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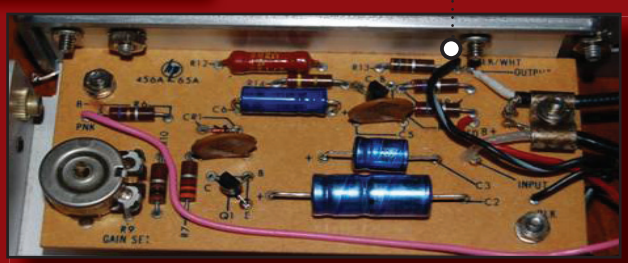

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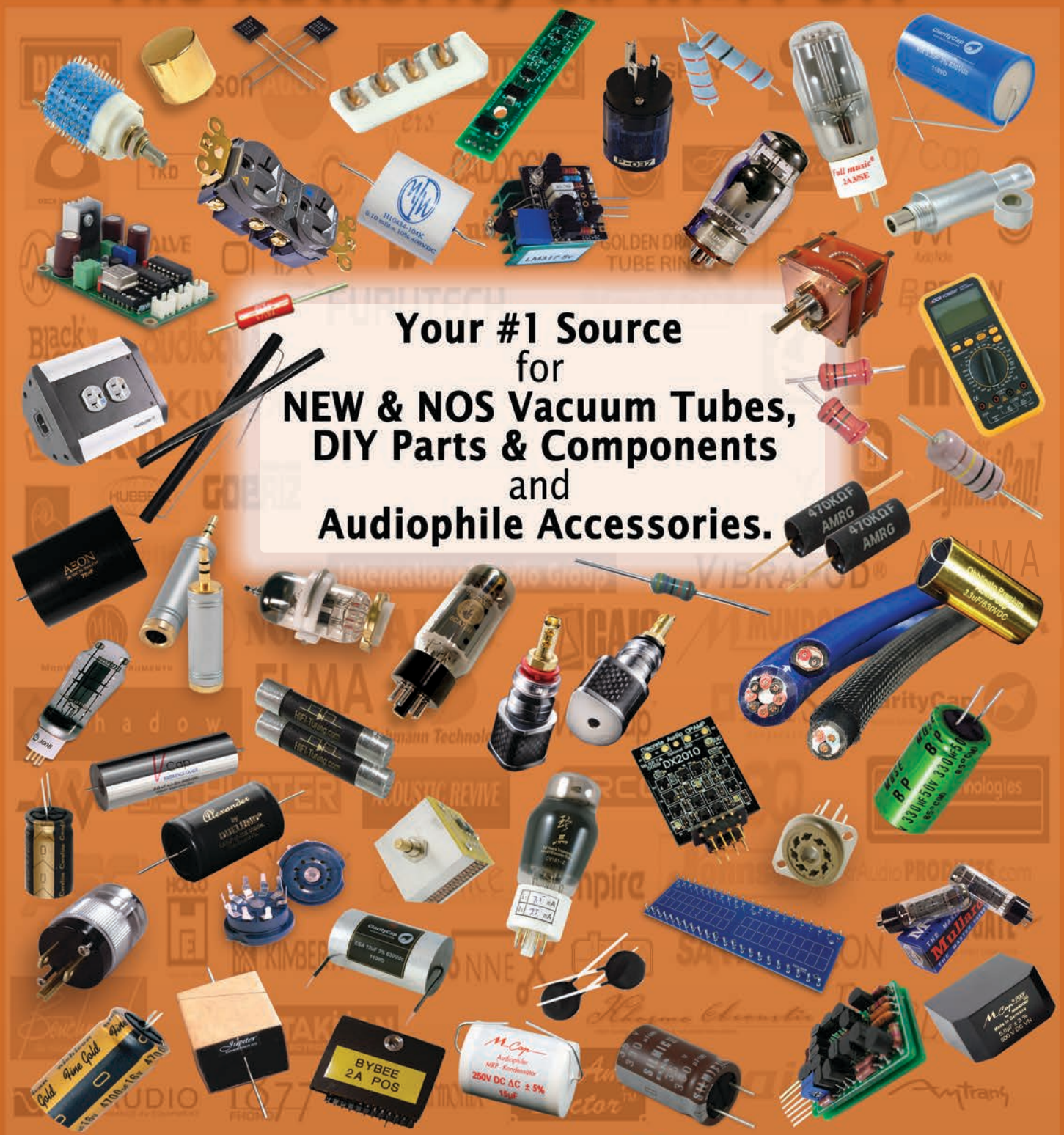

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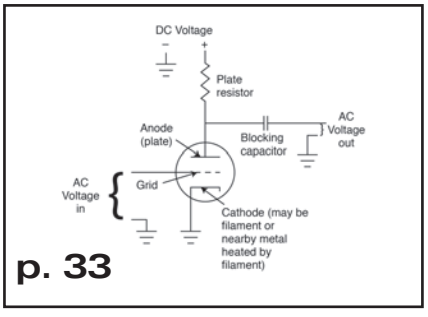

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audioXpress (US ISSN 1548-6028) is published monthly, at $50 per year for the U.S., at $62 per year for Canada/Mexico, at $85 per year Foreign/ROW, by Segment LLC, Hugo Van haecke, publisher, at 4 Park St., Vernon, CT, 06066, U.S.A. Periodical postage paid at Vernon, CT and additional offices.

POSTMASTER: Send address changes to: ***audioXpress***, Segment LLC, 4 Park St., Vernon, CT 06066, U.S.A.

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"Hollow-State Electronics" and More

Since September 2011, I've been preparing you for the exciting changes in store for *audioXpress*. Thus far, we've achieved the following: we've lengthened the magazine from 32 to 48 pages; we've published each issue in full-color; we've created high-quality columns to give you a steady supply of useful, predictable content; we've made subtle changes to the magazine's layout to make it easier on the eyes; we've expanded our coverage to bring you handy audio engineering articles on a consistent basis; we've begun profiling members of the international *audioXpress* community (see below); and more. This month, I'm proud to announce another milestone: a new monthly column on the topic of tubes and glass audio. You asked for it, so we're delivering.

During the last few months, one of the most consistent requests from members has been for more glass audio-related content. To meet this demand, Richard Honeycutt has begun a monthly column called "Hollow-State Electronics," providing glass audio content in each upcoming issue.

With a PhD in electroacoustics and a long list of published articles and papers on his resume, Richard is a perfect fit for *audioXpress*. Like many of our members, Richard's diverse interests range from speaker building to tube guitar amp design. Thus, I'm confident members from of different backgrounds will find his articles intriguing and informative. Turn to page 33 to read his first article.

After Richard's article satiates your craving for glass audio info, be sure to check out the rest of this issue's enlightening articles. On page 8, Mike Klasco and Steve Tatarunis examine "the surround." If you're interested in amp design, read about Robert Nance Dee's 5002 project on page 12. For insight about two new drivers from SB Acoustics, check out Vance Dickason's Test Bench column on page 17. If you're into retro electronics, test equipment, or both, Bill Reeve's article, "Restoring the HP 456A Current Probe," is for you. And if you didn't make it to AES 2011, don't worry. David J. Weinberg has you covered with his review on page 36.

Regards,

C. J. Abate
editor@audioxpress.com

MEMBER PROFILE

Member Name: Gary Galo

Location: Potsdam, NY, USA

Education: Gary holds a Bachelor's degree in Music Education (1973) and a Master's degree in Music History and Literature (1974) from the Crane School of Music, SUNY Potsdam.

Occupation: Audio Engineer, Crane School of Music, SUNY Potsdam, since 1976. Gary is now semi-retired and working half-time.

Member Status: Gary currently subscribes to *audioXpress*. He subscribed to all of *audioXpress*'s predecessors: *Audio Amateur*, *Audio Electronics*, *Speaker Builder*, and *Glass Audio*. Gary has published articles in all five periodicals.

Affiliations: Gary has been an AES member since 1975. His main professional organization has been the Association for Recorded Sound Collections (ARSC).

Audio Interests: Gary is fascinated by the following: pre-amps; digital players and DACs; and the playback of disc records, including recording curves and other special requirements for 78-rpm discs and other pre-RIAA formats. He prefers well-designed, solid-state equipment to tubes. Gary is also an avid classical record collector with 3,800 LPs, 7,000 CDs, and 3,000 78s.

Most Recent Purchase: Monarchy SE-100 MK2 Power Amps. Gary has advocated monaural power amps for many years. When Monarchy offered an affordable upgrade toward the new version of this amp, he traded in his older SE-100 Delux amps for the latest versions. Gary said: "The amps offer a new level of transparency and refinement that is truly remarkable in their price class."

Current Audio Projects: Gary is adding balanced outputs to his extensively modified Adcom GFP-565 pre-amps (*audioXpress:* November 2003, December 2003, January 2004, February 2004, December 2004), improving the line-stage buffers, and building a new external RIAA phono pre-amp of his own design.

Dream System: Gary doesn't spend much time dreaming about purchasing a newfangled system. Instead he concentrates on refining his existing system. Gary said his long-time dream of having a dedicated listening room for his audio system was fulfilled in 2005 when he and his wife Ellen bought a larger house. The home has a large listening room and two additional rooms for his record collection. Gary said: "There's nothing like having your audio system resting on a concrete floor! Adding dedicated AC power lines and clean AC regeneration put the icing on the cake (*audioXpress*, January 2011)."

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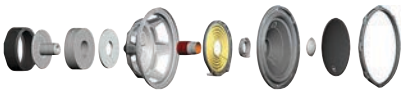
By Mike Klasco and Steve Tatarunis (United States)

The Surround



A look at surround materials, fabrication, and quality

Photo 1: Assorted surrounds (Source: Dr. Kurt Müller, GmbH & Co. KG)

Photo 2: Half-roll surround (Source: Dr. Kurt Müller, GmbH & Co. KG)

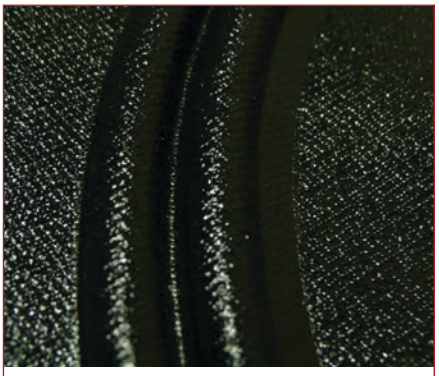


Photo 3: M surround (Source: Dr. Kurt Müller, GmbH & Co. KG)

The loudspeaker suspension system provides restoring force and contributes to the mechanical frictional losses. The electrical circuit analogy of the Thiele-Small parameters characterizes this as "Qms." The typical speaker cone is suspended by a two-point suspension, at the front edge (surround) and at the rear apex (spider or damper). Last month we explored many aspects of the spider suspension, which keeps the voice coil centered. This month our focus is the surround.

A surround's quality can have enormous impact on the frequency response of a loudspeaker. Appropriate selection of material and design ensures that the cone's behavior and the linearity of motion will maintain the speaker's dynamic integrity. This month our focus is the surround.

WHY SURROUNDS ARE NEEDED

The purpose of the surround (see **Photo 1**) is to attach and seal the cone to the speaker frame while allowing the cone to move forward and back. A loudspeaker's surround is the flexible joint between cone and frame. It serves to absorb the vibration energy without interfering with the dynamic behavior of the moving loudspeaker components. A good loudspeaker surround has several distinct functions:

- To absorb the sound energy traveling through the cone that reaches the edge, thereby reducing reflections back down the cone as well as suppress resonances passing from the cone to the chassis
- To stop air from passing around the cone
- To help center the voice coil in the magnetic gap in conjunction with the spider

SURROUND PROFILES

There are a couple of typical surround profiles that are popular, the half-roll and the accordion. Half-roll (see **Photo 2**) is commonly used in home speakers, offering high excursion, low-resonance frequency (extended low-end response). The accordion, or W edge, has a number of variations such as the M (two upward rolls) and triple roll (three upward rolls). Accordion (see **Photo 3**) and its variants can be found in musical instrument, sound reinforcement, and similar speakers.

Half-roll surrounds can be facing out (convex profile) or in (concave profile). To minimize edge resonance notches in the speaker's frequency response, abrupt discontinuities should be avoided—such as the inverted surround. Yet, when try-

Photo 4: Foam surround (Source: Dr. Kurt Müller, GmbH & Co. KG)

Photo 5: Textile surround (Source: Dr. Kurt Müller, GmbH & Co. KG)

Photo 6: Rubber surround (Source: Dr. Kurt Müller, GmbH & Co. KG)

ing to reduce clearance in front of the speaker to a required limited amount, using an inverted edge may be an expedient solution.

THE ATTACH

The surround attach is where the cone and surround overlap and are bonded together. The "attach" can be flat or angled. The angled attach is where the ID of the surround has the same angle as the cone body angle. The attach can be on the face or backside of the cone, and sometimes this decision is a fudge-factor when using an existing parts inventory and getting the assembly to stack up correctly. One alternative to bonding the surround to the cone is to overmold a thermoplastic elastomer (TPE) surround over the edge of the cone. When optimized, this approach provides a strong assembly with reduced edge resonance.

COMMON SURROUND MATERIALS

The surround may be resin-treated cloth, resin-treated non-wovens, polymeric foams, rubber compounds, or thermoplastic elastomers. An ideal surround has a linear force-deflection curve with sufficient damping to fully absorb vibrational transmissions from the cone/surround interface, and the "toughness" to withstand long term vibration-induced fatigue and exposure to the environment.

Let's discuss these materials in a little more detail.

Foam—Foam surrounds (see **Photo 4**) are a reasonably priced solution. Typical formulations are polyester or polyether. They are slit to thickness to achieve target resonance frequency. Material thickness and mechanical strength are adapted to the individual customer requirements. Foam surrounds are available in various durometers, thicknesses, and colors. Multilayer foam can be used for enhanced robustness. Water resistant chemically treated foams with high hydrolysis resistance are recommended for high humidity applications. Foam surrounds do disintegrate over time, but they are light and allow for high excursion.

Textile—Accordion surrounds (see **Photo 5**) are generally made of cloth, coated with a sealing/damping elastomer. Materials range from polyester cotton and cotton to aramids, like Conex and Nomex. These fabrics are available in a wide range of weights, weaves, mesh sizes, and treatments. During manufacture, the fabrics are impregnated with phenolic, coated and stamped. Fabric surrounds provide high geometrical precision. Mechanical characteristics (strength, damping) of Conex and Nomex fabrics are exactly reproducible and identical in the push-and-pull direction. This provides high linearity (low-distortion) and consistent unit to unit performance.

Rubber—Thermoset rubber surrounds (see **Photo 6**) can be molded with varying thicknesses. Their cross sections are fabricated with extremely high precision. By balancing geometry and material according to the customer's individual requirements, specific edge-damping characteristics of the surround can be realized. Rubber surrounds are heavy—which reduces

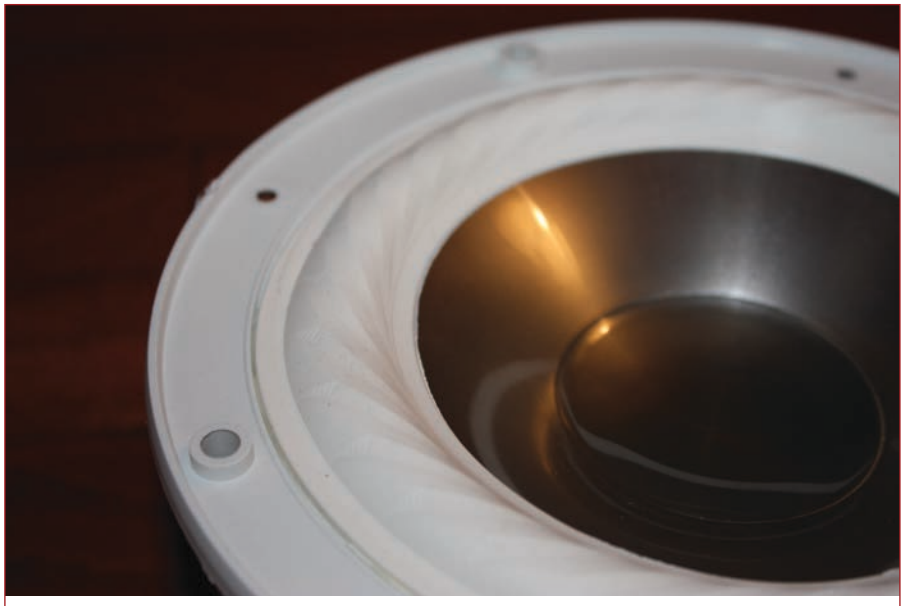


Photo 7: Santoprene surround (Photo courtesy of ExxonMobil Corp.)

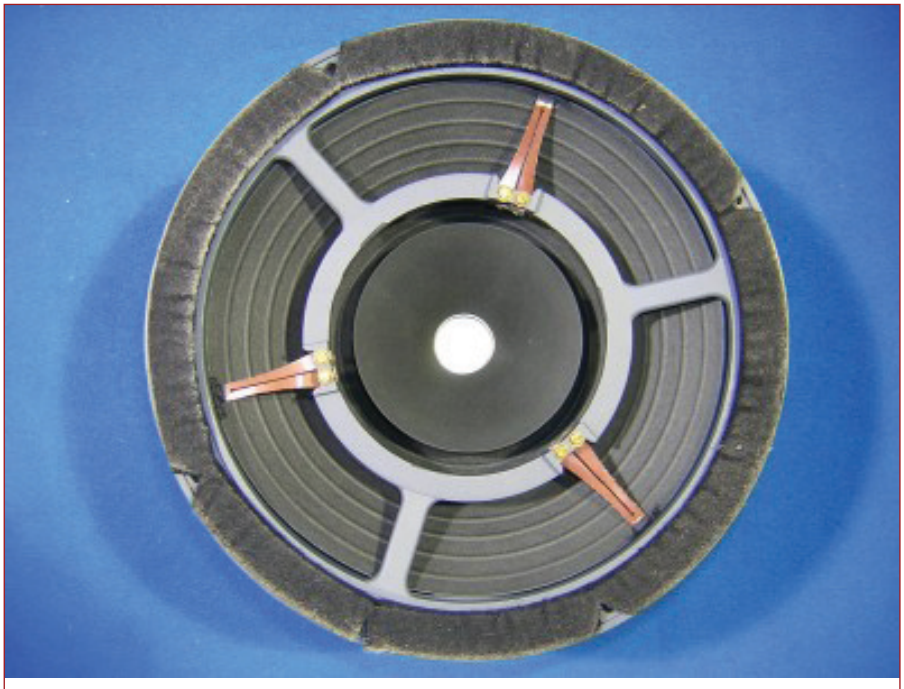


Photo 8: Goodmans Axiom 80 (Photo courtesy of German Vintage Loudspeakers.com)

efficiency, but they do not disintegrate. Rubber surrounds are also very robust and can be used in virtually any type of environment.

Thermoplastic Elastomers (TPEs)—TPEs are a mix of polypropylene and rubber (see **Photo 7**). ExxonMobile's Santoprene is quite popular in outdoor and autosound applications. TPE is injection moldable or vacuum formed. It is possible to overmold TPE over a speaker cone with excellent results.

WEIRD SCIENCE SURROUNDS

Over the decades, all sorts of unique surrounds have been patented with some used in commercialized products. Pioneer used a silicon surround on its marine speakers. Panasonic produced a speaker with alternating half-roll inverted and upward-facing sections. Goodmans, in the 1960s, offered the Axiom 80 with edgeless operation with centering by three cantilevers (see **Photos 8** and **9**). Foster also offered an edgeless surround woofer.

Photo 9: A close-up view of the Goodmans Axiom 80 (Photo courtesy of German Vintage Loudspeakers.com)

VOICE COILS

You're now familiar with spiders and surrounds. Next month we will explore the voice coil. *aX*

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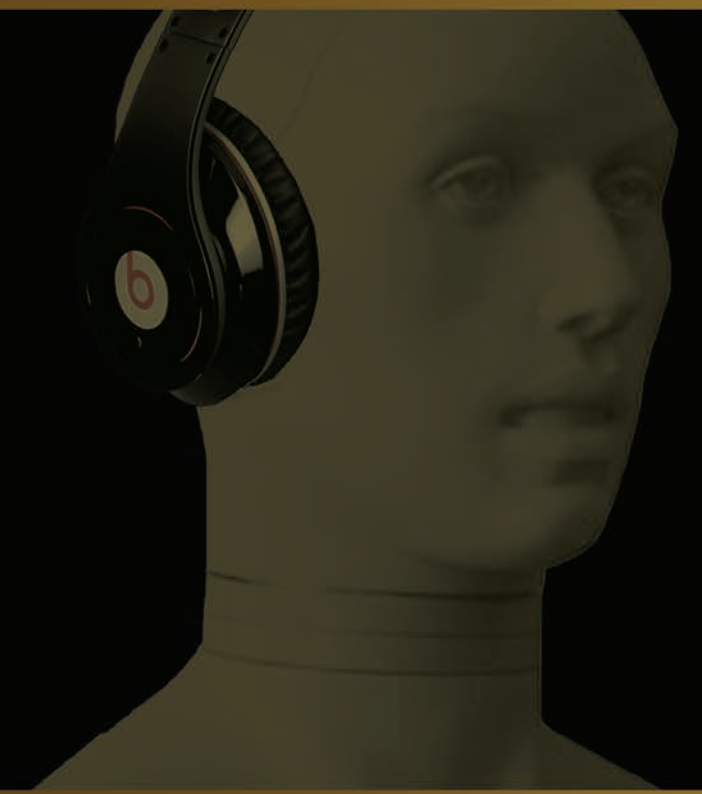

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By Robert Nance Dee (United States)

The 5002 Project

A do-it-yourself amp: the circuit, construction, and testing

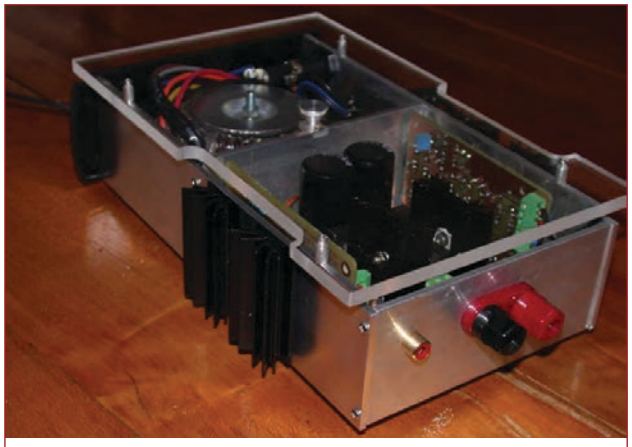

Photo 1: The 5002 amplifier project

I built this amplifier to run my Dynaudio X12 speakers. The X12s are not known for efficiency, but these little mono blocks run the accurate X12s with ease. Don't be fooled by their low power output. The limiting factor is voltage, not their ability to drive a load. Voltage limits the power because I conservatively limited the drive voltages to ±18 V. I could have gone higher but didn't need to. They are fine just as they are and with a damping factor of 0.219 Ω and better than 15 W each into 4 Ω, they drive my speakers with no fuss.

