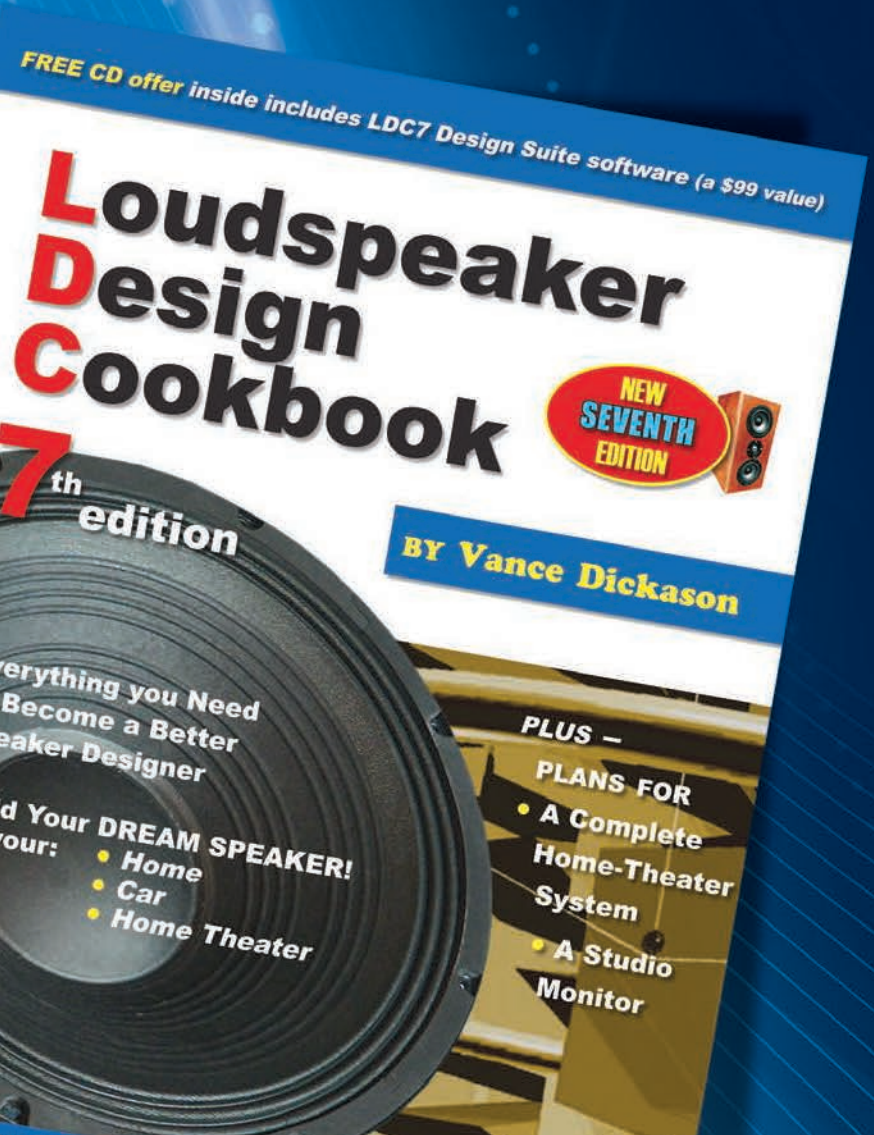
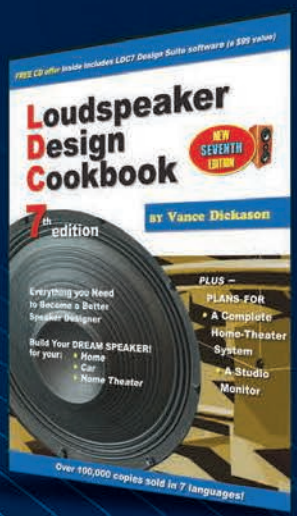
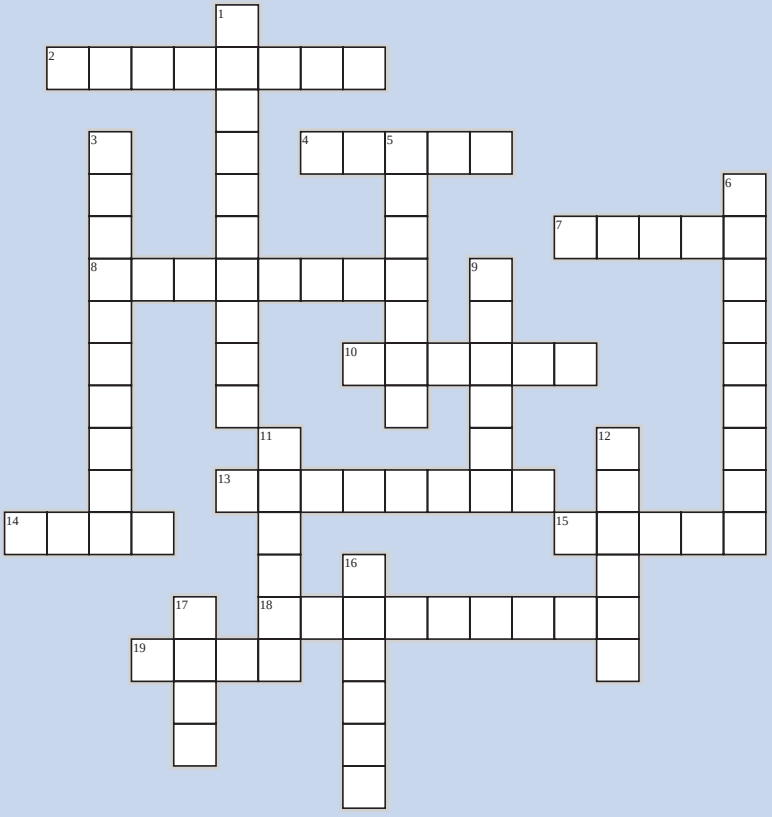