What you do get in spades is accuracy and an ability to drive difficult loads well. The Burr-Brown HA-5002 (see **Photo 1**) is a gem that I have used in several audio designs in both preamps and power amps. Why not? It has a slew rate of 1300 V/µs. **Figure 1** shows one block's square wave output, not at 10 kHz but 100 kHz. The 1 MHz trace is as equally impressive. Try and accomplish that with an op-amp!

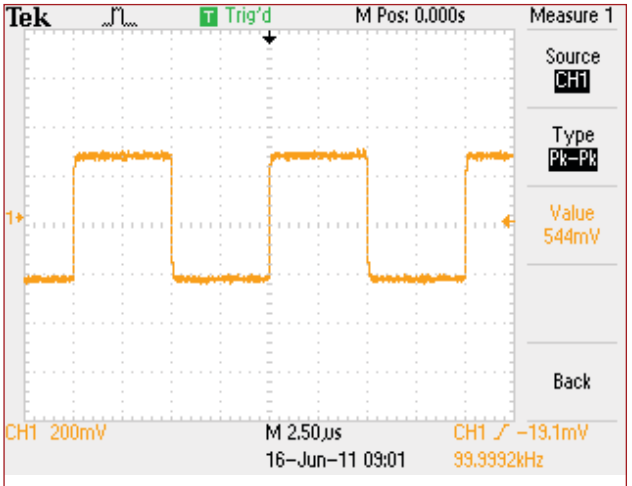

Figure 1: This trace shows one block's square wave output.

This is a current amp (ie., the gain is unity), but I added the ability to jumper in an op-amp for gain (see **Figure 2**). Why make the whole amp out of slow poke op-amps when all we need is one? Besides, I like to control amplification in my preamps where I can contour and control the delicate signal and not in the power amp where I want brute force to drive dynamic loads. Do we really need another 20 db of gain in the power amp after we already have 20 to 25 db of gain driving it?

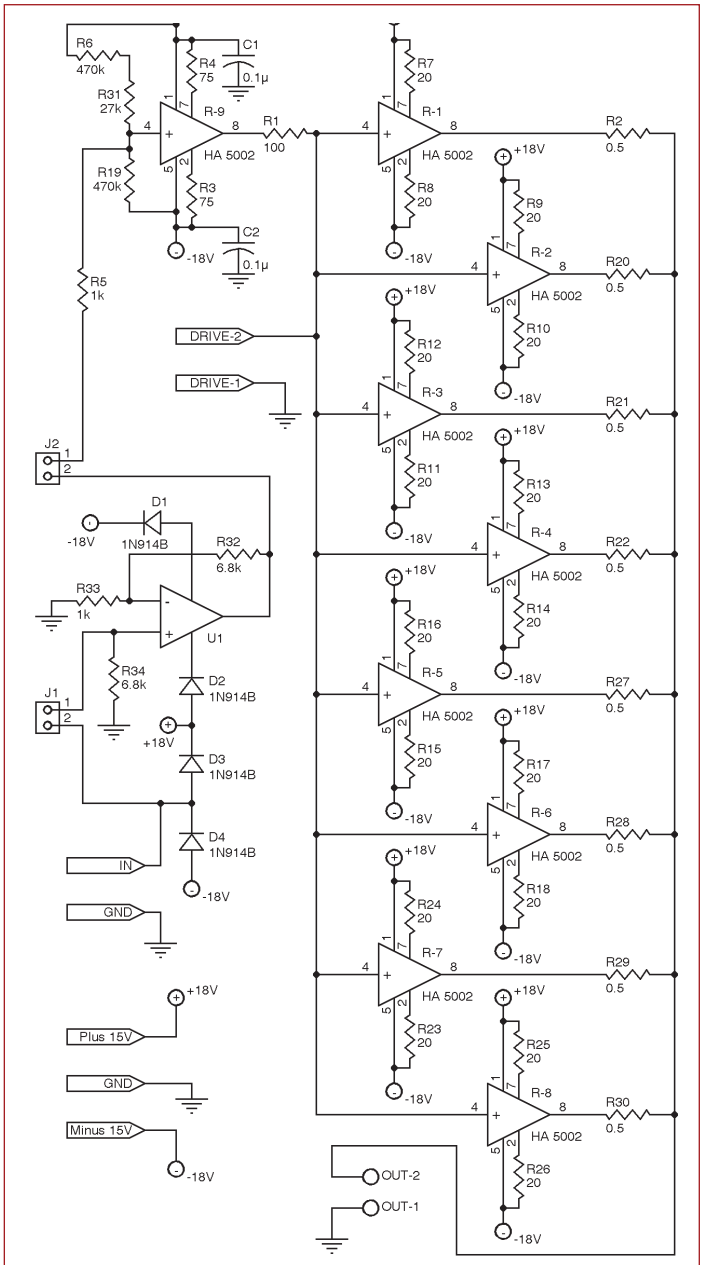

Figure 2: Power amp 627

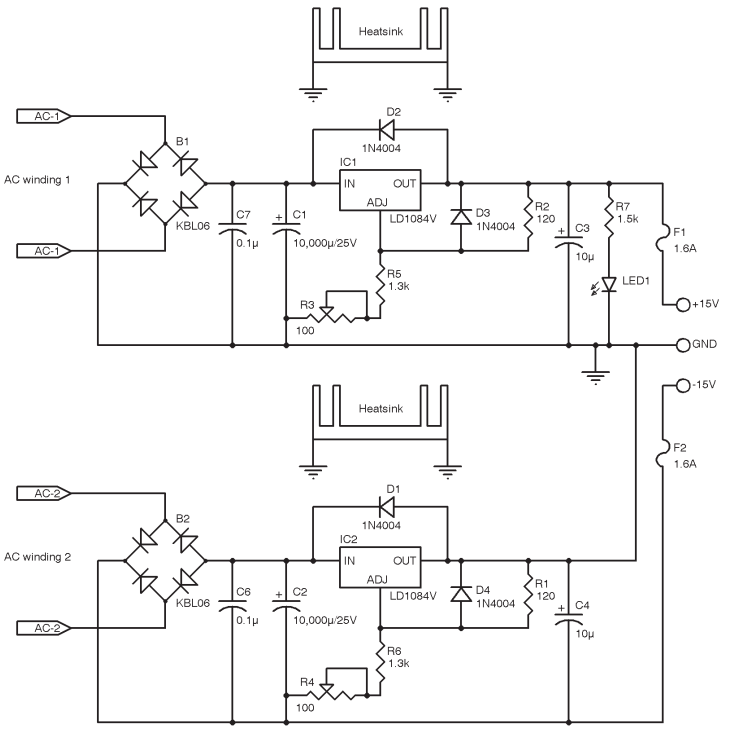

Figure 3: Power supply

While I added op-amp ability, I personally don't use it. This amp sits in my office and is run by my 20-db gain preamps streamed from my computer, CD player, and an analog tuner. What I did add is a powered subwoofer because I think two-way speakers with a subwoofer make the best sound. What I will add to the system is my venerable AR turntable and an MC cartridge. When I do, I'll build an MC amp. I tried the popular OPA227 and, with its better specs, the OPA627. Neither sounded bad, but I prefer my own JFET and 6SN7 tube preamps. They put out a sound I really enjoy for hours on end.

THE CIRCUIT

The power supply is straightforward. It runs two LM1084-5A regulators, each controlled by a multiple-turn pot (see **Figure 3**). (I like to balance my buffers with exactly the same ±voltages.) Voltage drop at full power is low driven from a dual output torroidal transformer. You do not have to use a torroidal transformer, in fact, two 15- to 18-V transformers will work fine, although likely take up more space. Just make sure to keep the transformer away from signal paths. The best solution is a dedicated power box.

The amplifier can be built to suit your needs. Running with only the drive board and four or eight 5002s in one case will give you a small footprint for tight spaces and nightstands. You can reduce the power supply accordingly.

The op-amp choice is yours. I tested the circuit with an OPA227 and an OPA627, but others will work fine. Because they max out at ±18 V, I used two silicon diodes to drop the ±18-V supply 0.7 V on each leg for op-amp safety. To incorporate the op-amp into the circuit, place two jumpers on the four pins, one jumper on J1 and the other on J2. To use the amp without the op-amp, place one jumper on the two middle pins. That's it; nothing else is required. In fact, I tested the amp in both configurations switching between the two often in a matter of seconds.

D3 and D4 keep the input signal from exceeding the power supply voltage and, thus, the input levels. The 0.5-Ω resistors on the outputs of each 5002 prevent hogging, but I ran this amp without them for a year with no problems, even so damping is quite good with them in or out. The 20-Ω current-limiting resistors save the chips, but a short on the output will blow the two 1.6-A power supply fuses before

any damage is done.

BUILDING AND TESTING

I have worked to keep this design within the scope of most builders. There are no SMD devices, and the boards go together well.

Figure 4 shows how to heatsink the 5002s. It works very well and by cutting standard heatsinks in half or using a thick aluminum case as the heatsink this amp will run very cool. The trick to get all the chips to touch the heatsink equally is to place all the sockets on the board loosely and then place a book or flat object on top of the sockets. Now, turn the complete board and flat surface over as one unit so that the board bottom faces up. Then, solder all the sockets into place.

What the builder should pay very close attention to is testing. One solder short can destroy all the chips in a few seconds.

Before any chips are inserted in their sockets (I prefer using sockets), power up the board and check pins 1 and 7 of the 5002s for +18 V. Once this is verified, check pins 2 and 5 for –18 V. On the main board, with power off, place one lead of an ohm meter on pin 8 of the drive 5002 (#1) and then check every other 5002 for 100 Ω on pin 4 with respect to pin 8 of the main drive chip. This is important because the drive chip keeps the bias of the chips at near 0 V. (Bias should be within 100 mV, my amps run about 10 to 20 mV max.)

Next, check again with the power off, from pin 4 to pin 8 on each chip. This should be an open circuit. On board two (the one without the drive chip), check from the input drive pin to pin 4 on each chip (see **Figure 5**). This should be a dead short.

Once all these test pass, place the 5002 drive chip in and one other 5002, power the amp up, put a 10-kHz signal on the input (make sure one jumper is on the two inner pins), and check the output for the identical signal. After this passes, place half the output chips in and retest, place the final chips in, and test again. Finally, place an op-amp in its socket place jumpers on the outside pins. You should see the input signal with the gain you have chosen. I set the gain at 6.8, but you can change the 6.8-kΩ resistor to a higher value to get the gain you want (e.g., a 10-kΩ resistor will give you 20-db gain or a voltage gain of 10). Don't go lower. Some chips don't work well under a gain of 5.

This amp is quite remarkable. As I said, I use it with my JFET and tube preamps without any harshness. In fact, it takes on the character of the preamp, which is exactly what I'm looking for. It will embarrass poor music and

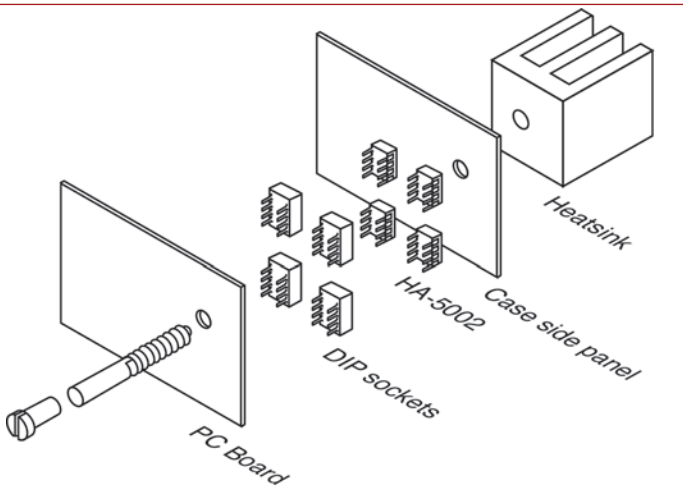

Figure 4: How to heatsink the 5002s

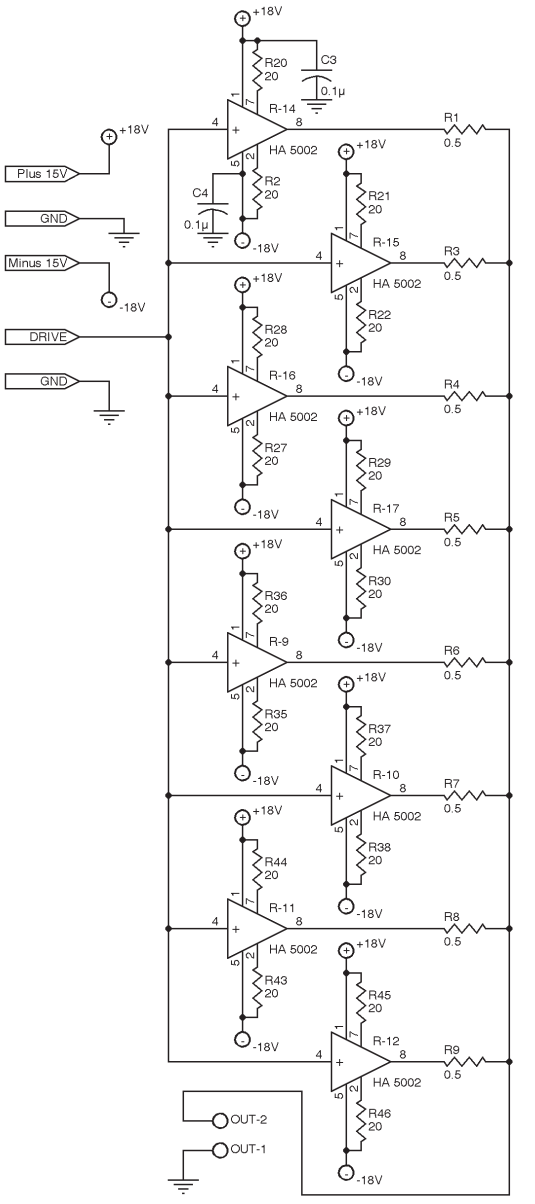

Figure 5: Power amp 627 bd2

poor preamps with its accuracy. It will reward solid designs and quality speakers with an exceptional sound that has exceeded most of the small amps I've heard.

SPECIFICATIONS

A word about the specs. Distortion weighs low on my scale. As I've said, often no one can hear the difference between 0.05% and 0.001%. I've heard wonderfully musical amps that have 0.5% distortion (both tube and solid state). So, when I check specs, if the distortion and noise are below 0.1%, I don't pay much more attention to them. Rise time is another issue, and I can hear this characteristic. I get weary of amps that have poor rise times. Orchestras get muddy and harmonics blur, guitar music bloats and loses clarity. This amp has exceptional slew rates, and I think you'll agree it makes a difference. Run a pair of two way book shelf speakers with a sub, and you'll be all smiles! *aX*

Editor's note: Robert Nance Dee provided supplemental information for the project, including a bill of materials. The list of materials and additional figures are available at www.audioamateur.com.

SPECIFICATIONS

- Bandwidth.....Less than 5 Hz to greater than 1 MHz
- Damping.....0.218 Ω (16 A)
- Distortion.....Less than 0.05%
- Slew rate.....Greater than 1000 V/µs (op-amp out)
- Power (each amp).....Greater than 10 W into 8 Ω
  Greater than 15 W into 4 Ω

CD:
- OPPO DV-970HD

Preamp:
- "Willow" JFET (my design)
- "ZuZu" 6SN7 (my design)
- David Hafler 110

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- Dynaudio X12
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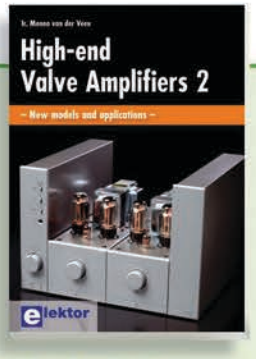

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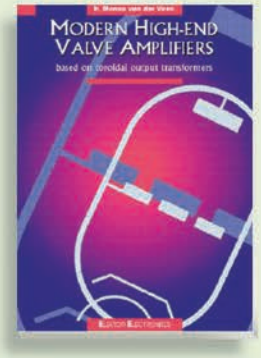

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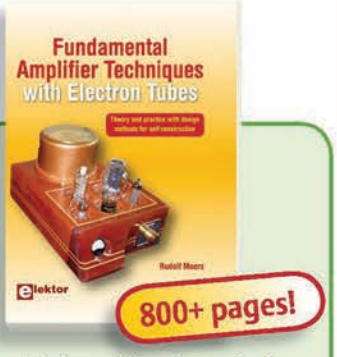

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By Vance Dickason (United States)

Two 4" Drivers from SB Acoustics

A test of two high-end home audio drivers

As most of you are probably aware, SB Acoustics was formed as a partnership between former DST (Vifa, Peerless, and ScanSpeak OEM) engineering sales representative David Stephens and Indonesia OEM loudspeaker manufacturer Sinar Baja, with an additional technical partnership with former Vifa/Scan-Speak engineers Ulrik Schmidt and Frank Nielsen, now co-owners of Danesian Audio. Danesian Audio does all the transducer engineering for SB Acoustics. *Voice Coil* featured transducers from SB Acoustics in November 2007 (SB24STC-C000-4 tweeter), December 2007 (SB29NRX75-6 10" woofer), March 2008 (SB34NRX75-6 12" woofer), September 2011 (SB26STCN-C000-4 tweeter), and November 2011 (SB26STAC-C000-4 tweeter and MW16R woofer).

SB Acoustics recently sent me two 4" drivers, one woofer, the SB12NRX25-4, and its 4" midrange version, the SB12MNRX25-4. Features for the SB12NRX and SB12MNRX, as you would expect, are nearly identical. Both use a proprietary coated paper and papyrus fiber curvilinear cone and coated paper dustcap, NBR surround, flat Conex spider, a very open proprietary cast frame with a substantially open area below the spider mounting shelf for cooling (this frame is both screwed and glued to the motor assembly), T-yoke, polished edges on the front plate and T-yoke, aluminum shorting ring, 25.4-mm diameter fiberglass non-conduction former, and gold-plated terminals. Both use ferrite magnets that are 20 mm × 70 mm, with the SB12MNRX having a secondary 9 mm × 60 mm ferrite magnet for additional Bl.

I decided to test the SB12NRX (see **Photo 1**) woofer version first. Analysis began using the LinearX LMS analyzer and VIBox to create both voltage and admittance (current) curves with the driver clamped to a rigid test fixture in free-air at 0.3 V, 1 V, 3 V, and 6 V. As has become the protocol for Test Bench testing, I no longer use a single added mass measurement and instead used actual measured mass, but the manufacturers measured Mmd data. At this point the 6-V curves were discarded as being too non-linear for the curve-fitting algorithm to resolve. Next, the remaining six 550-point stepped sine wave sweeps for each SB12NRX sample were post-processed and the voltage curves divided by the current curves (admittance) to derive impedance curves, phase added by the LMS calculation method, and along with the accompanying voltage curves, imported to the LEAP 5 Enclosure Shop software. Since most Thiele-Small data provided by OEM manufacturers is being produced using

Photo 1: The SB12NRX25-4, a 4" woofer from SB Acoustics

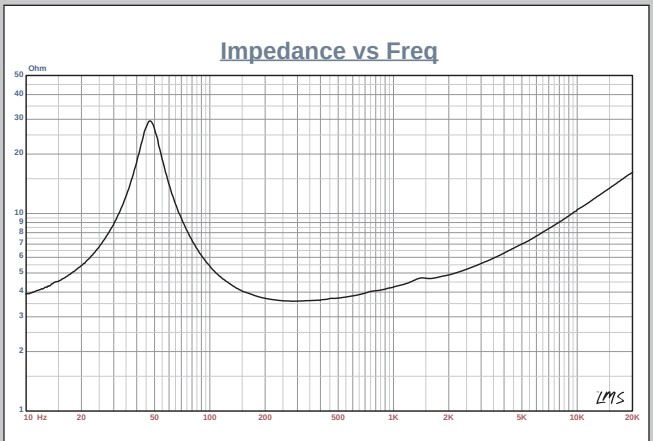


Figure 1: SB Acoustics SB12NRX25-4 woofer free-air impedance plot

	TSL model		LTD model		Factory
	sample 1	sample 2	sample 1	sample 2	
Fs	47.0 Hz	48.1 Hz	46.4 Hz	47.4 Hz	55 Hz
Revc	3.06	3.06	3.06	3.06	3.1
Sd	0.0055	0.0055	0.0055	0.0055	0.0050
Qms	3.76	3.68	4.69	4.73	3.20
Qes	0.44	0.44	0.53	0.53	0.46
Qts	0.39	0.39	0.48	0.48	0.40
Vas	8.95 ltr	8.53 ltr	9.25 ltr	8.85 ltr	5.6 ltr
SPL 2.83 V	85.1 dB	85.2 dB	84.2 dB	84.3 dB	87.5 dB
Xmax	5.0 mm	5.0 mm	5.0 mm	5.0 mm	5.0 mm

Table 1: SB Acoustics SB12NRX25-4 woofer

either a standard method or the LEAP 4 TSL model, I additionally created a LEAP 4 TSL model using the 1-V free-air curves. The complete data set, the multiple voltage impedance curves for the LTD model (see **Figure 1** for the 1-V free-air impedance curve), and the 1-V impedance curve for the TSL model were selected in the transducer derivation menu in LEAP 5 and the parameters created for the computer box simulations. **Table 1** compares the

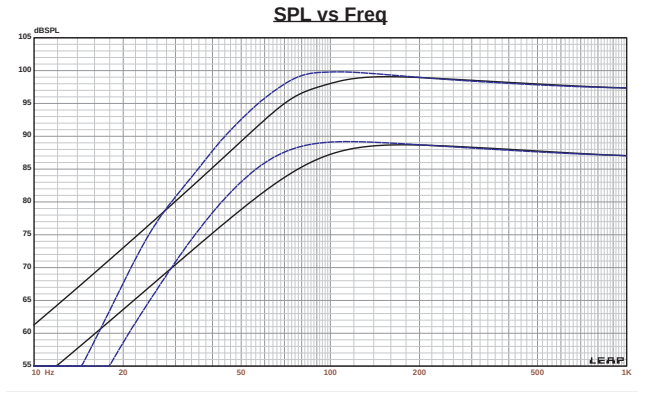


Figure 2: SB Acoustics SB12NRX25-4 computer box simulations (black solid = sealed 1 @ 2.83 V; blue dash = vented 2 @ 2.83 V; black solid = sealed 1 @ 10 V; blue dash = vented @ 10 V)

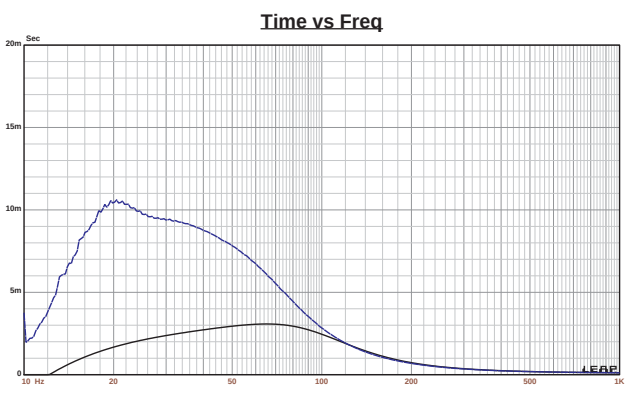


Figure 3: Group delay curves for the 2.83-V curves in Figure 2

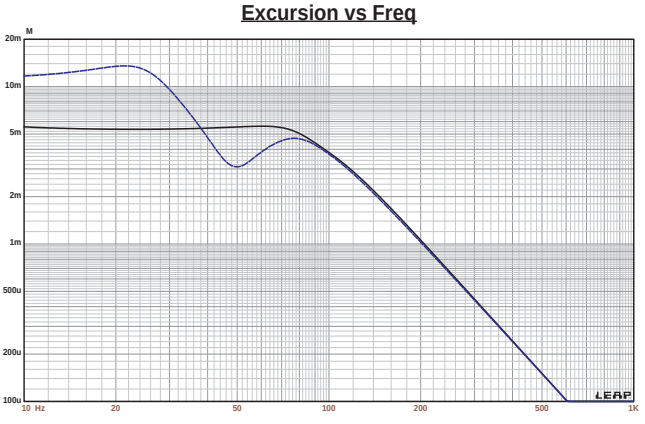


Figure 4: Cone excursion curves for the 10-V curves in Figure 2

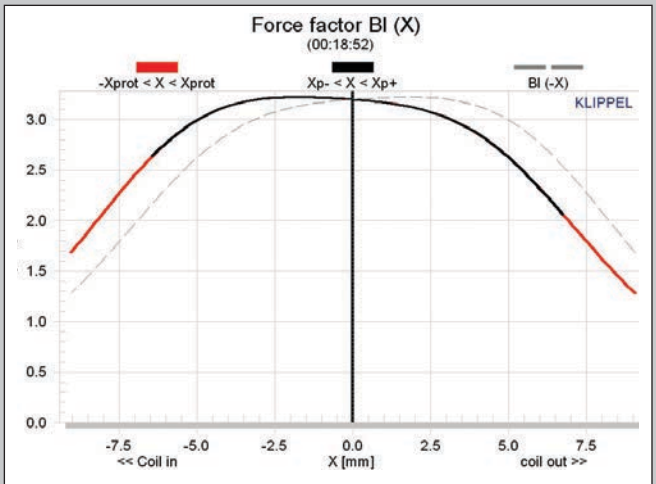


Figure 5: Klippel Analyzer BI (X) curve for the SB Acoustics SB12NRX25-4

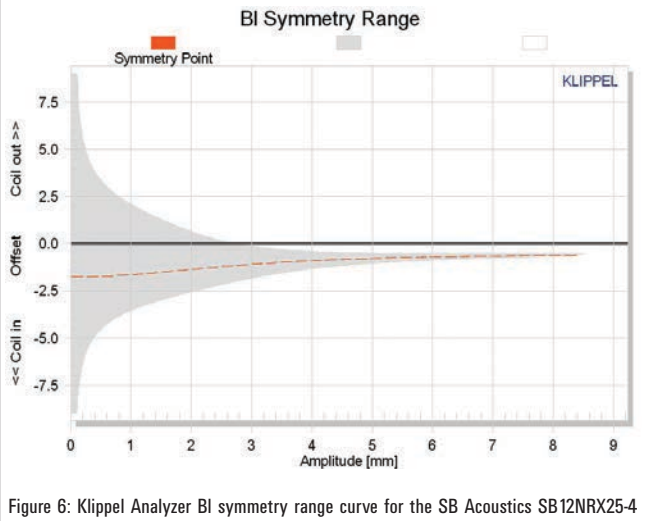


Figure 6: Klippel Analyzer BI symmetry range curve for the SB Acoustics SB12NRX25-4

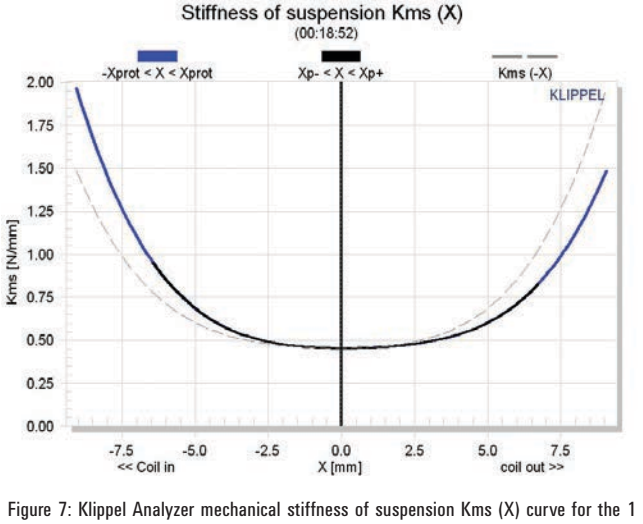


Figure 7: Klippel Analyzer mechanical stiffness of suspension Kms (X) curve for the 1 SB Acoustics SB12NRX25-4

LEAP 5 LTD and TSL data and factory parameters for both of SB samples.

LEAP parameter calculation results for the SB were fairly similar to the factory TS parameters, with a higher Vas and lower SPL for the measured data compared to the factory data. While TSL parameters were closer to the factory data (except the Vas and SPL as mentioned), I proceeded to put together computer enclosure simulations using the LEAP LTD parameters for Sample 1. I set up two box simulations, one sealed and one vented. This produced a 0.12 ft^3 sealed box with 50% fiberglass fill material, and a 0.20 ft^3 Qb^3 vented enclosure with 15% fiberglass fill material and tuned to 48 Hz.

Figure 2 gives the outcome for the SB Acoustics SB12RNX25-4 in the sealed and vented boxes at 2.83 V and at a voltage level sufficiently high enough to increase cone excursion to Xmax + 15% (5.75 mm for the SB12NRX). This generated a F^3 frequency of 79 Hz with a box/driver Qtc of 0.69 for the 0.12 t^3 sealed enclosure (perfect for a home theater satellite and meeting the requirements for THX certification) and –3 dB = 61 Hz for the 0.20 ft^3 vented simulation. Increasing the voltage input to the simulations until the maximum linear cone excursion was achieved produced 99 dB at 10 V for the sealed enclosure simulation and 100 dB with the same 10-V input level for the larger vented box (see **Figures 3** and **4** for the 2.83-V group delay curves and the 10-V excursion curves). Note that I left the input at 10 V since the Xmax was being exceeded at 40 Hz. As with all vented enclosures, power handing can be significantly increased by adding a steep high-pass filter approximately one octave lower than the F^3.

Klippel analysis for the SB Acoustics 4" woofer (our analyzer is provided courtesy of Klippel GmbH)—performed by Pat Turnmire, Red Rock Acoustics (author of the SpeaD and RevSpeaD software)—produced the Bl(X), Kms(X) and Bl and Kms symmetry range plots given in **Figures 5–8**. This data is extremely valuable for transducer engineering, so if you don't own a Klippel analyzer and would like to have analysis done on a particular driver project, Red Rock Acoustics can provide Klippel analysis of most any driver for a nominal fee of $100 per unit. (For contact information, visit the Red Rock Acoustics website at www.redrockacoustics.com.)

The Bl(X) curve for the SB12NRX (see **Figure 5**) is moderately broad and symmetrical with a small rearward (coil-in) offset Looking at the Bl symmetry plot (see **Figure 6**), this curve shows a coil inward offset at the rest position of about 1.78 mm that decrease to about 0.78-mm inward offset at the physical Xmax (5 mm) and then stays constant throughout the remaining operating range of the driver, suggesting that the voice coil is not located at magnetic center. With proper assembly jigs, a manufacturer should be able to hole a ±0.25-mm tolerance at the magnetic center. This problem is not what you would call huge, and it mostly results in somewhat increased 2nd harmonic distortion (which tends to be categorized as "pleasant" by listeners), but is fairly common. If you go through the cumulative review in Test Bench, this occurs fairly often in samples I receive.

Figures 7 and **8** depict the Kms(X) and Kms symmetry range curves for the SB 4" woofer. The Kms(X) curve has some minor asymmetry beyond about 4 mm of excursion. There is no offset at the rest position, and starting at about 2 mm, there is a forward (coil-out) offset increasing to about 0.5 mm at the 5-mm Xmax

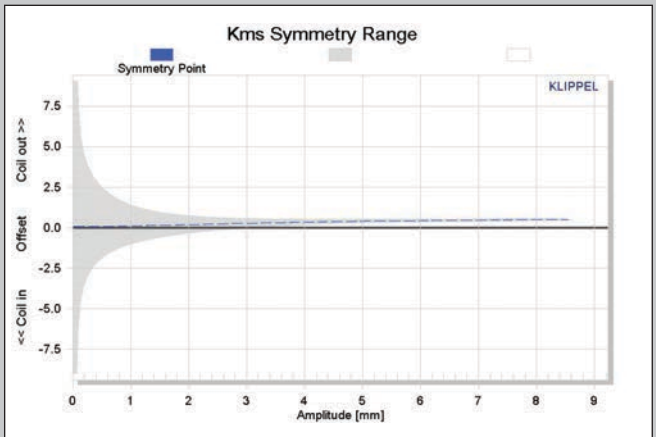


Figure 8: Klippel Analyzer Kms symmetry range curve for the SB Acoustics SB12NRX25-4

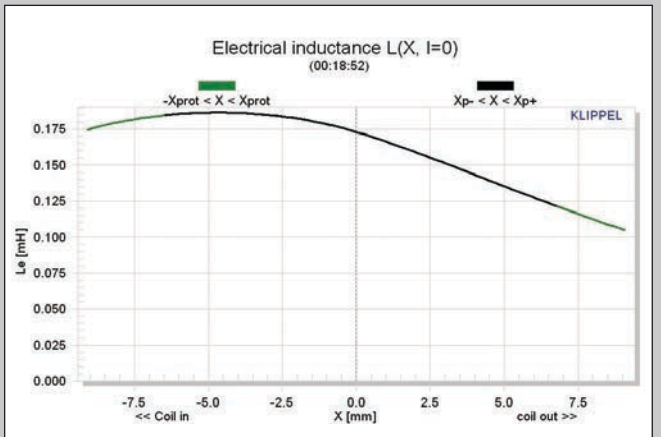


Figure 9: Klippel Analyzer Le(X) curve for the SB Acoustics SB12NRX25-4

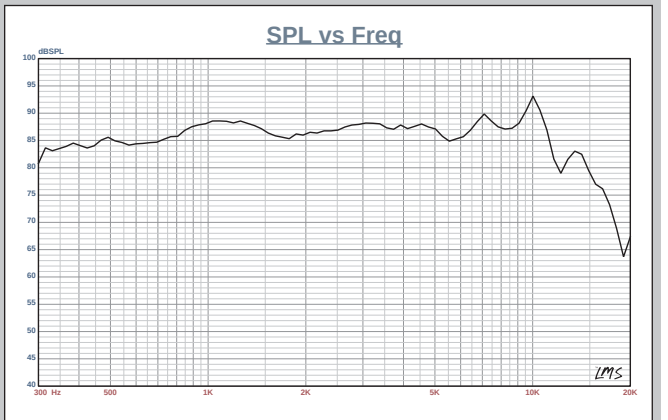


Figure 10: SB Acoustics SB12NRX25-4 on-axis frequency response

position. Displacement limiting numbers calculated by the Klippel analyzer for the SB12NRX were XB1 @ 82% Bl = 5.0 mm and for XC @ 75% Cms minimum was 4.3 mm, which means that for the SB 4" woofer, the compliance is the most limiting factor for prescribed distortion level of 10%. **Figure 9** gives the inductance curves Le(X) for the SB12NRX. Inductance will typically increase in the rear direction from the zero rest position as the voice coil covers more pole area, and although that is what you see for this driver, the variation in inductance across the operating range is only about 0.05 mH, which is a function of the aluminum shorting ring.

Next, I mounted the SB12NRX25-4 woofer in an enclosure which had a 13" × 6" baffle (filled with foam

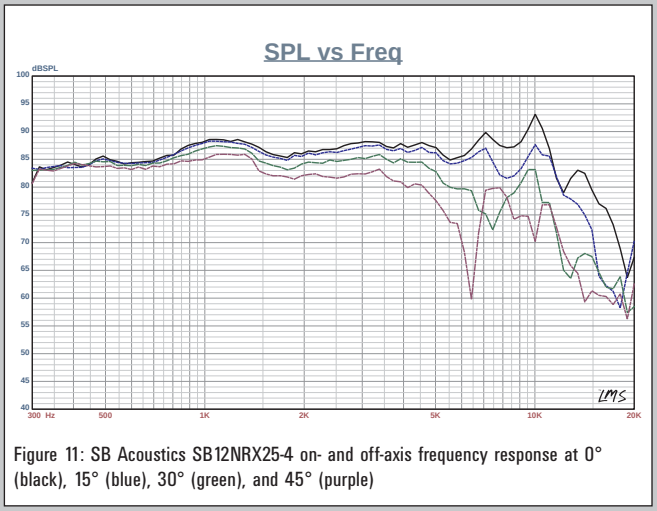


Figure 11: SB Acoustics SB12NRX25-4 on- and off-axis frequency response at 0° (black), 15° (blue), 30° (green), and 45° (purple)

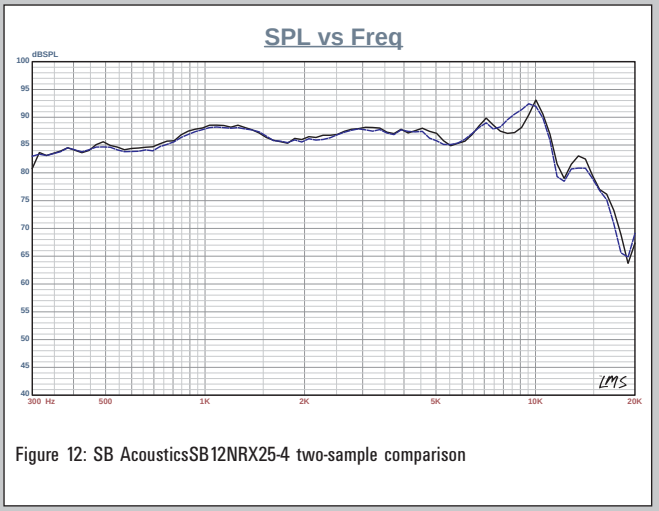
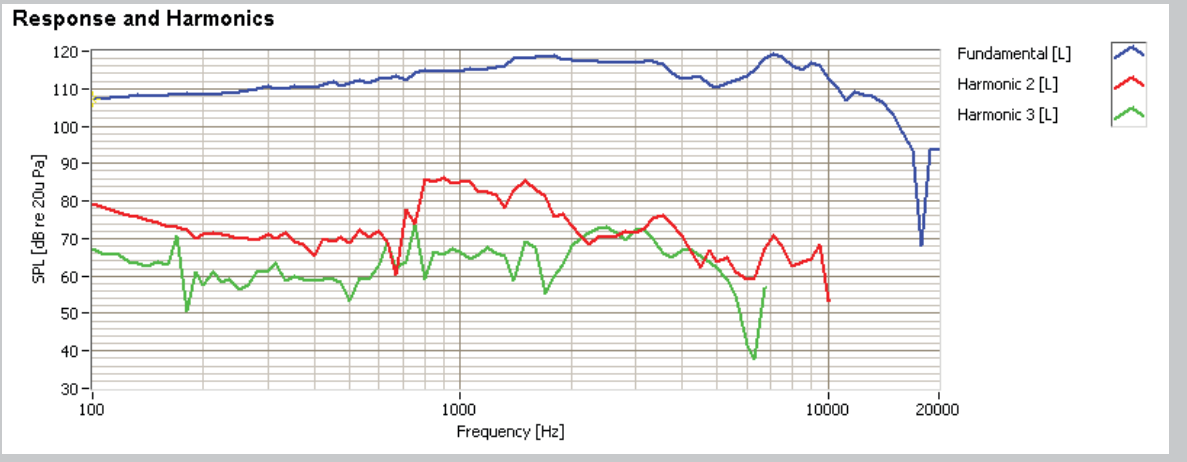


Figure 12: SB AcousticsSB12NRX25-4 two-sample comparison



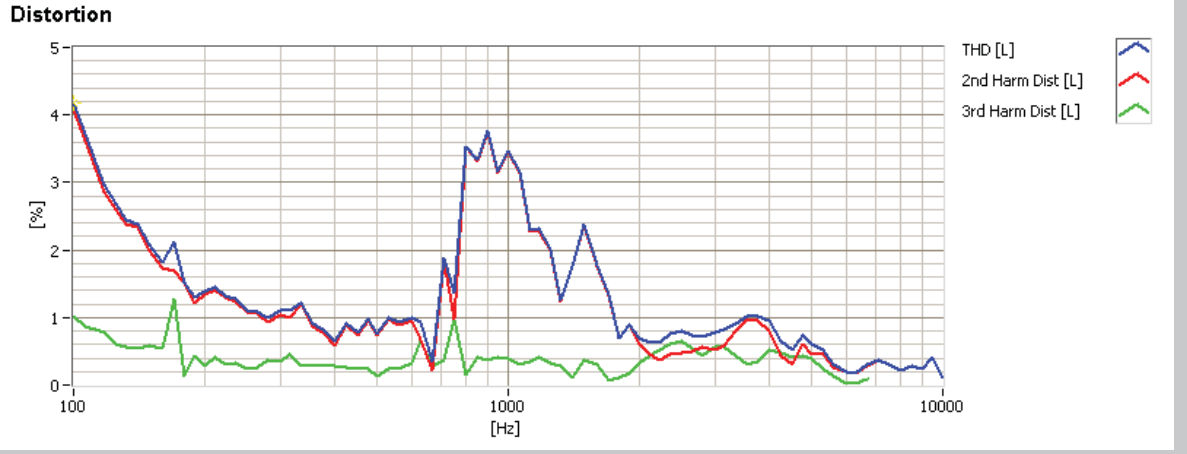


Figure 13: SB Acoustics SB12NRX25-4 SoundCheck distortion plots

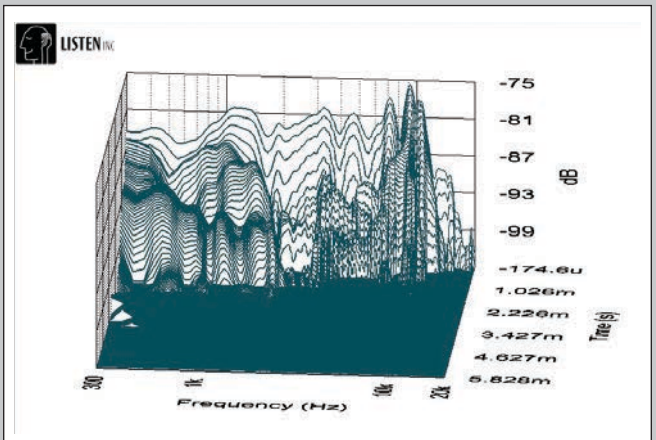


Figure 14: SB Acoustics SB12NRX25-4 SoundCheck CSD waterfall plot

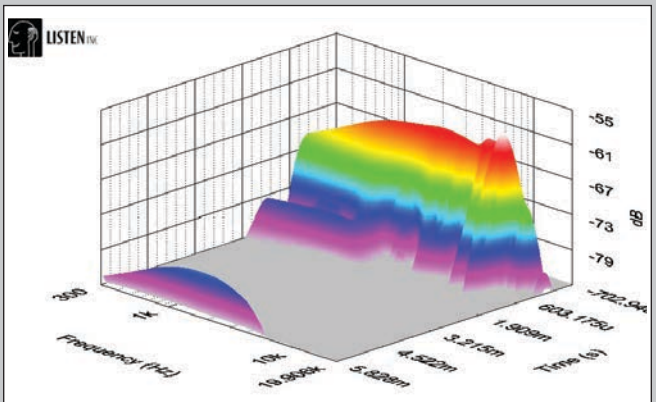


Figure 15: SB Acoustics SB12NRX25-4 SoundCheck Wigner-Ville plot

damping material) and then measured the driver on- and off-axis from 300 Hz to 20-kHz frequency response at 2.83 V/1 m using the LinearX LMS analyzer set to a 100-point gated sine wave sweep. **Figure 10** depicts the SB12NRX's on-axis response indicating a smoothly rising response to about 5 kHz, with the typical 3- to 4-dB peaking centered on 1.2 kHz, and a couple of breakup modes centered on 7 kHz and 10 kHz. **Figure 11** displays the on- and off-axis frequency response at 0°, 15°, 30°, and 45°. –3 dB at 30° with respect to the on-axis curve occurs at 4.5 kHz, so a cross point in that vicinity or lower should be optimal. And finally, **Figure 12** gives the two-sample SPL comparisons for the 4" SB Acoustics driver, showing a very close match throughout the operating range.