Editor's Note: Answers to this puzzle will appear in the next issue.



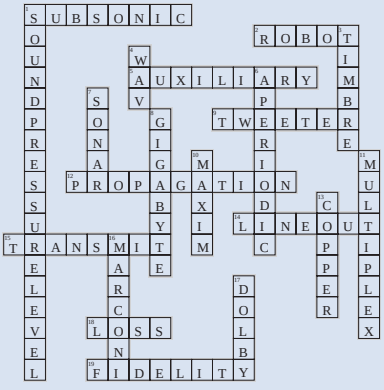
ACROSS

2. German-American inventor (1851–1929) who received a patent for the disc record gramophone in 1887
4. Metric prefix for 0.000001
7. A method for detecting objects, especially those under water, with sound waves
8. An electronic device that receives and amplifies a weak signal before retransmitting it
10. Constant signal processing
13. A component on a satellite dish that captures a reflected signal and directs it into an amplifier
14. Line that transmits an RF signal to an antenna
15. Electrical connection on a PCB that carries an electronic signal.
18. Deep, low female voice
19. A directional TV or radio antenna popular among amateur radio operators. A named after a Japanese electrical engineer (1886–1976), the antenna enables directional communication using electromagnetic waves.

DOWN

1. 10^{-3} Hz
3. Bob Dylan's transducer
5. Eddy
6. Transmit sound waves
9. Can block or allow to pass
11. For more than a century, Alexander Graham Bell was credited with inventing the telephone. But on June 11, 2002, the U.S. House of Representatives (House Resolution 269) recognized this Italian engineer (1808–1896) as the phone's inventor.
12. The "T" in T = 1/f
16. A sudden input current surge
17. To cover unwanted sound

Answers to the October 2011 Puzzle



BEST WISHES

I would like to express my gratitude and all my best wishes to Mr. Ed Dell for the goodwill he showed a few years ago, accepting my manuscripts to be hosted in *audioXpress*'s pages. I wish I had found him and his publications many years ago, when I was trying to build my first tube amplifiers in my poor lab in communist Romania. Fortunately, though, these most creative years of my life as a musician and tube amplifier designer led me to Athens, Greece, where his publications and work were available.

As with all the guys of my generation, the well-known '60s generation, I remain a tube lover, and the contribution of American pioneers in this area was tremendous.

I have learned many things due to the retro collection of *Glass Audio*, *Elektor*, *Audio Craft*, *Audio Anthology*, and, of course, the *audioXpress* series. As a former guitar player, I could order many books regarding musical instruments, amplifiers, and also books from prestigious authors.

I thank Ed and his staff for all the guidance, knowledge, and experience the publications brought to my life, and for so much more. I wish Ed good luck in whatever else he might get involved. Also, I hope *audioXpress*'s new owner, the Elector group, will build on your good work and values. Best regards.

Alexander Arion
Athens, Greece

LEGACY AUDIO REVIEW

Karl Nehring did a great job with his review of the Legacy Audio Focus SE speaker ("Legacy Audio Focus SE Speaker," p. 24, *audioXpress*, September 2011). I would like to add my perspective on why Legacy continues to turn out speakers that are such winners. I have known Bill Dudleston for many years beginning with some of the meetings that he hosted for the Prairie State Audio Construction Society.

In my opinion, Bill is one of the few designers who "gets it." There is so much snake oil in the audio business, it is refreshing to know someone who understands both the physics of how sound is reproduced and how we react to music. Bill also does very high-quality recordings so his expertise extends to both ends of the reproduced musical process. Whether he is talking about volume velocity issues or polar response impact on perceived sounds, Bill not only understands what is going on but can communicate to people at all levels. And, he is a real expert at separating the wheat from the chaff and turning that knowledge into very high-performance speakers at all price points, resulting in great values.