For the remaining group of tests, I used the Listen, Inc. SoundCheck analyzer (courtesy of Listen, Inc.) to measure distortion and generate time frequency plots. For the distortion measurement, the woofer was mounted rigidly in free-air, and the SPL set to 94 dB (6.85 V) at 1 m using a noise stimulus, and then the distortion measured at with the Listen, Inc. microphone placed 10 cm from the dust cap. This produced the distortion curves shown in **Figure 13**. I then used SoundCheck 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 14** and the Wigner-Ville (for its better low-frequency performance) plot in **Figure 15**.

SB ACOUSTICS 4" SB12MNRX

With analysis on the first SB woofer accomplished, I

Photo 2: The SB12MNRX, a 4" driver from SB Acoustics

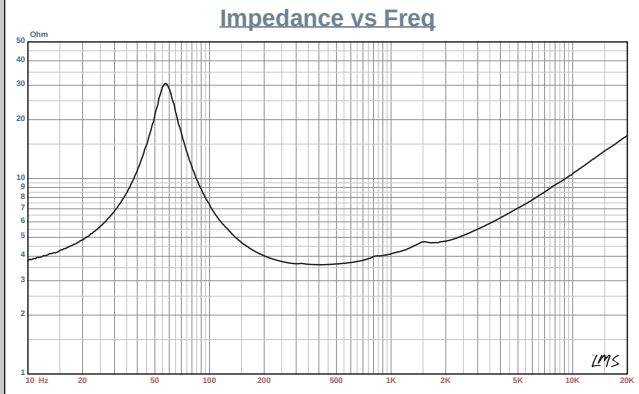


Figure 16: SB Acoustics SB12MNRX25-4 free-air impedance plot

	TSL model		LTD model		Factory
	sample 1	sample 2	sample 1	sample 2	
Fs	57.3 Hz	59.5 Hz	55.2 Hz	57.4 Hz	58 Hz
Revc	3.18	3.14	3.18	3.14	3.2
Sd	0.0055	0.0055	0.0055	0.0055	0.0050
Qms	3.79	4.25	4.25	4.05	3.6
Qes	0.44	0.43	0.53	0.47	0.42
Qts	0.39	0.39	0.47	0.42	0.37
Vas	6.88 ltr	6.37 ltr	7.46 ltr	6.90 ltr	5.5 ltr
SPL 2.83 V	86.5 dB	86.8 dB	85.6 dB	86.3 dB	88.5 dB
Xmax	2.25 mm	2.25 mm	2.25 mm	2.25 mm	2.25 mm

Table 2: SB Acoustics SB12MNRX25-4 midrange

performed an identical group of tests on the SB Acoustics 4" SB12MNRX (see **Photo 2**). As before, the LinearX LMS analyzer and VIBox were employed to create both voltage and admittance (current) curves. This test is accomplished with the driver clamped to a rigid test fixture in free-air with sweeps taken at 0.3 V, 1 V, 3 V, and 6 V. Like the SB12NRX, the 6-V curve was too nonlinear and was discarded, which is interesting considering the SB12MNRX has on 2.25-mm Xmax compared to the 5-mm Xmax of the woofer version. The remaining six 550-point stepped sine wave sweeps for each SB12MNRX sample were post-processed and the voltage curves divided by the current curves (admittance) to create impedance curves, phase added using LMS calculation method, and along with the accompanying voltage curves, saved to the LEAP 5 Enclosure Shop software. In addition to the LEAP 5 LTD model results, I also created a LEAP 4 TSL model set of parameters using just the 1-V free-air curves. The final data set, which includes the multiple voltage impedance curves for the LTD model (see **Figure 16** 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 2** compares the LEAP 5 LTD and TSL data and factory parameters for both of SB Acoustics 4" midrange samples.

LEAP parameter calculation results for the SB12MNRX midrange, like the woofer version, only moderately close to the SB factory data, again with the measured data yielding a closer match with the TSL data except for the higher Vas and lower SPL. I proceeded, however, to set up computer enclosure simulations using the LEAP LTD parameters for Sample 2. I set up two box simulations, one sealed and one vented. For the closed box simulation I used a 0.16 ft^3 enclosure with 50% fiberglass fill material, and for the vented box, a 0.15 ft^3 Qb^3 vented enclosure with 15% fiberglass fill material and tuned to 61 Hz.

Figure 17 displays the results for the SB12MNRX in the sealed and vented boxes at 2.83 V and at a voltage level sufficiently high enough to increase cone excursion to Xmax + 15% (2.6 mm for the SB12 midrange). This resulted in a F^3 = 106 Hz with a box/driver Qtc of 0.69 for the 0.16 ft^3 sealed enclosure and a –3 dB = 76 Hz for the 0.15 ft^3 vented simulation. Increasing the voltage input to the simulations until the maximum linear cone excursion was reached generated 97 dB at 6 V for the sealed enclosure simulation and 97.3 dB with the same 6-V input level for the larger vented box (see **Figures 18** and **19** for the 2.83-V group delay curves and the 6-V excursion curves). As with any midrange transducer, the box size is usually only a method to locate the resonance as low in frequency as possible to facilitate a passive high-pass section of a band-pass filter. Power handling with such a filter is substantially increased from the non-filter simulation above.

Klippel analysis for the SB Acoustics 4" midrange produced the Bl(X), Kms(X) and Bl and Kms Symmetry Range plots given in **Figures 20** to **23**. The Bl(X) curve for the SB12MNRX (see **Figure 20**) is fairly narrow, but very symmetrical, about what you expect from a small Xmax midrange. For all practical purposes, there is zero offset indication, so about as good as it ever gets. **Figures 22** and **23** give the Kms(X) and Kms symmetry range curves for the SB Acoustics 4" mid. The Kms(X) curve, like the Bl curve, is very symmetrical with practically no offset, indicating a lot of design integrity. Dis-

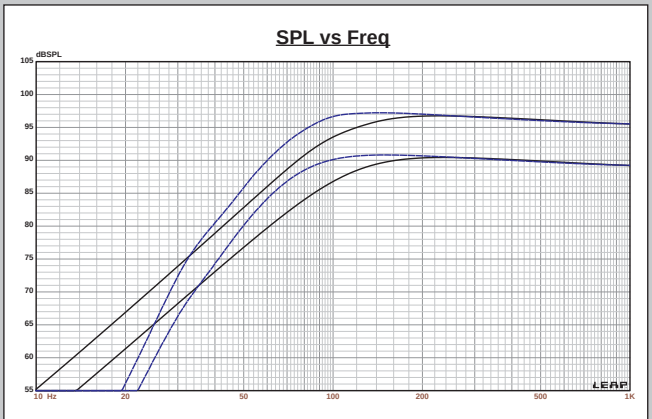


Figure 17: SB Acoustics SB12MNRX25-4 computer box simulations (black solid = vented @ 2.83 V; blue dash = vented @ 2.83 V; black solid = vented @6 V; blud dash = vented @ 6 V)

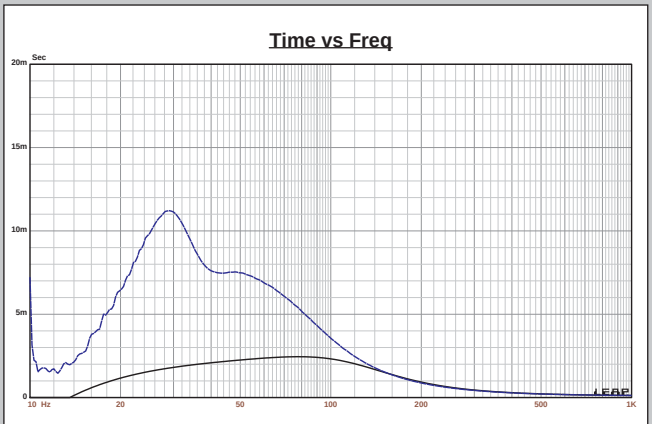


Figure 18: Group delay curves for the 2.83-V curves in Figure 17

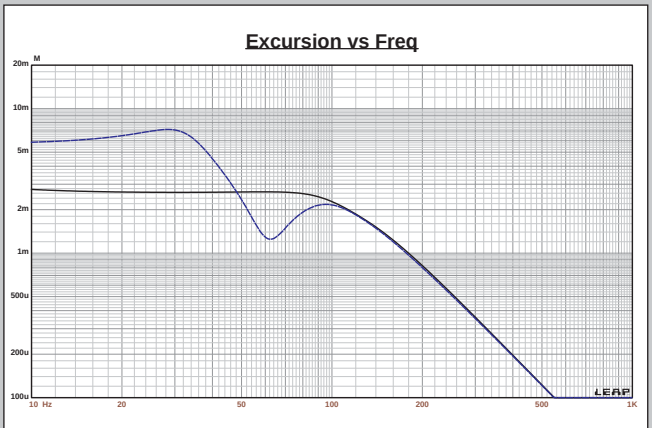


Figure 19: Cone excursion curves for the 6-V curves in Figure 17

placement limiting numbers calculated by the Klippel analyzer for the SB12MNRX were XBl @ 82% Bl = 3.3 mm (1.05 mm greater than the physical Xmax!) and for XC @ 75% Cms minimum was 4.0 mm, which means that for the SB12MNRX woofer, the Bl is the most limiting factor for prescribed distortion level of 10%. **Figure 24** gives the inductance curves Le(X) for the SB12MNRX. This also very similar to the SB12NRX, with a very small inductance change across the operating range of 0.03 mH, indicating that the aluminum shorting ring does a good

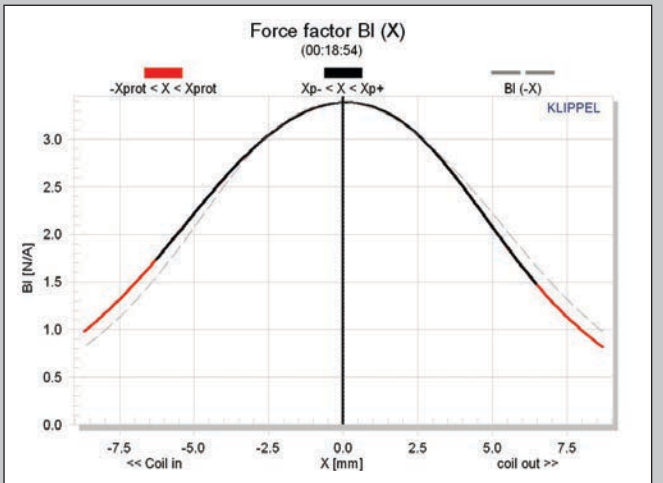


Figure 20: Klippel Analyzer Bl (X) curve for the SB Acoustics SB12MNRX25-4

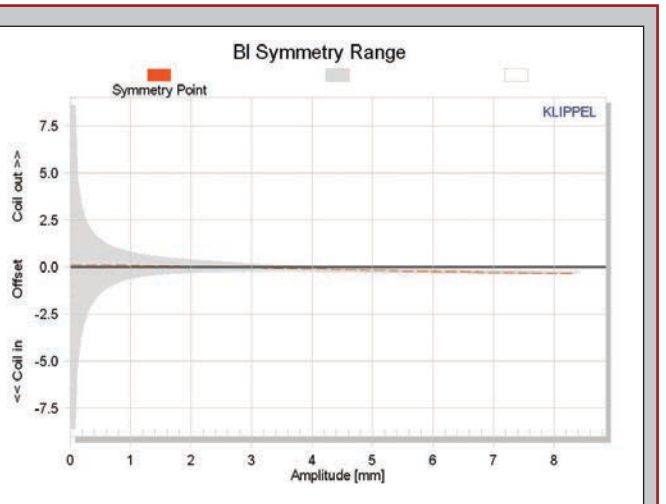


Figure 21: Klippel Analyzer Bl symmetry range curve for the SB Acoustics SB12MNRX25-4

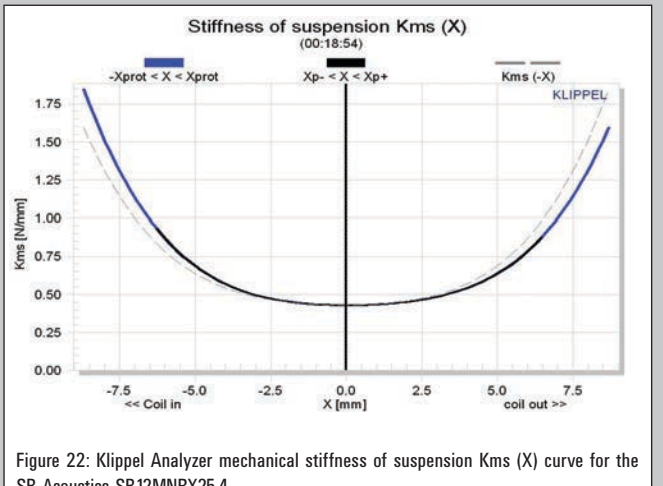


Figure 22: Klippel Analyzer mechanical stiffness of suspension Kms (X) curve for the SB Acoustics SB12MNRX25-4

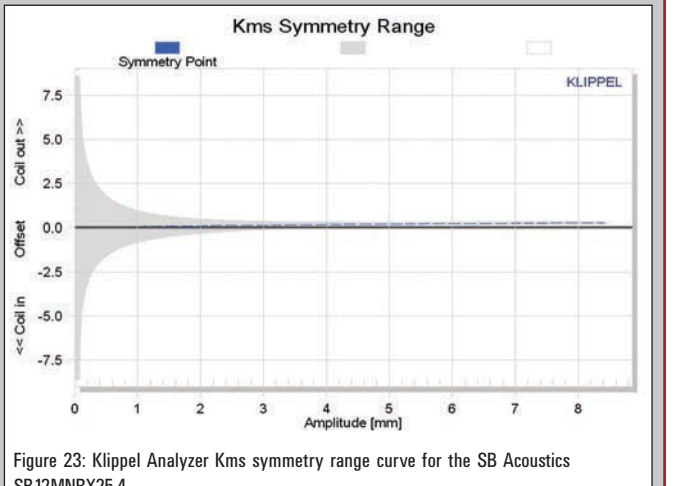


Figure 23: Klippel Analyzer Kms symmetry range curve for the SB Acoustics SB12MNRX25-4

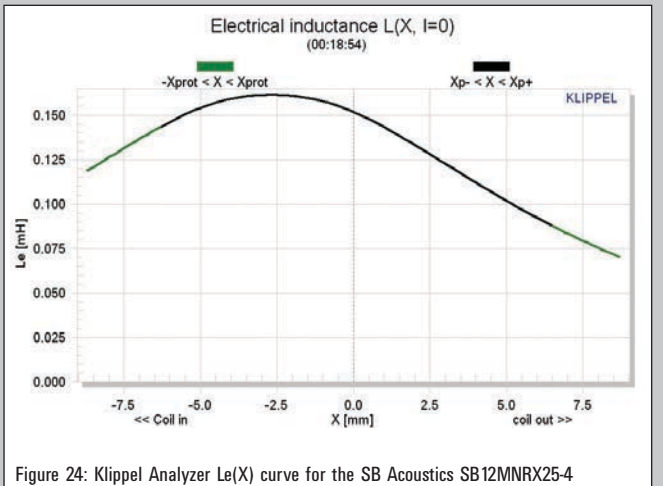


Figure 24: Klippel Analyzer Le(X) curve for the SB Acoustics SB12MNRX25-4

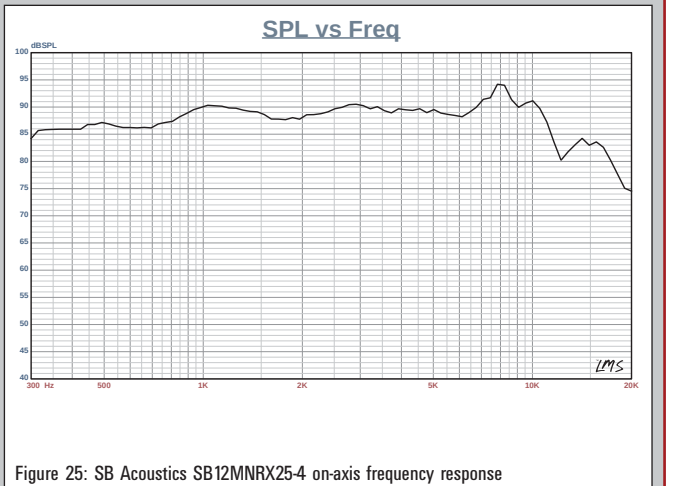


Figure 25: SB Acoustics SB12MNRX25-4 on-axis frequency response

job of controlling inductance.

With the Klippel testing accomplished, I mounted the SB12MNXR25-4 midrange driver in an enclosure which had a 13" × 6" baffle, which was filled with foam damping material, and measured the driver both 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 25** depicts the SB12MNRX's on-axis response displaying a very smooth rising response to about 6 kHz, with a 2-dB peak centered on 1.2 kHz and breakup mode

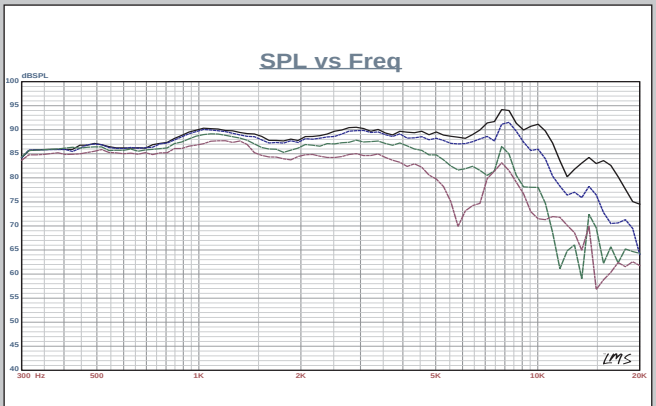


Figure 26: SB Acoustics SB12MNRX25-4 on- and off-axis frequency response at 0° (black), 15° (blue), 30° (green), and 45° (purple)

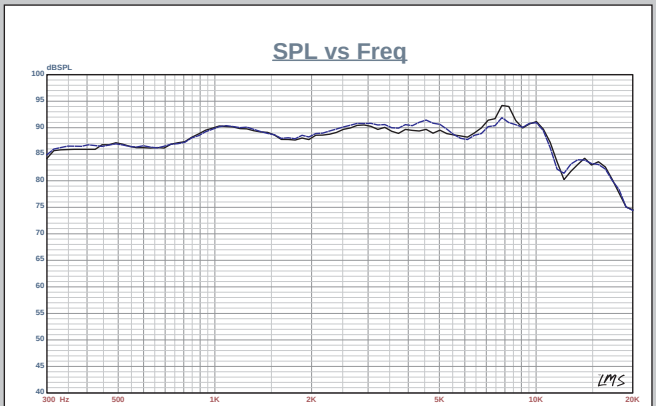
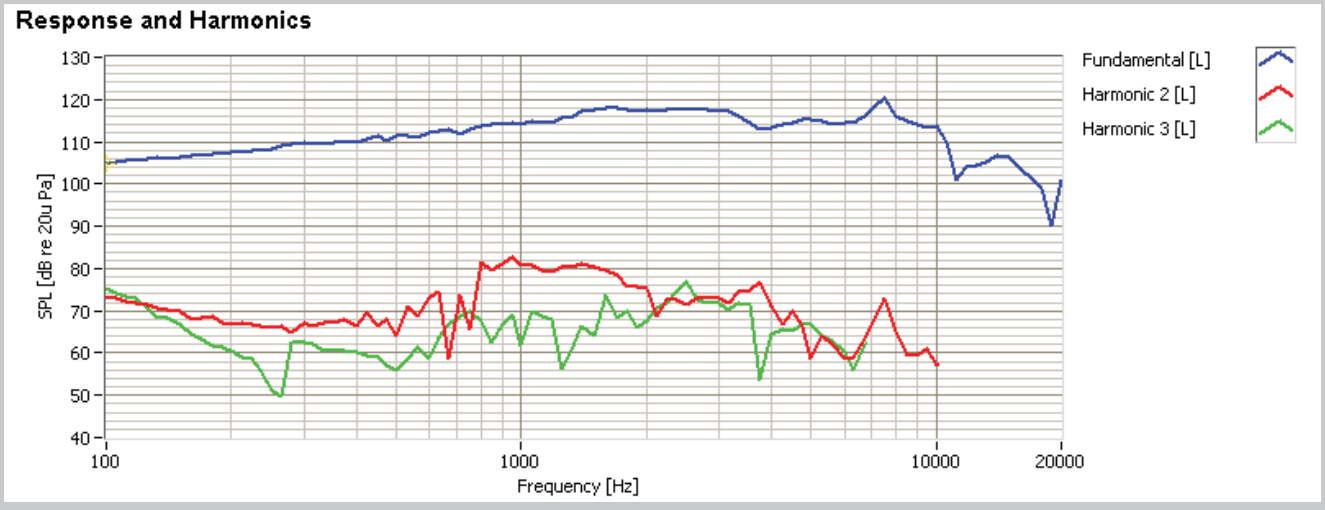


Figure 27: SB Acoustics SB12MNRX25-4 two-sample SPL comparison



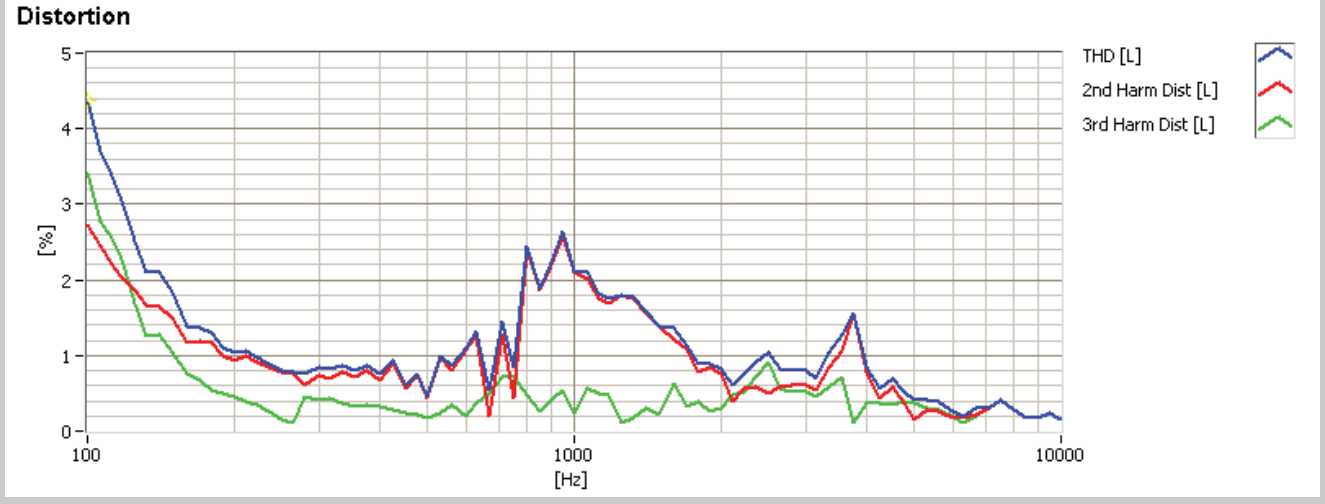


Figure 28: SB Acoustics SB12MNRX25-4 SoundCheck distortion plots

centered on 8 kHz. As you would expect, it was very similar to the SB12NRX. **Figure 26** has the on and off-axis frequency response at 0°, 15°, 30°, and 45°. –3 dB at 30° with respect to the on-axis curve occurs at 4.5 kHz, so a 4.5-kHz to 5-kHz crossover frequency would be appropriate for this SB driver. And finally, **Figure 27** gives the two-sample SPL comparisons for the 4" SB midrange driver, showing a less than 0.5-dB match up to 4 kHz.

For the last body of testing on the SB 4" midrange, I again fired up the Listen, Inc. SoundCheck analyzer (courtesy of Listen, Inc.) and SCM microphone and power supply to measure distortion and generate time frequency plots. Setting up for the distortion measurement again consisted of mounting the woofer rigidly in free-air, and the SPL set to 94 dB (5.27 V) at 1 m, using a noise stimulus (SoundCheck has a software generator and SPL meter as two of its utilities), and then the distortion measured with the SCM microphone placed 10 cm from the dust cap. This produced the distortion curves shown in **Figure 28**. Finally, 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 29** and the Wigner-Ville (for its better low-frequency performance) plot in **Figure 30**.

For more information, visit the SB Acoustics website at www.sbacoustics.com. *aX*

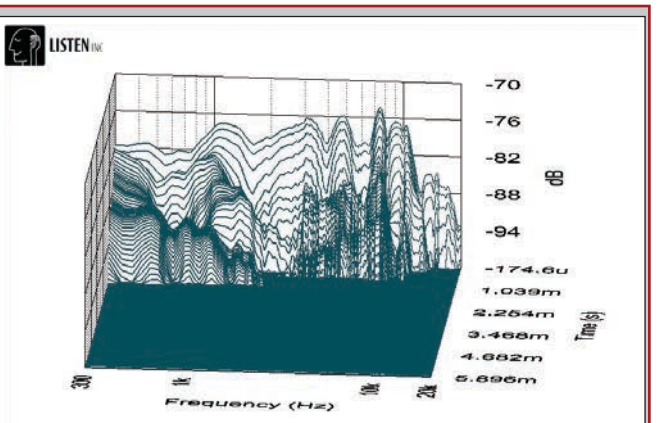


Figure 29: SB Acoustics SB12MNRX25-4 SoundCheck CSD waterfall plot

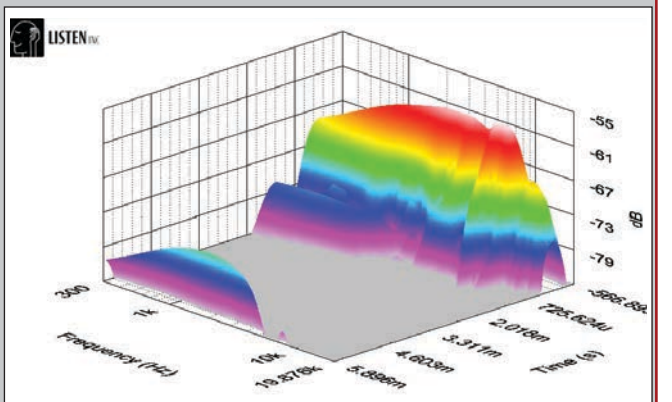


Figure 30: SB Acoustics SB12MNRX25-4 SoundCheck Wigner-Ville plot

By Bill Reeve (United States)

Restoring the HP 456A Current Probe

Once restored, the HP 456A performs as well as modern instruments.

The Hewlett-Packard 456A AC current probe (see **Photo 1**) is a treasure. It can be bought cheaply because many of the units sold were battery-powered and all were designed with a now-out-of-date oscilloscope interface connector. However, when restored, the 456A is a fabulous addition to any test bench, matching the performance of more expensive modern instruments.

Released as a new product by the Hewlett-Packard Company in 1960, the 456A was HP's first solid-state, stand-alone, clip-on current probe. It's elegantly designed amplifier uses two—then "state-of-the art"—PNP germanium transistors.

Photo 1: An HP 456A fresh from a surplus electronic equipment store. Note the lack of a power cord (indicating that it contains batteries) and the covered banana plug output connector.

THE ORIGINAL PROBE

In 1960, *The Hewlett-Packard Journal* (July-August, Vol. 11) proudly announced: "This new probe measures current over the full range of the frequencies most commonly used in typical work—25~ to 20 megacycles—and over an amplitude range from below 0.5 mA to 1 A rms…The probe operates with an accompanying small amplifier…to convert the AC current being measured to a proportional voltage. This voltage can then be measured with a suitable oscilloscope or volt meter. The current-to-voltage conversion factor is 1 mV/mA."

The 456A operating and service manual is available at www.hparchive.com, but this scanned copy contains incorrectly annotated schematic values for R7 (should be 3300 Ω), R8 (should be 2700 Ω) and C5 (should be 0.01 μF).

Old battery-powered 456As are usually in excellent physical shape because when their batteries ran down these instruments were often shelved and forgotten. Another 456A advantage is that its probe head is wired directly to the amplifier, so they cannot be separated by surplus electronics dealers.

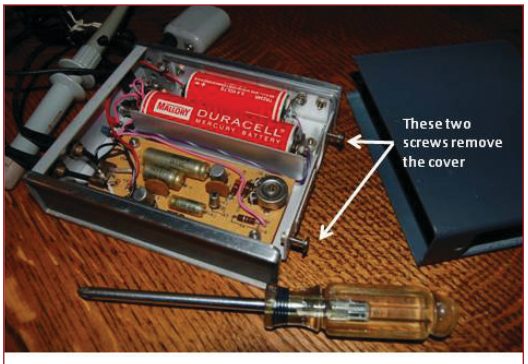


Photo 2: The amplifier cover (right) slides off the amplifier frame after removing two Phillips-head screws. The screws are shown re-threaded into the frame for safekeeping.

RESTORATION

Restoration of the 456A consists of three steps: replacing the old battery pack with DC power, restoring the amplifier electronics, and converting the obsolete oscilloscope banana plug interface to a BNC connector.

Step 1: Replace the old battery pack. Remove the two Phillips-head screws on the housing back to slide off the 456A's cover (see **Photo 2**). Re-thread the screws into the frame to keep them from getting lost.

Photo 2 shows the original batteries in place. The shorter battery provides +5.4 V (the shorter battery's positive terminal is toward the amplifier front), and the longer battery provides –8.1 V (the longer battery's positive terminal is toward the amplifier rear).

Obviously, the first thing to do is remove the batteries and dispose of them at a battery recycling facility. They contain mercury, and must be disposed of safely.

Optionally, at this point you can check the 456A's functionality. Using an adjustable dual-output bench power supply and clip leads, the correct DC

voltages can be applied to the battery terminals, as shown in **Photo 3**. You can view the 456A's output on an oscilloscope using a banana-to-BNC adapter. A signal can be easily obtained by clipping the probe over a loudspeaker wire.

The battery holder is easily detached by removing two flat-head screws on the frame's bottom side (see **Photo 4**) after unsoldering the red (+5 V), pink (–8 V), and purple/white (return) wires.

There are many ways to design a circuit that replaces the batteries. My parts selection (see **Figure 1**) was based upon what I had on hand. The only significant component constraint is that the dual secondary 12-VAC output transformer must be relatively small to fit within the cover. To select R1's value, I initially powered the circuit through a 1-kΩ potentiometer, adjusted it until the output was –8 V, and then selected the nearest standard value fixed resistor.

Photo 5 shows the battery replacement board installed in the 456A. I used the original battery bracket to hold the board. I removed the battery holders from the bracket by drilling out their rivets. The input power cord is protected by a rubber grommet and strain relieved with a tie-wrap. The vector board is mounted on the battery bracket with #4 screws. Two nuts on each screw serve as standoffs. I also coated exposed high-voltage points with hot glue.

Step 2: Restore the amplifier electronics. At this point, if you are happy with your current probe's performance, you can skip the following upgrades, but these are five modifications you might need to perform to get your 456A working or improve its performance:

- Replace the electrolytic capacitors
- Replace the two germanium transistors
- Replace the 8-V breakdown diode (CR1)
- AC-couple the output
- Flow solder onto the printed circuit traces

Photo 6 is an annotated close-up of the amplifier's single-sided printed circuit board. Following vacuum tube circuit convention, the +5 V is labeled "B+" and the –8 V is labeled "B–".

There are three electrolytic capacitors in the amplifier (see the horizontal silver cylinders in **Photo 6**), and their replacement is straightforward. **Table 1** lists substitute parts available from DigiKey.com. There are many other commercially available substitutes. Note

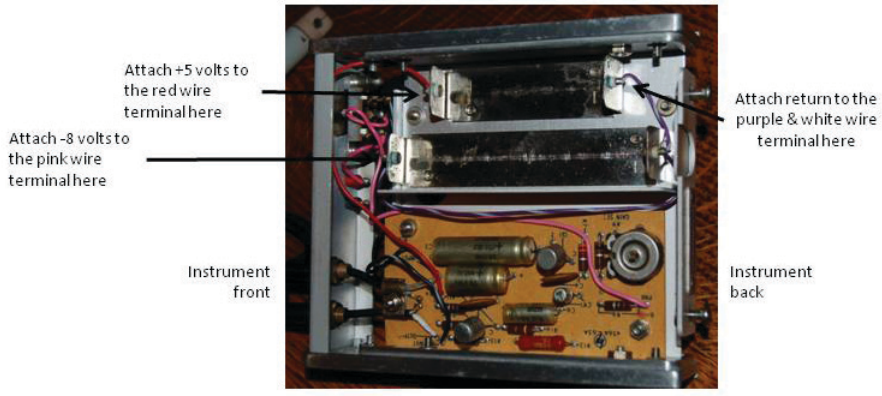


Photo 3: Optionally, after removing the batteries, you can check the 465A's functionality by applying DC power as shown.

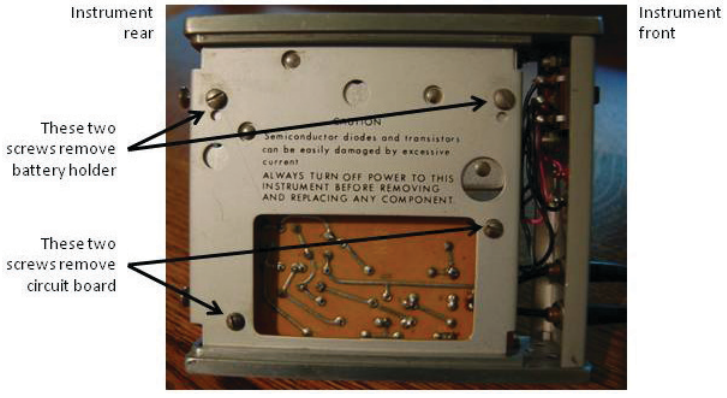


Photo 4: This is a bottom view of the amplifier with its cover removed showing two pairs of flat-head screws that mount the battery holder and the printed circuit board, respectively.

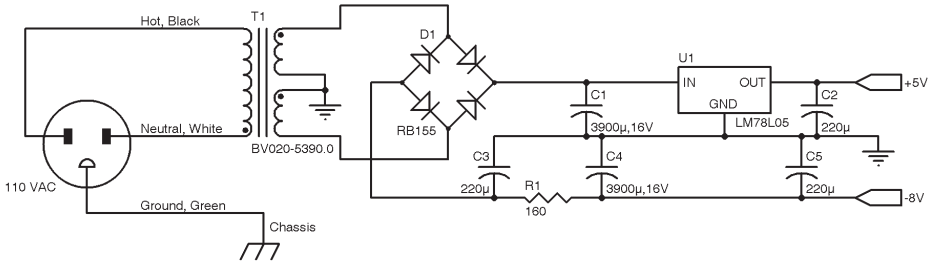


Figure 1: Battery replacement schematic

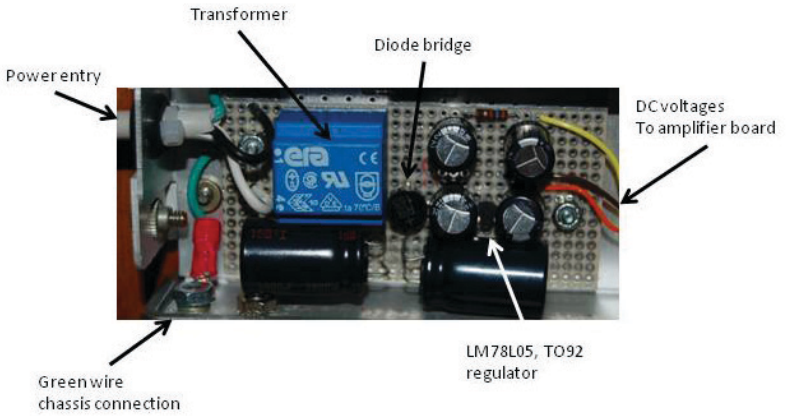


Photo 5: The battery replacement board

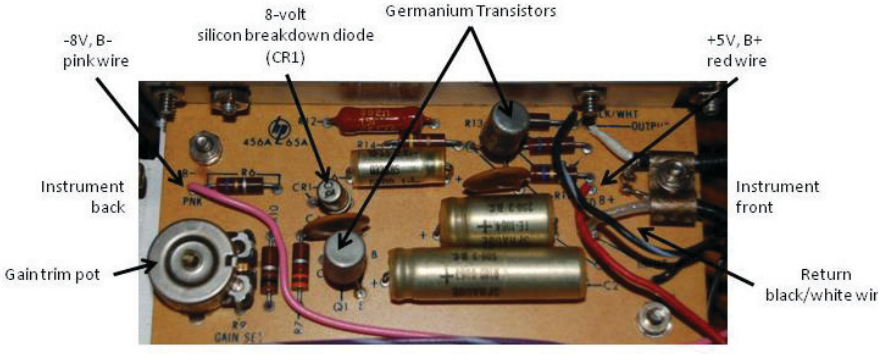


Photo 6: The amplifier's original printed circuit board

Reference Designator	Original	Replacement	Part Number
C2	200 µF, -10%/+100%, 3 V	220 µF, ± -20%, 16 V	Vishay MAL202135221E3
C3	500 µF, -10%/+100%, 3 V	680 µF ± -20%, 25 V	Vishay MAL202190528E3
C6	10 µF, -10%/+100%, 25 V	10 µF ± -20%, 63 V	Vishay MAL211838109E3

Table 1: Electrolytic capacitor replacement parts

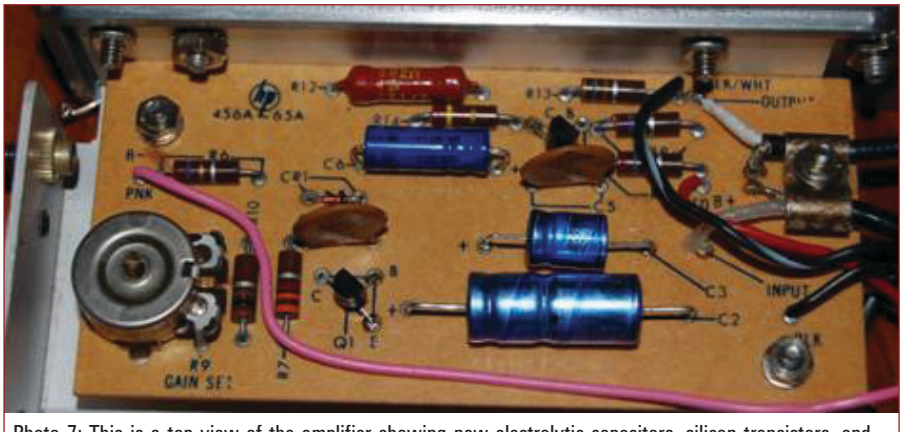


Photo 7: This is a top view of the amplifier showing new electrolytic capacitors, silicon transistors, and Zener diode (CR1) installed.

that these are all polarized capacitors, so solder them into the circuit with the same orientation as the originals.

Replacing the germanium transistors may not be necessary, but I have found that 2N3906 transistors (in TO-92 packages) are excellent, and commonly available, substitutes. The 1N5237 (8.2-V Zener diode) is a drop-in replacement for CR1.

Photo 7 shows the 456A amplifier board with its electrolytic capacitors, germanium transistors, and Zener diode replaced. The 456A was designed to have a 1-V output bias. I find this distracting, so I add a pair of parallel capacitors (220 µF and 0.1 µF) in series with the output signal, and a 51-kΩ pull-down resistor at the output, to AC-couple the signal. After power on, it takes a minute for the DC bias to drain off the output through the 51-kΩ resistor, but I think having an AC-coupled output is appropriate for an AC current probe.

As a final rework step, I recommend flowing solder over all the traces on back side of the amplifier printed circuit board. The original solder joints are old and their reliability is greatly improved by this reconditioning. **Photo 8** shows the refurbished amplifier ready for cover installation.

Step 3: Convert the oscilloscope interface to a BNC connector. This final modification can be performed one of three ways.

Pomona electronics (visit the website pomonaelectronics.com) sells a female banana to male BNC adapter (Model 1296).

Photo 8: The restoration is finished and ready for cover installation.

Photo 9: BNC output cable installed

You can cut the banana plug connector off the existing cable and attach a male BNC connector. This requires special tools.

You can replace the output cable with coax having one BNC end. This is a straightforward replacement. **Photo 9** shows the new BNC output cable.

If you have difficulties, e-mail me at billreeve@yahoo.com. This restoration should make your 456A ready for another 50 years of service. *aX*

Electroacoustics and Beyond

Shannon Becker discusses electroacoustics and more with aX's newest columnist

SHANNON: Where do you live? What city, country?

RICHARD: Lexington, NC, USA

SHANNON: What is your current occupation?

RICHARD: I am a consultant in acoustics, elecroacoustics, and sound/video design.

SHANNON: How long have you been reading *audioXpress*?

RICHARD: I have been reading the magazine since its inception.

SHANNON: You spent nine years as a part-time professional musician. What instruments do you play?

RICHARD: In order of competence: guitar (acoustic, classical, electric), electric bass, saxophone, trumpet, and bassoon.

SHANNON: Are you still performing and did your musical experiences feed your desire for better sound-system and speaker design?

RICHARD: I occasionally perform now, but not often. During my years in folk music, stage band, and rock bands, I developed an interest in speakers for live sound, and designed and built several systems, including all-horn systems (see **Photo 2**), mixed horn/vented systems, and line arrays. I also designed and built a twin-twelve tube guitar amplifier, and modified several other tube guitar amps.

SHANNON: How did your interest in the field of acoustics begin?

RICHARD: Knowing my lifelong interest in music, my dad brought me a copy of Leo Beranek's *Acoustics* from the library of Western Electric, where he worked. At the time, I was in the ninth grade. I read the book and was hooked for life.

SHANNON: What was your first acoustic design? Was it a speaker system design and did you actually build it?