So I would like to thank Bill not only for the great products, but also for helping me immensely in my journey all these years through my audio hobby. It has always been a great pleasure to deal with him on both a technical and personal level. I wish there were more like him.

Tom Perazella
North Potomac, MD

Tell Us What You Think!

We want our readers to take an active interest in *audioXpress*. If you have comments, suggestions, or questions, send an e-mail to sbecker@audioxpress.com.
Letters to the editor may be published in the XPRESSMail section of an upcoming issue.

Contributors

Patrick Brunner ("Phase Linear 400 Amp Upgrade," p. 36) is an independent electronic design consultant. He started Applied Technical Services in 1982, designing primarily in the field of medical electronics. Prior to 1982, he worked at various military, industrial, and medical electronics companies. Patrick resides in Florida. The electronics bug first bit him at age 8 with a crystal radio and a CK722 for an amplifier. The bug was exacerbated later after hearing a Heath kit W9 driving a homemade 15" JBL-based speaker system. Hi-Fi has remained a hobby since that time.

Carl Casper ("The Casper PC Audio Amplifier," p. 11) became interested in electronics as a child and has worked in the electronics field for 40 years. He has a degree in electrical engineering. He has worked as an electrical engineer and electronic engineer, designing signal conditioning circuitry for gas sensors, electronic product testing, and calibration engineering, using an automated precision voltage divider/amplifier system that he designed. He is currently employed at an electronic controls repair facility, troubleshooting electronic controls and PLC's to the component level. In his free time, he designs and builds audio and test equipment.

Marcel Cremmel ("MP3 Pre-Stereo Microphone," p. 28) first contributed his article to *Elektor*. It appeared in the February 2007 issue of Elektor and turned out to be very popular at the time. Marcel is one of countless people who have contributed projects to *Elektor*.

Wim de Haan ("The Abletec Class D Amplifier," p. 22) is a medical engineer with a degree in electronics and a great interest in DIY audio. He started his first audio project at age 14 and got his first Keith Jarrett CD as a present at age 16. Some of his projects have been published in *Electronics World + WW, Elektor,* and *audioXpress*. He is webmaster of the official websites of classical pianist Mikhail Voskresensky and pianist Evelina Vorontsova. His adoration for music is apparent in the websites, www.tatiana-nikolayeva.info and www.robmadna.info.

Vance Dickason ("Scan-Speak Discover Drivers," p. 15) has been working as a professional in the loudspeaker industry since 1974. He is the author of the *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 currently the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, which is published on a monthly basis. For several years, Vance has been working as an engineering design consultant for various speaker manufacturers and is responsible for numerous award-winning designs, including 20 THX home theater certifications. Speakers that Vance designed that have won magazine and industry design awards include the Artison Portrait, Atlantic 170, 270, 370, 450, and 8200, and the coNEXTion Sonoma 10 and z600c.

Mike Klasco ("Loudspeakers Revealed: A Look at the Bits and Pieces of a Loudspeaker," p. 7) is the president of Menlo Scientific Ltd. in Richmond, CA., a consulting firm for the loudspeaker industry. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike contributes monthly to *audioXpress*. He specializes in materials and fabrication techniques to enhance speaker performance.

Ed Simon ("Is There DC on Your AC?," p. 35) received his BSEE from Carnegie Mellon University. Since then he has provided over 500 commercial sound systems to places that include: The Verizon Center, Washington DC; Flint Center, DeAnza Community College; KFC YUM Arena, Louisville; and the Miller Performing Arts Center, Alfred, N.Y.

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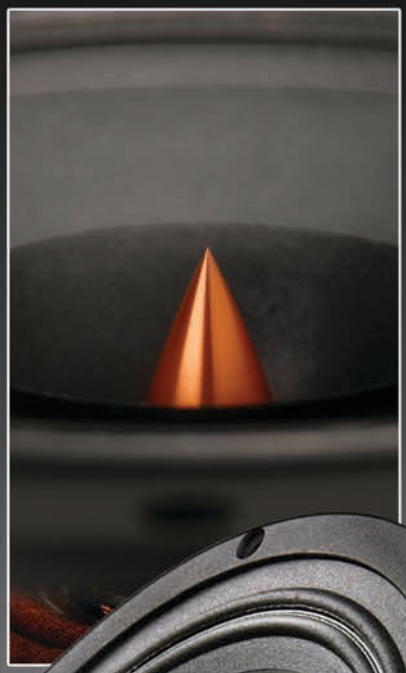

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FROM EVERY ANGLE

The Dayton Audio PS Series full-range Neodymium drivers do it all: true bass, articulate midrange, and balanced, extended highs. Although an ideal match for lower powered Class A and Class T designs, the PS Series drivers' superb resolution and natural sound quality make them a viable option for an even wider variety of applications. The PS Series' audio performance is complemented by the drivers' striking appearance from any angle.

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