RICHARD: It was a home-stereo speaker design using mixed horn and direct-radiator technology. I built two of them, and they are still in use in my living room, although I have a better design that will replace them when I have the time to finish building it.

Photo 1: Dr. Richard Honeycutt

SHANNON: For 19 years, you taught electronics at the college level. What specific courses did you teach?

RICHARD: I taught courses in Electronics Engineering Technology and Computer Engineering Technology: DC Circuits, AC Circuits, Semiconductors, Linear Integrated Circuits, Vacuum Tube Technology, Radio Communication, FCC Commercial License Preparation, Digital IC's, Microprocessors, CAD for Electronics, Electronic Construction Techniques, Programming for Electronics, Intro to Computers, Computer Repair, and also Physics.

SHANNON: What changes did you see in the field over that time?

RICHARD: The most significant changes were the virtual disappearance of vacuum tubes—except in audio applications—the eclipsing of discrete semiconductor circuits by ICs, and the change from traditional circuit boards to surface-mount technology.

SHANNON: How long have you specialized in electroacoustics?

RICHARD: Electroacoustics has been my major interest ever since high school. My senior research project in the physics department at Wake Forest University dealt with electret microphones. My first published magazine article (*The Audio Amateur*, 1972) was on microphones. The first paper I presented to a professional society (the North Carolina regional chapter of the Acoustical Society of America, Spring, 1971) was on modulation distortion in loudspeakers.

SHANNON: What do you see as the most significant innovations in the field of acoustics?

RICHARD: Here we have to distinguish between electroacoustics and the other areas, such as architectural acoustics. Both of these sub-fields of acoustics have benefitted greatly from

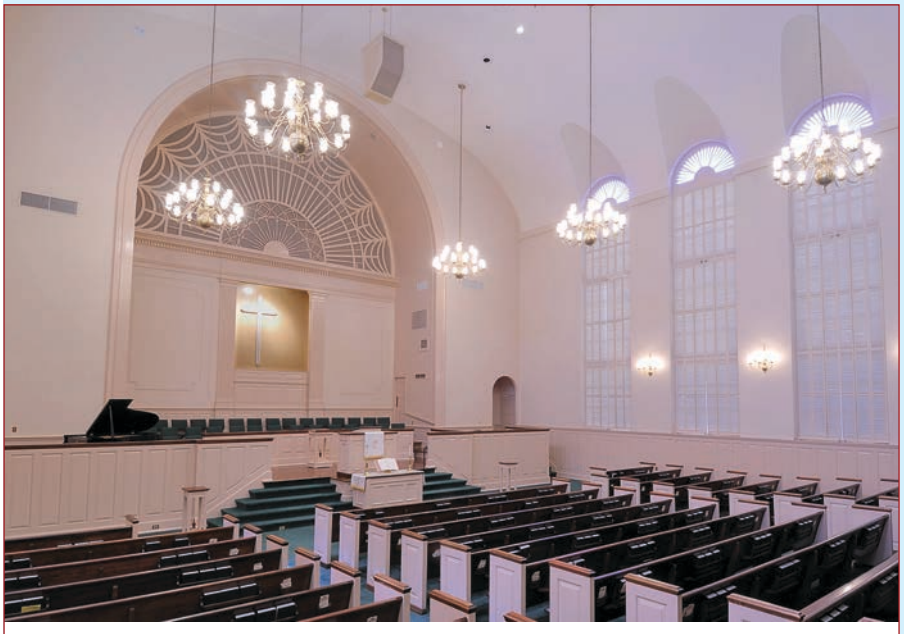


Photo 2: Custom All-Horn Speaker System Richard Honeycutt designed and built for the First Baptist Church of Lexington, NC

affordable computer modeling tools that run with reasonable speeds on accessibly priced computers. Part of my 1970 senior research project involved writing a program to compute room modes of our test chamber. It ran on a Hewlett-Packard rack-mount machine with 8 k of RAM, supported by a paper-tape reader. (Floppy disks were still in the future.) The program took about five minutes to run—for a small rectangular room.

In elecroacoustics, great progress has been made in understanding the behavior of horns, and in development of materials for cones, etc.

In architectural acoustics, wonderful progress has been made in quantifying the relationships among measurable acoustical quantities and listeners' preferences.

SHANNON: What prompted you to write your two books, *Electromechanical Devices: Theory, Applications, and Troubleshooting* and *Op Amps and Linear Integrated Circuits*? Do you have any other books in the works?

RICHARD: Both of those books were written in response to a need for comprehensive textbooks in their respective fields. I currently have a book on the acoustics of worship and performance spaces under review by publishers. On a vastly different note, I have a book on prayer currently in production for release in early 2012.

SHANNON: For 11 years, you were a contributing editor to *Voice Coil*. How did you get involved with the magazine?

RICHARD: When Ed Dell came up with the idea for *Voice Coil*, I had been writing for his publications for many years. He e-mailed me and asked whether I would be interested in serving as a contributing editor.

SHANNON: Can you show us or tell us about your own audio setup?

RICHARD: Tell, yes; show, no—showing would require too much time for neatening things up to make the system photogenic! My sources include a Philips turntable with an Audio Technica cartridge, a Sony TC-377 reel-to-reel deck, a Sony WE305 cassette deck, a Sony CDP-CE375 CD player, and a Panasonic DMR-EZ37V DVD/VCR. Processing, amplification, and video include a Panasonic SA-HE100 receiver, an

Photo 3: The Evenstar Peregrin

Photo 4: The Gabriel

Alesis RA100 subwoofer power amplifier, and a TV that my son and I bought at a yard sale about 25 years ago, that serves for video. I would prefer to use my tube-type H. H. Scott tuner for what little FM listening I do, as it sounds much better than the Panasonic receiver, but surround-sound decoding makes discrete-component systems too expensive at present. Also, I don't have space to locate the Scott in the audio area, nor does the receiver have any spare inputs.

My speaker complement features the system mentioned earlier that I built long ago, a center speaker of my own design (a modification of a studio monitor that was marketed as an Evenstar Peregrin (see **Photo 3**) for a few years, and a horn

subwoofer utilizing interleaved impedance loading (my invention that was the basis of my doctoral dissertation) that provides flat response down to 20 Hz (1/3-octave smoothing) in my listening room. The rear surround speakers serve their purpose, but do not really deserve to be mentioned, and I don't actually remember the brand.

SHANNON: Are you working on any projects? If so, what? Can you send us photos?

RICHARD: I continue to be fascinated by the promise of interleaved-impedance-loaded horns. The first full-range design that I completed using this principle was intended for the high-end home market, and was dubbed "The Gabriel." My first paper on interleaved impedance loading was presented to a professional society in October, 2000, and it concluded with a demonstration of the Gabriel prototype (see **Photo 4**) with a wide range of program material. The very positive response the design received has encouraged me to explore further designs. I am working on a pair of full-range all-horn speakers using interleaved impedance loading. Unlike the Gabriel, these will cover the range down to 50 Hz, making a subwoofer optional for many home systems.

SHANNON: Do you have a dream project? Imagine you have a nice budget and three months on your hands. What would you build?

RICHARD: Having provided the acoustical design for a THX-level cinema for a nearby university a few years ago, I would very much like to design a no-holds-barred THX cinema, preferably including designing all the speakers. The "nice budget" would have to cover obtaining THX approval for the new speaker designs. *aX*

By Richard Honeycutt (United States)

The Birth of Hollow-State Electronics

An introduction to vacuum tubes

The vital place played by vacuum tubes in the development of modern electronics is widely known. Even though they find a restricted range of applications today, the development of radio, TV, movies, home stereo, electronic computers, radar, and many industrial processes absolutely depended upon vacuum tubes. Less widely known is the story of the birth and infancy of the vacuum tube.

Photo 1: Vacuum tubes

EARLY TUBES

During the 1800s, scientists did much experimentation using glass envelopes that had been partially or completely evacuated of air, and in some cases, partially filled with another gas. Geissler tubes (such as neon lamps) and Crookes tubes were the first ones to become commonly used in laboratories. A Crookes tube, invented by English physicist William Crookes about 1870, consisted of a slightly conical glass enclosure from which as much air as possible had been removed. It had an electrode at each end, and when a high voltage was applied between the electrodes, something traveled from the cathode to the anode. This "something"—at first called cathode rays—was later discovered by J. J. Thompson to be negatively-charged particles. Later, these particles were named electrons.

In 1873, the English professor Frederick Guthrie discovered that a positively-charged, red-hot iron sphere in air would lose its electrical charge over time, although a negatively-charged one would not. This effect was called thermionic emission, but was not understood theoretically at first. In 1883, Thomas Edison was investigating why incandescent lamp filaments first darkened and then burned out at a point very near one of the terminal electrodes. In one experiment, he used a lamp that contained an extra metal support structure that was not connected to the filament. Upon connecting a galvanometer to that electrode, Edison discovered that a small current flowed in the vacuum inside the bulb if the unconnected metal structure was electrically positive relative to the filament. He patented the Edison Effect, but made no particular effort to understand it, and made little practical use of it.

British scientist William Preece delivered a paper on the Edison Effect bulbs in 1885, and physicist John Ambrose Fleming heard the paper. His interest piqued, he experimented with the Edison Effect and developed the Fleming valve, the first commercial vacuum diode, which he patented in 1904. The valve was first used as a radio detector, being more sensitive and stable than the previously-used coherers. In 1906, American engineer Lee de Forest independently invented a vacuum diode, which he patented. Two years later, he found that by adding a third metal structure called the grid, he could make the vacuum detector more sensitive still. Originally called the de Forest valve, or the Audion, we now know this device as the vacuum triode.

From the standpoint of today's understanding of physics, we describe thermionic emission as the neutralization of the positive charge of the hot iron sphere by electrons from gas molecules in the air. The Edison Effect is understood to involve the release of electrons from the heated filament (or cathode) under the influence of an electric field that draws them toward the positive anode (plate, in today's parlance). Current can flow in only one direction because the anode is not heated to free electrons.

Tubes can have either a directly-heated cathode (the filament itself is the cathode) or an indirectly-heated cathode (the filament heats a separate metal envelope which emits electrons). Most amplifying tubes have indirectly-heated cathodes.

TRIODES

In the triode, the grid may have a positive or negative electrical poten-

tial with respect to the cathode. Being much closer to the cathode than the plate, a positive grid exerts considerable pull on the cloud of electrons, accelerating them from the cathode. Since the grid has a small surface area (in modern tubes, it is a spiral of thin wire), most electrons do not strike it, but continue accelerating toward the plate. If the grid has a negative electrical potential with respect to the cathode, the opposite happens, and current from cathode to anode is inhibited. Thus, varying the voltage between the grid and cathode controls the current from the cathode to the plate.

If the grid is made negative enough that current in the tube is cut off, then a small AC voltage applied to it, such as that from a radio antenna, will cause current to flow on positive half-cycles, but no current on negative half-cycles. This "amplified detector" is the application that de Forest saw for the Audion. Today, the triode is most commonly used with the grid biased negatively with respect to the cathode, but not so negatively that current is cut off. Application of a small AC voltage between grid and cathode causes a current of the same waveform to pass through the tube. This current usually flows through a resistor in series between the positive DC supply (called the B+ supply) and the anode, creating an AC voltage drop across that resistor that can be from a few times to about 100 times as large as the AC voltage applied between grid and cathode. Thus, we have amplification of the AC voltage. The basic circuit is shown in **Figure 1**. In typical audio applications, the plate voltage of a triode is in the range from 100 to 300 V. Thus the voltage at the plate is—let's say—150 VDC with AC superimposed on it. The AC voltage is the product of the AC voltage at the grid × the gain of the stage. So if the stage has a gain of 80, and a 10 mV_{PP} input signal is applied to the grid, the plate voltage will be 150 V_{DC} with 800 mV_{AC} peak-to-peak superimposed. In other words, the plate voltage will vary from 149.6 V to 150.4 V, in terms of instantaneous voltages. Since the useful part of this composite voltage is the AC part, a capacitor is used between the plate and the output from the stage to allow the AC voltage to appear at the output terminals, while the DC voltage does not. In most cases, a capacitor is used to block DC from the grid, but this is for slightly different reasons.

This simple triode amplifier stage is only one of a number of circuits in which a triode can be connected to provide amplification. In future

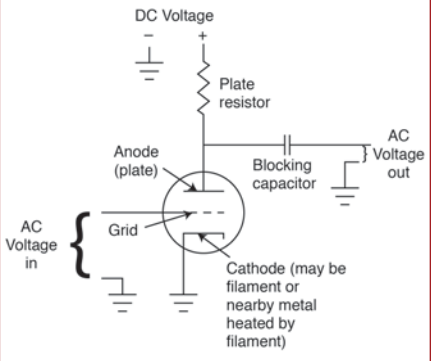


Figure 1: Basic triode stage

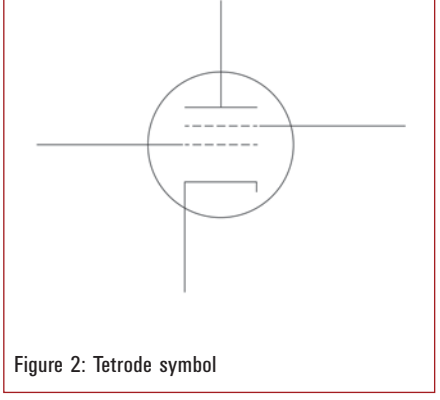


Figure 2: Tetrode symbol

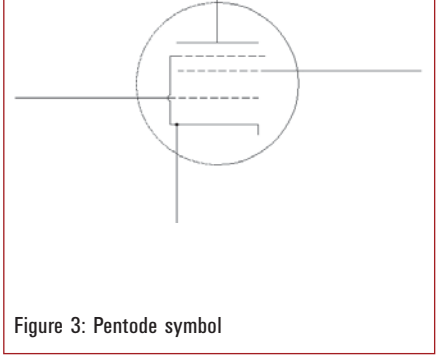


Figure 3: Pentode symbol

columns, we will look at some others, and their particular uses.

TETRODES

Walter Schottky, of the German company Telefunken, analyzed the triode to determine whether the gain could be increased. He decided that some of the energetic electrons striking the plate bounced off, creating a cloud of electrons near the plate. These are called the space charge. They neutralize some of the pull of the plate's positive potential, causing fewer electrons to flow from the cathode. Schottky suggested a screen grid located between the control grid and the plate. A positive potential on the screen grid overcame the repulsion of the space charge, allowing higher gain. Because of the small surface area of the screen grid, little current flowed to it. Schottky patented a tube using this principle, the tetrode, in 1916-1919 (sources vary on the exact date). The schematic symbol for a tetrode is shown in **Figure 2**. Tetrodes are seldom used in audio electronics.

PENTODES

Under certain conditions, the tetrode proved unstable, breaking into spontaneous oscillation and distortion. This problem was overcome by the addition of still another grid, dubbed the suppressor grid. The new, three-grid tube was called a pentode. Bernard Tellegen and Gillis Holst of Philips of the Netherlands, and Round of Marconi of England created pentodes about the same time: 1926. The schematic symbol for a pentode is shown in **Figure 3**.

Usually the screen-grid voltage of a pentode amplifier is somewhat (perhaps as much as 100 V) less than the plate voltage. This voltage is taken from the B+ supply via a screen dropping resistor. The suppressor grid is almost always operated at the same voltage as the cathode, and in many pentodes, is internally connected to the cathode, as indicated in the symbol.

Work by J. Owen Harries, Cabot Bull, and Sydney Rodda of England led to a modification of the pentode that could handle higher power. This beam power tube replaced the suppressor grid by shaped metal plates called "beam-forming electrodes." Since beam power tubes function as pentodes, they are usually represented by the same symbol as pentodes. (However, some technical writers refer to beam power tubes as beam power tetrodes, since they, like tetrodes, have only two grids, not three.) The development of the beam power tube occurred in the late 1930s, concluding the introduction of new varieties of vacuum tubes that are important in audio today.

TUBE FILAMENT TODAY

Today, tube filaments can be made of tungsten, which must be heated white-hot to emit sufficient electrons. Tungsten filaments can be impregnated with thorium oxide, which has a higher emissivity, and thus emits enough electrons at a lower temperature. Thoriated tungsten filaments glow bright yellow in use. Filament wire can also be made of a nickel-alloy wire with a relatively thick coating of alkaline earth elements (beryllium, magnesium, calcium, strontium, barium), which only have to be heated to a dull red glow to emit sufficient electrons. Naturally, tubes using filaments that operate at lower frequencies require less power and produce less heat during use. Indirectly heated cathodes enclose the filament in a metal sleeve coated with a high-emissivity material such as alkaline earth oxides.

During the evolution of vacuum tubes, the envelope of the tubes also evolved. The earliest tubes had no standardization in envelope style, shape, or size. The first standardized envelopes looked much like early light bulbs, with an oval glass envelope and a Bakelite base. Connections were made through four or five metal pins protruding from the base. These were about 4.5" to 5" long. Later variants included the octal (eight-pin base tube), loctal (seven- or eight-pin base), and "miniature" (seven- and nine-pin base versions). There were also a number of tubes, designed for battery-operated equipment, that used cylindrical glass envelopes less than 0.5" in diameter, and had flexible wire leads for soldering to the rest of the circuit.

In the next column, we will look at the basic operation of triode and pentode amplifiers. *aX*

By David J. Weinberg (United States)

AES 2011
Multitrack on Sound and Hearing

A review of the 131st AES convention

The 131st Audio Engineering Society (AES; www.AES.org) convention took place in the midst of Roger Furness's departure from his position as AES executive director (that took effect at the end of December 2011, although he will remain as a consultant through a transition period), the search for a new AES executive director, newly-elected AES officials assuming their duties, and reports of a slight increase in AES membership to approximately 14,700 (not counting sustaining members, the most in the organization's history).

Even so, the event offered much to learn that was related directly to audio and hearing, of which this report provides a mere sampling.

The AES reported that there were approximately 16,000 attendees (down from approximately 18,000 at the 2009 convention, the most recent previous one held in New York) and 310 exhibitors (just under the count in 2009). Several companies' representatives were happy with the "numbers and quality" and "qualified traffic" at their booths.

Convention program details are accessible via www.AES.org/events/131/.

PRESENTATIONS AND EVENTS

The keynote, delivered by Dr. Charles Limb (associate professor, Department of Otolaryngology, The Johns Hopkins School of Medicine), was titled **"Sound, Hearing and Music: A Journey from the Ears to the Brain."** He explored "the physical and intellectual intricacies of musical creativity. … Through the application of rigorous scientific methods to the study of musicians, we have been able to get a glimpse of how this remarkable process of artistic creativity takes place in the brain."

There were 25 technical paper sessions comprising approximately 125 presentations, plus poster sessions (papers posted in the lobby of the presentation rooms with the presenter available for discussion).

• **"Differences between the EBU R-128 Meter Recommendation and Human Subjective Loudness Perception"** (Begnert, Ekman, and Berg; session P8-3; convention paper 8489) reported on their test. "Several sound stimuli with large loudness differences, representing five different types of broadcast program material, were normalized to have equal meter-measured loudness level. Subjects listened to pair-wise presentations of the normalized stimuli, which they subsequently set to have equal perceived loudness." Analysis of the results showed that subject perceptions matched the meter-normalized levels within ±2.82 dB.

The 12 **special events** included Phil Ramone talking about his work with Tony Bennett (who was scheduled but was not able to attend, attributed to tour fatigue), focusing on the his new album "Duets II;" a panel on "The Art and Science Behind Motown and Verve Catalog Reissues;" panels of platinum producers, mastering and recording engineers; a multiformat critical listening session; Graham Blyth's convention-traditional organ concert (this year at the Central Synagogue); and others.

Discussing Tony Bennett, **Ramone** said that Bennett did not like monitoring on headphones. He prefers the "artist performance" style recording, even in the studio, with no punch-ins or overdubs — rerecord the entire track. Ramone added that rarely has more than three takes been required to get the desired recording. Bennett (85 years old) is still passionate about music, performing, and recording. "He wakes up in the morning and can't wait to sing." For "Duets II" Bennett traveled to the other artists' locations, sometimes recording in their homes. Each take is unique, like jazz. His duet partners were prepared, but found that they had to adjust during each take to what they heard from Bennett. In a video clip, Bennett said, "I'm definitely never going to retire, definitely going to keep on performing, because I love it."

Five **engineering brief** sessions, comprising 22 presentations, addressed recording/production, audio engineering applications, signal processing, and aural perception.

• **"The Williams Star Microphone Array Support System"** (convention Engineering Brief EB2-8) is a lightweight, relatively inconspicuous, configurable and adjustable "kit" designed by Michael Williams (www.MMAD.info) for arrays from two to seven channels, plus it has the ability to add microphones above and below the array. It was specifically designed for the Schoeps Colette series mikes, since the capsules are mounted in the supports with the amplifier modules remotely located. The support system employs a circular hub, to which you can affix radial and vertical rods of different lengths (depending on the mikes' directivity patterns and the chosen spacing). The mikes, in their shock mounts, are attached to the ends of the rods and aimed as desired. Williams recommends using the Rycote Lyre anti-vibration suspension (part of Rycote's InVision Broadcast series) to mount the hub on the stand to reject handling noise and stand-borne vibration. See www.Posthorn.com/Micarray_williamsstar.html.

• **"Achieving Great Sound in the Age of the Loudness Wars"** (Bryan Martin and Adrian Carr; convention Engineering Brief EB6-18) takes as its theme: "Good mastering is directly related to the enhancement of musical dynamics from a well done mix that has not already been overcompressed. … It is time to use your ears because pictures of waveforms from DAWs are not yet able to show the dif-

ferences in sound." The authors believe too many mixes are adjusted (compressed, EQed) using waveforms more than their ears. They are of the opinion that there is a lack of understanding of proper operating levels, which are different in analog and digital equipment including recorders, converters, mix platforms and plugins, and that this "contributes to the ever-increasing levels of distortion within recorded music." While they correctly point out that "digital audio is a peak-based medium" and analog is not, they incorrectly state that "the analog VU meter is an RMS-based meter." (It is driven by a full-wave averaging circuit.) They make the obvious statement that proper gain-stage management is important in both analog and digital processing. Their major push was for VU metering, with no mention of LKFS, ITU-R BS.1770-2, or EBU R128, which have been shown to provide relatively more accurate perceived loudness measurements than the VU meter.

There were six **"hot lunch and lunchtime keynotes."** The hot lunch events gave attendees the opportunity to chat with noted professionals in small gatherings on specific topics, including Peter Mapp ("Acoustic and Audio Apps") and Ronald Ajemian ("Demystifying Fiber Optics for Audio"). Lunchtime keynotes included Skip Pizzi ("The Future of Broadcasting in a Connected World") and Karlheinz Brandenburg ("The MP3 Story").

One of the three **historical sessions** featured Robert Auld discussing "Leopold Stokowski and the History of Analog Recording," which included photos plus audio and video clips of Stokowski's activities, including his involvement in Disney's *Fantasia* and his encouragement of quadraphonic recording during the 1970s. Stokowski was involved in recording from 1917 through 1977, "virtually the entire period of recording music by analog technology. He was frequently in contact with engineers and researchers who were working on technical advances." Auld played acoustic and early electric Stokowski recordings with and without EQ to show how the sound can be improved, part of a Glenn Gould interview of Stokowski, plus a 1932 dual-groove stereo recording (which predates Blumlein's stereo recordings). Stokowski's first movie role was a non-speaking part in *The Big Broadcast of 1937*. In his next movie he had a small speaking part (the first movie overdub).

Nineteen **workshops** dealt with "File Formats for Higher-Resolution Audio," "Capturing Height in Surround," metadata, "Sound Quality Evaluation," neodymium, "Authentication of Forensic Audio—Then and Now," "The Mobile Generation of Music Creation and Production," "Loudness Wars: The Tides Have Changed," "Listening Professionally or Train Your Ear," and more.

• "Sound Quality Evaluation" (Workshop 4) was evidence that interest in listening tests continues, and that there is still much to learn about their design and data analysis. "This workshop aims at providing researchers with a better understanding of what happens during listening tests and giving guidelines for a better listening test practice." Issues discussed included impairment score (when the subject thinks the processed choice sounds better than the reference; a data point that is rejected [I believe it should be included, or at least acknowledged in evaluating the subject's performance]) and the hedonic scale (a nine-level rating scale [I was told that at 11 levels, the scale is perceived as linear, and the nine-level scale accommodates those, like me, who don't find five levels to be fine enough gradation.]). Dolby's Poppy Crum presented the possibility that non-parametric data analysis could lead to more accurate conclusions, especially when the data does not conform to a Gaussian distribution and when relatively few data points are taken. I asked if a comparison had ever been made with real-world data to determine if non-parametric analysis would have led to a different conclusion (and therefore a different business decision) than the parametric (traditional) analysis that was performed? I followed that by asking if the resultant different business decision would have led to a better business success. Panel members generally felt that non-parametric statistical analysis gives correct results in many situations. Crum said that when multiple data sets are being combined, non-parametric tests involve fewer assumptions and provide a more robust measure. During the discussion of MPEG verification testing, I asked if applause or subtle movie sound effects had been used as test signals, and was told that applause and rain, which are especially tough to encode/decode cleanly, had been used during MPEG audio coder algorithms' development, and that a single localized effect was fairly simple to encode/decode cleanly.

• **"Listen Professionally or Train Your Ear!"** (Workshop 19) Paraphrased from the workshop description: "Considering the growth of ear training in the audio and music communities, it is important to discuss opinions from experts about necessary features and methods that assist trainees in acquiring critical listening ability with efficiency." Two sites with online training are www.SmackMyPitchUp.com and www.TrainYourEars.com. There is evidence that learning in a group setting yields more overall improvement in critical listening performance than learning on your own. Yamaha engineers and factory staff receive University of Kyusyu's expert listening training. All other employees receive basic listening training. Harman's Sean Olive listed four characteristics of critical listening: timbre, spatiality, dynamics, and distortion.

Eighteen **tutorials** covered effects delay; ear training; sound reinforcement acoustics; fiber optics; the Library of Congress's national jukebox (www.LoC.gov/jukebox); basic archival preservation; "Telephonometry: The Practical Acoustics of Handsets, Headsets & Mobile Devices;" audio metadata; and more.

• Some of us felt that one tutorial ("After Market Power Cords — 'Snake Oil' or Legitimate Audio Accessory") should not have been included in the program. At least one attendee filed a formal protest to the AES to that effect. No scientific evidence was provided, nor did the presenter, who had a booth on the exhibit floor, acknowledge that if a power cord (decently made with conductors of adequate gauge for the current it will carry) affects the sound quality delivered from the equipment, then there is something wrong with the equipment grounding or its power supply design/construction.

• **"Noise On the Brain — Hearing Damage on the Other Side"** was delivered by Poppy Crum, who talked about signifi-

cant effects in higher brain areas that can be caused by acoustic and chemical stimuli. Crum talked about research in this area, including how this "damage manifests throughout the auditory pathway as changes in hearing sensitivity, cognition, and the experience of tinnitus."

There are three primary indicators for hearing loss: Age, exposure to ototoxic drugs, and exposure to loud sounds. Studies have led to the conclusion that we all have hearing damage and hearing loss, some of which might have been preventable.

Research within the past decade has revealed that:

1. Most noise-induced hearing loss (NIHL) and age-related hearing loss result from chemical imbalances that effectively poison cochlear cells;
2. Dietary changes (especially antioxidants—one source is Vitamin C) and an increased SIRT3 level (one source is reduced caloric intake) have been shown in some species to help limit presbycusis (gradual hearing-acuity loss—more so with increasing frequency above 1 kHz—with advancing age) and NIHL, plus possibly partially reverse some immediate conditions of noise-induced trauma;
3. There is hope for tinnitus sufferers (promising clinical studies outside the U.S.); and
4. Substantial advancements have occurred in the use of stem cells, pharmacological treatments and cochlear implants toward preventing and correcting hearing problems.

The brain performs substantial processing on the cochlear response to sonic stimuli, and through innate and learned means overcomes some of the ear's limitations.

Recent research has found that acoustic trauma not only can damage the ear's mechanisms, but can cause auditory-cortex cell death throughout the auditory pathway, all the way up to the auditory cortex. This damage affects the brain's ability to analyze/categorize sounds, including speech comprehension, and can play a significant role in the experience of tinnitus.

Hypoxia and reduced O_2 concentration can amplify the effects of acoustic trauma.

The OSHA standard for permissible SPL(A)-vs.-time exposure doesn't take into account much of the new understanding of our hearing system's potential for damage at points beyond the middle ear. Changes beyond the cochlea do not always show up clearly with standard audiogram measurements. "In some cases, thresholds and audiograms can look completely normal but listeners might have great difficulty hearing a conversation in a noisy environment. … Additional means of determining the effects of noise exposure might be warranted."

A majority of damage beyond the cochlea is a result of some form of cochlear abnormality. Protecting the cochlear hair cells helps protect higher-level audio processing.

"Some simple dietary measures and awareness of the surrounding environment beyond the noise level can have potentially crucial effects on hearing protection."

Four two-hour **master classes** discussed "The Fine LIne Between Voicing and Design" of a D/A converter, human hearing, "Superstar Sessions," and "The Hybrid Edit/Mix Approach to Feature Film Sound."

Dolby sponsored the **Broadcast and Media-Streaming Audio Conference**, comprising 17 sessions addressing facility design, listener fatigue, TV loudness and metadata, lip sync, content streaming and networks, file management, and an opportunity for applicants to take Society of Broadcast Engineers certification exams.

Thirteen **live sound seminars** on DSP algorithms, wireless systems and RF band planning/coordination, networks, power and grounding (presented by Bill Whitlock), subwoofer arrays, beam steering, etc.

Thirteen sessions on **game audio** looked at trends, audio tools, game-audio secrets, sound processing plugins, speech recognition control, plus education and careers in gaming audio.

The 12 **product design sessions** talked about noise and its control/elimination, audio system grounding, Ethernet and Bluetooth for audio, loudspeakers at high amplitudes, analog circuitry, alternate measurement methods, and more. As part of this track, manufacturers offered presentations and training.

• "De-integrating Integrated Circuit Preamps" was presented by Les Tyler (President and co-founder, THAT Corp; www.THATCorp.com). Tyler covered some of the history of standard IC preamp configurations. Dating back as far as the early 1980s, he delineated some of their benefits (wide bandwidth at high gain, can be very quiet at high gain, easy to control with a single resistor, wide input dynamic range) and some of their detriments (feedback network adds noise at low gain, smooth taper is hard to achieve, must convert single-ended output to differential to drive A/D converters). THAT's approach is to "de-integrate" the IC design—remove: the output differential amp, the internal feedback resistors, the external differential gain resistor—leaving an IC (THAT 1570) that is completely configurable with differential in/out and low voltage-noise. This facilitates customized optimization for the parameters that the user considers important. His presentation explained optimization approaches for noise; bandwidth v. gain; gain control, taper, and DC offset; plus output and common-mode rejection. The slides are available at www.THATCorp.com/datashts/De-integrating_Integrated_Circuit_Preamps.pdf.

• "Is Your Equipment Design a Noise Problem Waiting to Happen?" (Product Design session 2), presented by Jensen Transformers' Bill Whitlock (www.Jensen-Transformers.com; where many of his papers can be found), addressed the balanced interface and why some equipment performs quietly in the lab but not in the field.

• "An Overview of Audio System Grounding and Interfacing" (Product Design session 10) was also delivered by Whitlock.

Nine **technical tours** included broadcast/recording studios, performance arts centers, a pipe-organ recording session in St. Peter's Lutheran Church of Manhattan, and Charlie Morrow's Soundscspes at Marimekko.

At www.AES.org/students/blog/2011/8/131st-aes-convention is AES Education Committee Chairman John Krivit's report on the **student and education program** part of the convention. It included technical tutorials, stu-

dent recording critiques, student delegate assemblies, the education fair, the scheduled opportunity for students to talk with "professionals representing a full spectrum of audio careers," and more.

• Academic Institutions offering studies in audio (from short courses to graduate degrees) offered information on their programs during a table top session. There was no charge for schools/institutions to participate, plus admission was free to all convention attendees.

• The **Education Forum Panel,** "Teaching the Teachers—A Round Table Discussion Among Audio Educators" was moderated by Alex Case. The concept of the session was how and what to teach the teachers, and how to reach the teachers. This session was for brainstorming, and to stimulate attendees to offer ideas then or later, to improve audio education, which will improve the field for professionals. They are not looking to develop a specific curriculum, but to generate ideas that faculty can use. A plea for input from anyone with ideas and interest: "Help us design the next steps for increasing our productivity, accelerating our innovation, enriching our camaraderie, and enhancing our quality as educators."

The **Career/Job Fair** featured several companies from the exhibit floor. Student and professional attendees visited with representatives to learn about job and internship opportunities in the audio industry.

For those AES members who wanted to get involved there were 22 **technical committee (TC) meetings** (www.AES.org/technical) plus standards groups (www.AES.org/standards) meetings. Technical committees "track advances in technologies and applications, reporting them in white papers, tutorials, AES journal articles, and AES website content. They also recommend important technical issues to the standards committees for further exploration. Technical committee topics included: semantic audio analysis, automotive audio, speakers, microphones, audio coding, archiving/restoration, digital libraries, electromagnetic compatibility, acoustics, audio forensics, hearing loss and prevention, human factors, and more.

If all of that can't keep you busy and overwhelmed with information and discussions, you're better than I.

Convention papers are available as preprints, or the set on CD-ROM, from www.AES.org.

About 200 hours of convention presentations are available as for-fee downloadable MP3 files from MultiView (www.SoftConference.com/aes/slist.asp?C=4579). The presentations are sampled at 16-bit/44.1-ksps PCM and converted to 128-kbps MP3 files, which are recorded. Postproduction performs slight editing (mostly to find the beginning and end of the session) and converts these files to 24-kbps MP3 files for distribution by download or CD-R. When necessary to fit an approximately 90-minute session on an 80-minute CD-R they use Sound Forge to time-compress the 128-kbps MP3 file, with pitch correction, before converting it to a 24-kbps MP3 file for the CD-R. I would like the AES to offer the session recordings at a lower cost, especially to students, to make the knowledge more accessible to those who can't afford to attend the convention and broaden awareness of the AES's value.

EXHIBITS OF INTEREST

I don't spend time at very many equipment and software exhibits, as we can get most of that information online.

The **Audio History Library** (Louis Manno, director; 212.988.0642) is an "educational charity serving as the audio industry's archive. [They] are collecting paper and electronic data on every manufacturer internationally over the history of the industry, and are cataloging this information into a database. As an Internet digital library it will service professionals, audiophiles, and historians." They collect equipment literature, spec sheets, photos, manuals, and some equipment.

E.A.R. (www.EARinc.com; www.FreeHearingTest.com; www.HearDefenders.com; www.FreeFitCheck.com; 800.525.2690) sells an assortment of hearing protection products and in-canal earphone accessories.

Jan Didden started **Linear Audio** (www.LinearAudio.net; "Your tech audio resource") in 2010 "to help spread technical audio information and knowledge in printed format." While most of the industry concentrates on digital, Didden focuses on the analog audio domain, since all audio starts and ends as analog wavefronts, and almost all audio is exposed to some analog circuitry. He has published four issues (each approximately $30): volumes 0, 1, and 2, plus a special issue titled "Baxandall and Self on Audio Power," which "reprints classical papers." This 120-page issue includes "Peter Baxandall's *Wireless World* article series on audio power amplifier designs" from the late 1970s, "Douglas Self's *Wireless World* series on power amplifier design" from the mid 1990s, and 35-pages of letters, never published before, from Baxandall to Self "commenting and expanding on Self's articles." Didden also has his personal professional website: www.LinearAudio.nl.

• The **Bob Moog Foundation** (www.MoogFoundation.org) was created after his death in 2005 by his family and colleagues, and is headed by Executive Director Michelle Moog-Koussa (Moog's third daughter). Officially launched in August 2006, the foundation "is to ignite the innovator in us all through exposure to interactive educational experiences that draw from Moog's legacy of

ground-breaking work at the intersection of music, science and imagination." They have begun archiving Moog's writings and project notes, as well as electronic music instruments and breadboards, all of which will be exhibited in the Moogseum (planned). The foundation offers education to children (K to eighth-grade level) at Dr. Bob's SoundSchool, and has curricula and teacher training materials available. Information about Moog and his work (including the invention of the Moog synthesizer that dramatically changed almost every genre of music) is available on the foundation's website.

The Museum of Sound Recording (in New York; contact SoundRecordingMuseum@Gmail.com for information) booth handed out a copy of Emile Berliner's "His Master's Voice" trademark #34,890, registered July 10, 1900. The trademark is of the dog (with tilted head) listening at the horn of a record player. Berliner's statement and declaration states that the famous caption "is unimportant and may be omitted, since the essential feature of my trademark is the picture of the dog listening to the sound-reproducing machine." It's not clear from the copy, but it appears that the name Nipper is printed on the dog's collar.

The National Music Council of the United States (www.MusicCouncil.org) represents the U.S. to the International Music Council/UNESCO and is "dedicated to strengthening the importance of music in our life and culture, … promoting and supporting music and music education as an integral part of the curricula in schools." Among other activities, the council offers suggestions on how to support and promote music education in your area.

Rane (www.Rane.com) handed out a booklet of four Rane Notes (white papers. These and others are available on their website): "Why DSP Boxes Set the Same Way Differ" (RaneNote 167; the premise is that different manufacturers use different definitions of filter bandwidth); "Bandwidth in Octaves Versus Q in Bandpass Filters" (RaneNote 170); "A Bessel Filter Crossover, and Its Relation to Others" (RaneNote 147); and "Second-Order Digital Filters Done Right" (RaneNote 157).

Rational Acoustics (www.RationalAcoustics.com), developers of the Smaart acoustic test and measurement software, handed out a 6"-wide oval sticker (see **Photo 1**). A T-shirt version is also available from their website store.

StudioShare (http://Audio.StudioShare.org) "is an online business platform where musicians and audio professionals can exchange equipment and services to fully utilize their resources and those owned by other members." Member accounts are $49/yr; studio owner accounts are $79/yr. An individual may try it for free using coupon code SOUND49.

Women's Audio Mission (WAM; www.WomensAudioMission.org; $40/yr standard membership; $30/students) "is a San Francisco-based non-profit organization dedicated to the advancement of women in music production and the recording arts. [It provides] training, experience, career counseling, and job placement to women and girls in media technology for music, radio, film, television, and the Internet." Membership offers access to WAM's recording studio; discounts on recording equipment, books, magazines and training; free training (local and online), etc. You can try a free sample course at www.SoundChannel.org.

Zaxcom's TRX900LTS "body pack" transmitter has a dual microphone input "for independent transmission of two isolated audio channels on a single RF carrier," built-in 96-hour backup recording with timecode stamp, and a five hour runtime on a single AA lithium battery.

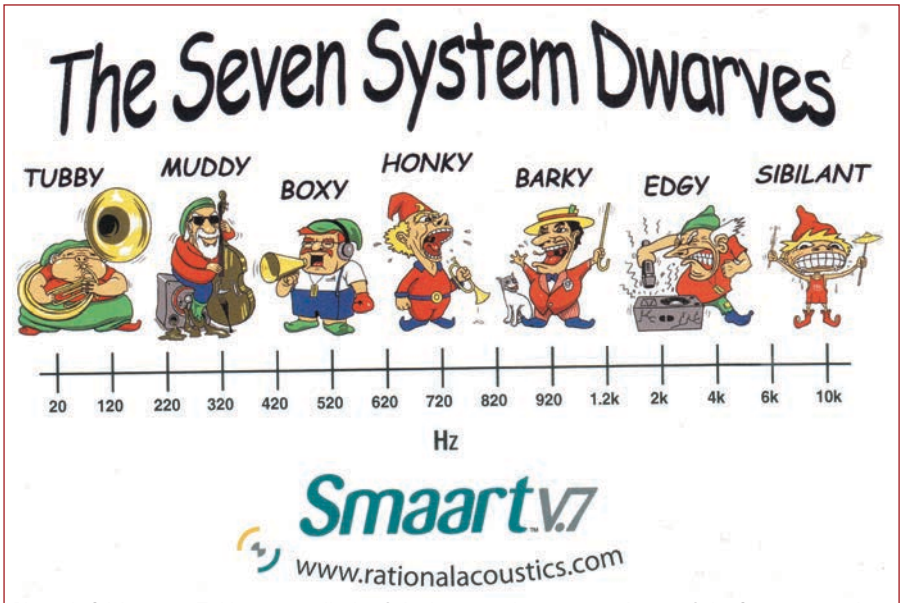


Photo 1: Stickers and T-shirts are available of the image designed by developers of the Smaart acoustic test and measurement software.

EPILOGUE

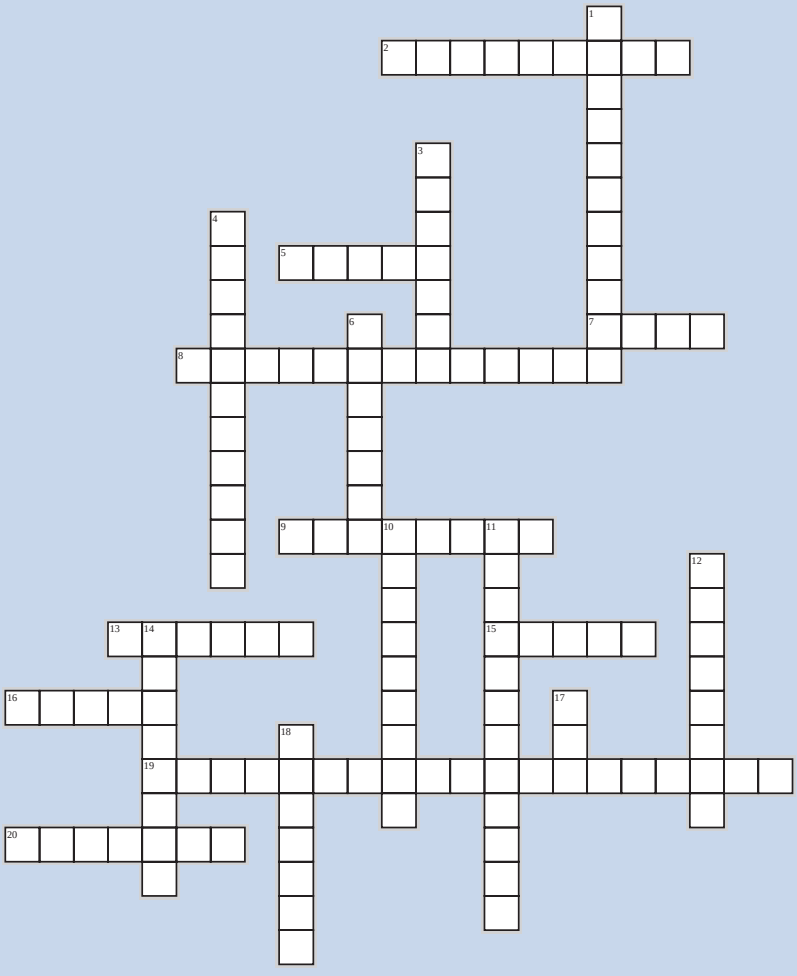
The Javits convention center was undergoing a long-awaited renovation, which led to the exhibits and registration being at the north end of the building, while the sessions were at the far south end. The AES convention did not use most of the available space. Even once the reconstruction is completed, I wonder if a smaller facility wouldn't be more suitable for a convention of this size, especially if it is outside of New York, which is expensive for attendees.

The 2012 AES convention will be held October 26 to 29, 2011 at the Moscone Center in San Francisco, CA, where there will likely be a session on otoprotective drugs.

Another AES event worth considering is the 47th conference ("Music-Induced Hearing Disorders: New Technologies for Measurement and Protection," Chicago, IL; June 20–22; an information webpage was posted in December).

A reminder: If you become an AES life member, society membership and convention attendance are free.

It is likely that with new officers in charge of the AES the society will change. It is equally likely that the AES conventions will continue to adjust to our needs while retaining strong educational and informational value to members and students. *aX*



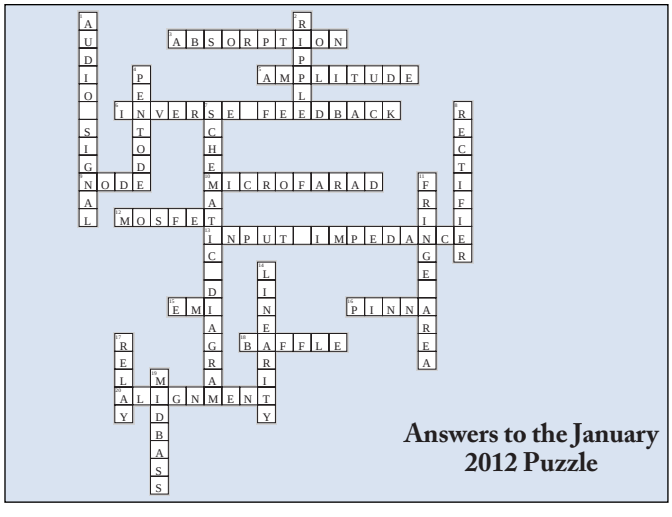
Editor's Note: Answers to this puzzle will appear in the next issue.

ACROSS

2. To reduce the intensity of an electrical signal
5. The opposite of flatness
7. A group of adjacent digital bits comprising a data unit
8. Having a nominal impedance of 600 Ω or less
9. Signal energy that is returned from the output of a device or system to its input
13. Any quantity having direction
15. The number of cycles of repetitive change which a sound wave or electrical signal undergoes in one second
16. The SI unit of energy and of work done by that energy
19. The heat-induced flow of charge carriers from a surface or over a potential-energy barrier (two words)
20. A vacuum tube having four elements: cathode, plate, control grid, and screen grid

DOWN

1. Any active device in which electrons flow between two electrodes through an intervening evacuated space (two words)
4. A pressure-sensitive switch requiring a very small force to actuate it
6. An abrupt, extremely brief burst of signal energy
10. The type of coaxial loudspeaker in which a separate tweeter cone is mounted on the front of the woofer's voice-coil form
11. Electrons that are emitted from gases trapped inside a glass tube from which most of the air has been evacuated (two words)
12. A multiple of any fundamental frequency
14. A negatively charged part of an atom
17. The unit of resistance, reactance, or impedance
18. That region of a bipolar transistor into which electrons are introduced to the semiconductor material



Answers to the January 2012 Puzzle

Outline's GTO-DF is designed to provide exceptionally high-quality sound.

NEW GTO-DF SPEAKER CABINET FROM OUTLINE

Outline, the maker of high-quality, industry leading speaker systems for touring, installation, and studio applications, highlighted the GTO-DF, the newest member of the Grand Touring Outline (GTO) Series of line array speakers during AES 2011. First introduced at PLASA 2011, the GTO-DF is designed to provide exceptionally high-quality sound to the first several rows of the audience at an event, filling the sonic gap often experienced by event attendees who are seated near the stage.

The unique new waveguide was designed at the request of Jason Farah, co-owner and vice president of Special Event Services in Winston-Salem, NC, who identified the need to address the issue of downfill coverage from a new perspective. Outline's engineering team looked at the problem and designed a technically superior solution to simply tagging on an Outline Mantas speaker to the bottom of the array. The GTO-DF, with its purpose built, radical new horn design and acoustic lens, delivers accurate performance within the form factor of a GTO array. Special Event Services was one of the first companies to adopt the GTO Line Array system and the first to deploy the GTO-DF with simultaneous debuts of the cabinet on both the Darius Rucker Tour and ZZ Top/Lynyrd Skynyrd Tour, both with the desired effect.

GTO-DF is an eye-catching enclosure, with Outline's exclusive acoustic lens aperture gradually flaring out downward. The GTO-DF offers four 8" mid-woofers and two 3" compression drivers (the standard GTO line array cabinet features 10 speakers with two 15" woofers, four 8" mid-woofers and four 3" compression drivers). GTO-DF is mechanically compatible with GTO, so it can be easily added to a flying array. Unique to the GTO-DF is a vertical dispersion range of 0° to –25° to ensure precision aiming of the sound into the audience and not onto the stage. The exclusive waveguide offers 120° of coverage from 315 Hz to 17.5 kHz.

The GTO-DF, a bi-amped system, can handle 800-W AES and 3,200-W peak for the mid-woofer section, covering the 200-Hz to 1-kHz range. The high-frequency section handles 250-W AES and 1,000-W peak, covering a 1-kHz to 17.5-kHz range. The cabinets weigh in at 172 lbs (78 kg) each.

For more information, visit Outline at www.outlinearray.com.

NEW RF MODULE

Radiocrafts AS, a leading provider of compact RF modules, has expanded its product line with a new module complying with the KNX-RF Multi specification. KNX is the only open international standard for Home and Building Control, used in Smart Home, Building Automation, and Building Management Systems.

RC1180-KNX2 is the world's first RF module, including a complete KNX-RF Multi protocol stack.

The KNX-RF Multi is an extension of the European Norm for building automation adding redundancy and increased reliability. The main features of the new Multi standard are: frequency agility by using up to five frequencies, fast link acknowledgement of up to 64 receivers with automatic retransmission, and multi-hop repeaters extending the range by two hops. Battery-operated transmitters and receivers are also supported by the new standard. The embedded protocol is backward compatible with KNX-RF 1.1 and KNX Ready, and can be used for unidirectional and bidirectional devices.

The new module is designed for sensors, actuators, and other home and building automation equipment. Due to its small size, easy-to-use interface, complete embedded protocol, and low power consumption, it can easily be integrated into any product making a very cost-efficient solution.

Radiocrafts is a member of the KNX Association and has participated in the development of the new standard. KNX is one of the leading standards for home and building control. The interest for such systems is increasing, meeting the demands for energy saving technology. Studies have shown up to 50% energy savings using KNX technology.

Interoperation between Wireless M-Bus and KNX-RF is also possible, using the new module. The module operates in the 868-MHz band, using Listen Before Talk (LBT) and frequency agility to reduce collisions. Up to five frequencies are scanned and automatically selected. One receiver can be linked with up to 64 transmitters, enabling very large RF networks. The fast acknowledge and retransmission feature ensures link reliability. Complete repeater functionality is also built in the protocol stack, and can retransmit messages in two hops. It can be used with S, A, and E modes of installation. Among other features, the module offers automatic battery supervision and signal strength information.

The new RC1180-KNX2 is a surface-mounted, high-performance transceiver module measuring only 12.7 × 25.4 × 3.3 mm. A UART interface is used for serial communication and configuration. An antenna is connected directly to the RF pin. The RC1180-KNX2 module is certified for operation under the European radio regulations for license-free use. When used with quarter-wave antennas a line-of-sight range of 800 m can be achieved.

The module and Demo Kits are available now. The module is delivered on tape and reel for volume production.

For more information, visit Radiocrafts AS at www.radiocrafts.com. *aX*

The RC1180-KNX2 RF module

PEAKS DON'T ALWAYS INDICATE DC COMPONENT

I have greatly enjoyed Ed Simon's articles throughout the years. However, I must respectfully take issue with "Is There DC on Your AC?" (*audioXpress*, November 2011, p. 35). Unequal ±peaks do not necessarily indicate a DC component on a waveform. Viewing any live music waveform will show this (no DC, very unequal ±peaks).

Consider summing two pure sine waves: 60 Hz, 117.58 V RMS, and 120 Hz, 24 V RMS. The sum will be a 60 Hz, 120 V RMS wave with 20% 2nd harmonic distortion. There will be a variety of waveform shapes, depending on the relative phase between the 60 Hz and 120 Hz components. I've sketched two possible waveforms in **Figures 1a** and **1b**. (They're not to scale but are qualitatively correct.)

In **Figure 1a**, the zero-crossing points of the 60 Hz and 120 Hz since components (not shown) are synchronized. Note that the rising and falling portions are dissimilar ("lopsided"), but the peaks are equal (178.11 V).

Conversely, in **Figure 1b** the peaks of the two components are synchronized. Here, the rising and falling portions are symmetrical, but the peaks are unequal (+200.22 V and –132.34 V, a difference of 67.88 V). However, note that the time duration of the negative half-cycle is longer than that of the positive half-cycle. The areas enclosed by the + and – half cycles are equal, because there is no DC component. (Of course, because no DC was "put in the mix.")

What about nonlinear loads on an AC line? A diode in series with a light bulb will of course draw only one polarity of current; a half-wave rectified current has both AC and DC components. Now, if your house wiring and the line transformer secondary winding had zero DC resistance, no DC voltage could be present. But since they do have resistance, some DC voltage would be present.

I have two suggested ways to measure any DC voltage on your AC: 1. Use a lab-grade meter such as the Fluke 189 or 289, which can accurately measure DC with a much higher AC voltage present; 2. Insert a low-pass filter between the AC line and a cheap DC meter, with a series look 0.25-W resistor, followed by a shunt 10 µF film cap, the –3-dB frequency is 0.159 Hz. The 60-Hz attenuation is 377 to 1, or 51.5 dB, with 120 V RMS input, the 60-Hz filter output is only 0.318 V RMS, too low to nonlinearly influence even the cheapest DCV meter.

Dennis Colin
Gilmanton, NH

Ed Simon responds:

Thank you for your kind words and letter. You are correct it is possible to have a difference in the voltage peaks on an AC waveform while still not having a DC offset.

The more interesting mechanism is from the use of AC induction motors. Such a motor will try to lock to the AC line and produce an output speed of either 1800 (or 3600) rpm for a 60-Hz line frequency. Due to what is called slippage or loose coupling the actual speed is often 1780 rpm or lower. This slippage will cause the current draw from an AC line to vary an introduce a 0.33-Hz modulation (or higher) on the 60-Hz AC line. This will show up with the measurement method shown as a swing between positive and negative readings on the meter. The average between the values will be the DC peak difference.

I have found that kind of line distortion to be common. The other common type is induced when items such as hair dryers or heat guns are used on their low setting. They use a diode in series with the heating element to reduce the heat, thus, producing a DC offset on the power line. I have not noticed the type of issue you bring up when I have had to look for power line problems and am interested in what mechanism would cause it.

As to the cases you mention, the issue is what do these true AC waveforms but with unequal peaks do to the power supply in actual equipment? As you are aware with a diode capacitor filter the current is only drawn when the AC line is at a peak. In the extreme case you presented, the capacitor may actually only charge on one peak of the AC line. The result is that the power transformer would behave almost exactly as if there were a DC offset in the AC supply line. That is why the discriminator circuit shown would caution the user that the AC line had problems even though there was theoretically no DC offset present. A true DC offset reading would miss the problem!

I have the problem of trying to present the issue in a manner that is easily understood and thank you for the letter allowing me to expand on some of the other issues.

But, the bottom line is the circuit will show problem power lines and even allow the skilled user to rebalance their own power system to minimize it. *aX*

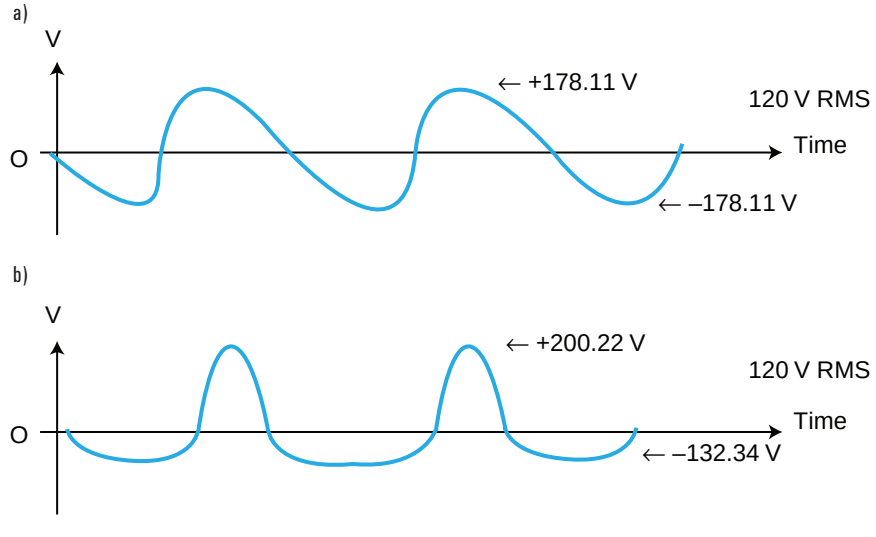


Figure 1a: Zero-crossing points of the 60 Hz and 120 Hz. b: The peaks of the two components are synchronized.

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 XPRESS-Mail section of an upcoming issue.

CONTRIBUTORS

Robert Nance Dee ("The 5002 Project," p. 12) is a retired electronics engineer. He worked with the first military computers, has several patents, and has written many magazine articles. He enjoys machining, watch making, bicycling, cabinet making, and philosophy. He is an amateur arborist, an environmentalist, and a vegetarian who thinks his life is a hobby. His passion for audio, in every form, spans 50 years and started in high school while working part time in a radio/television repair shop. He lives off the grid in a house he built, surrounded by animals and fruit trees in the Catskill Mountains. His audio projects are named after his animals and friends.

Vance Dickason ("Two 4" Drivers from SB Acoustics," p. 17) 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 7th 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 monthly.

Richard Honeycutt ("The Birth of Hollow-State Electronics," p. 33) has been an audio enthusiast, musician, recording engineer, broadcast radio engineer (FCC First Class Commercial licensee since 1969), radio announcer, electronics repairman, tube guitar amplifier designer, speaker system designer, pro-sound/video designer, and college electronics teacher. He is now a consultant in acoustics and electroacoustics. He possesses a Ph.D. in Electroacoustics from the Union Institute, and lives in Lexington, NC, with his wife Betty Jane, near his daughter Alyson, her husband, and two of Richard's three grandchildren.

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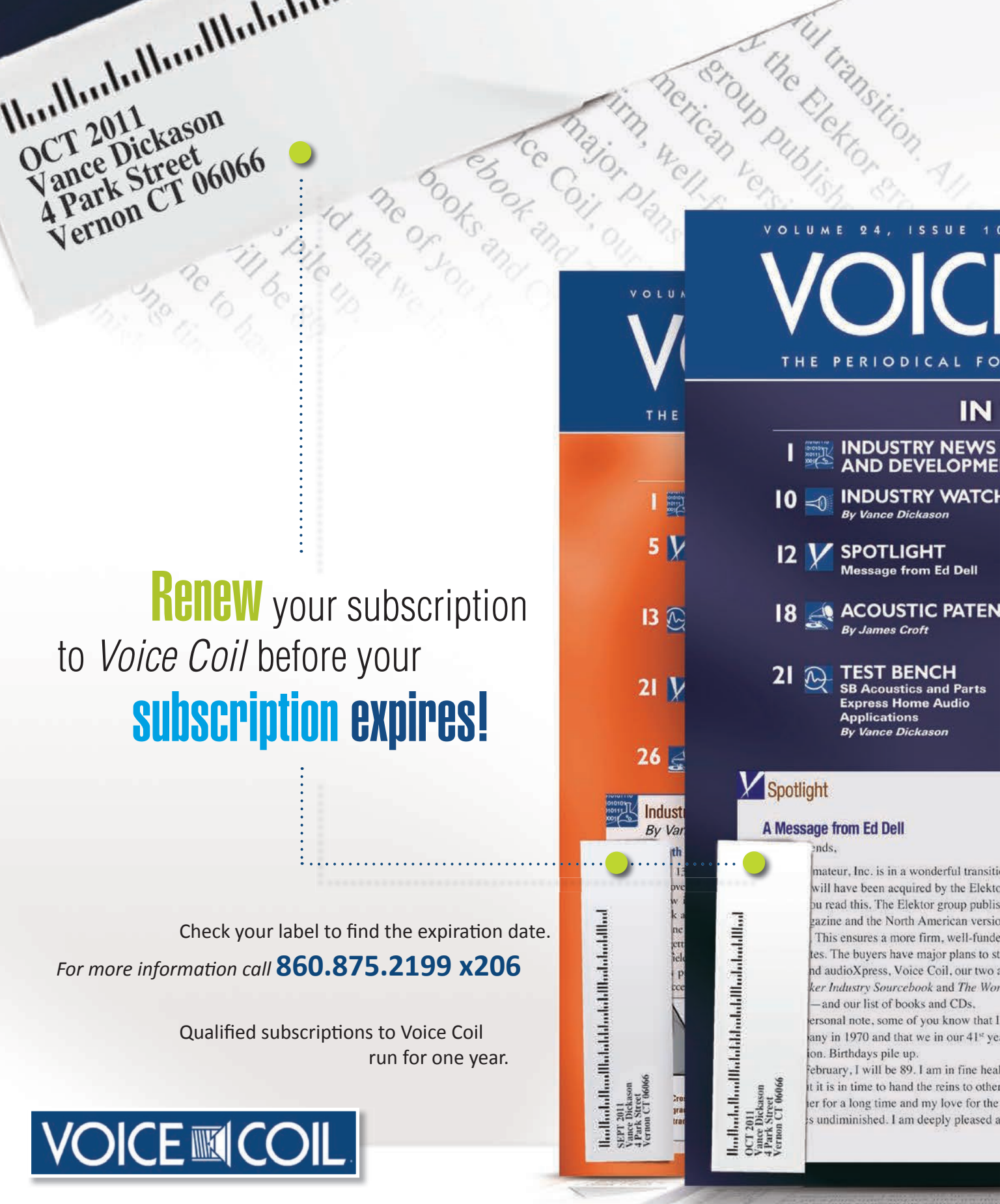

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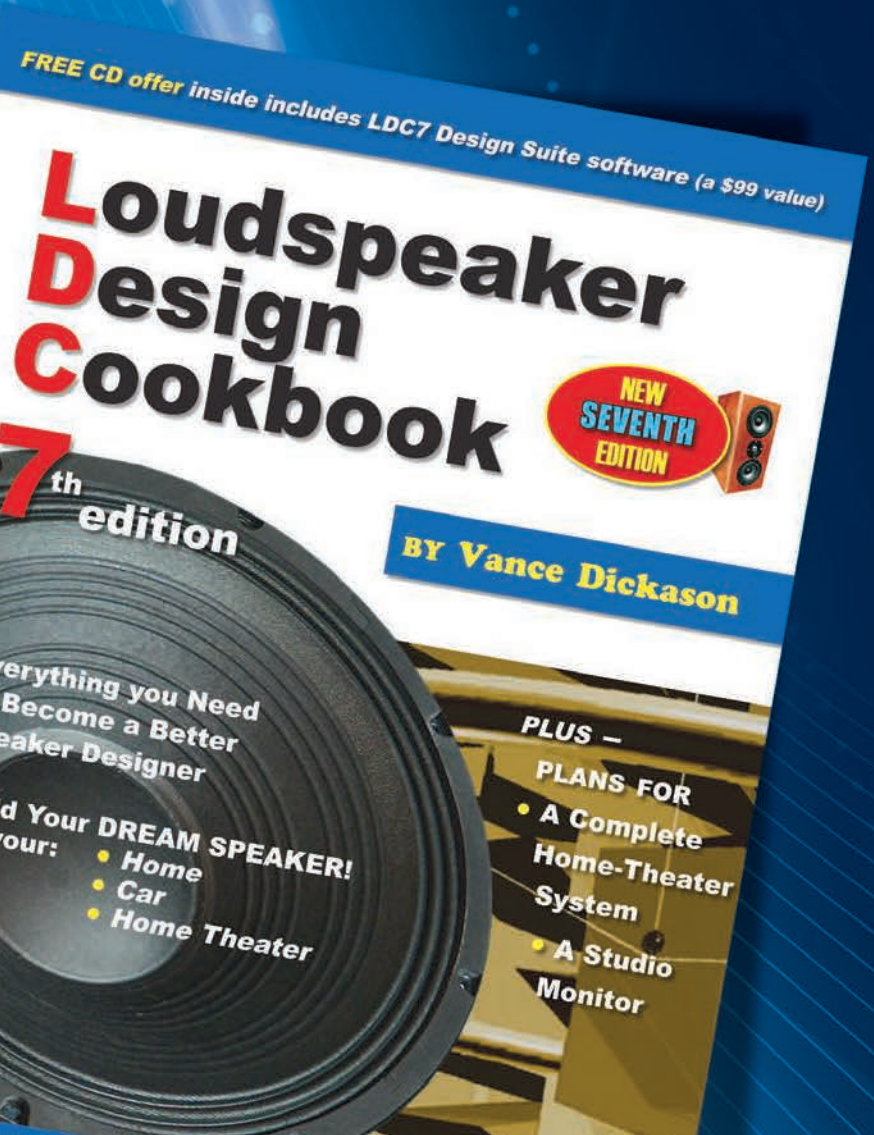

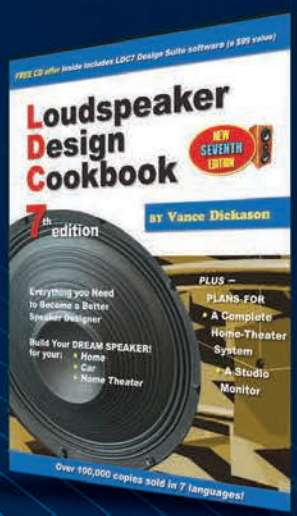

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