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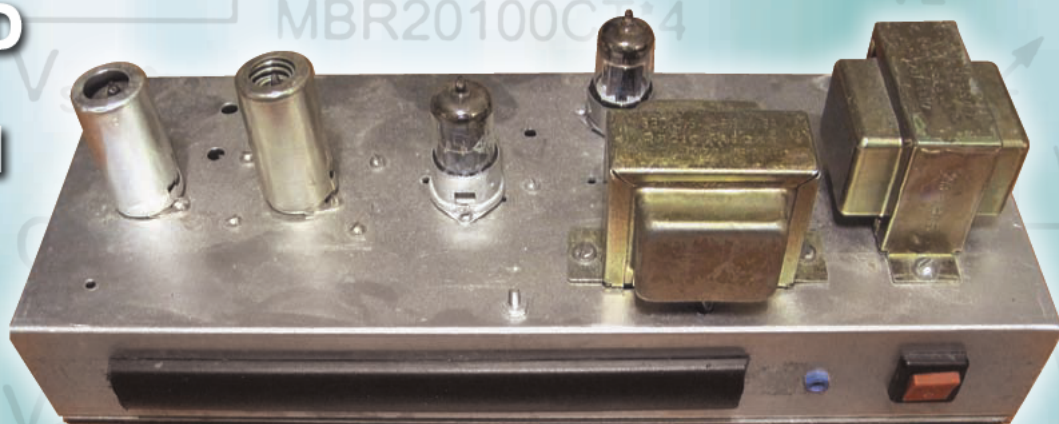
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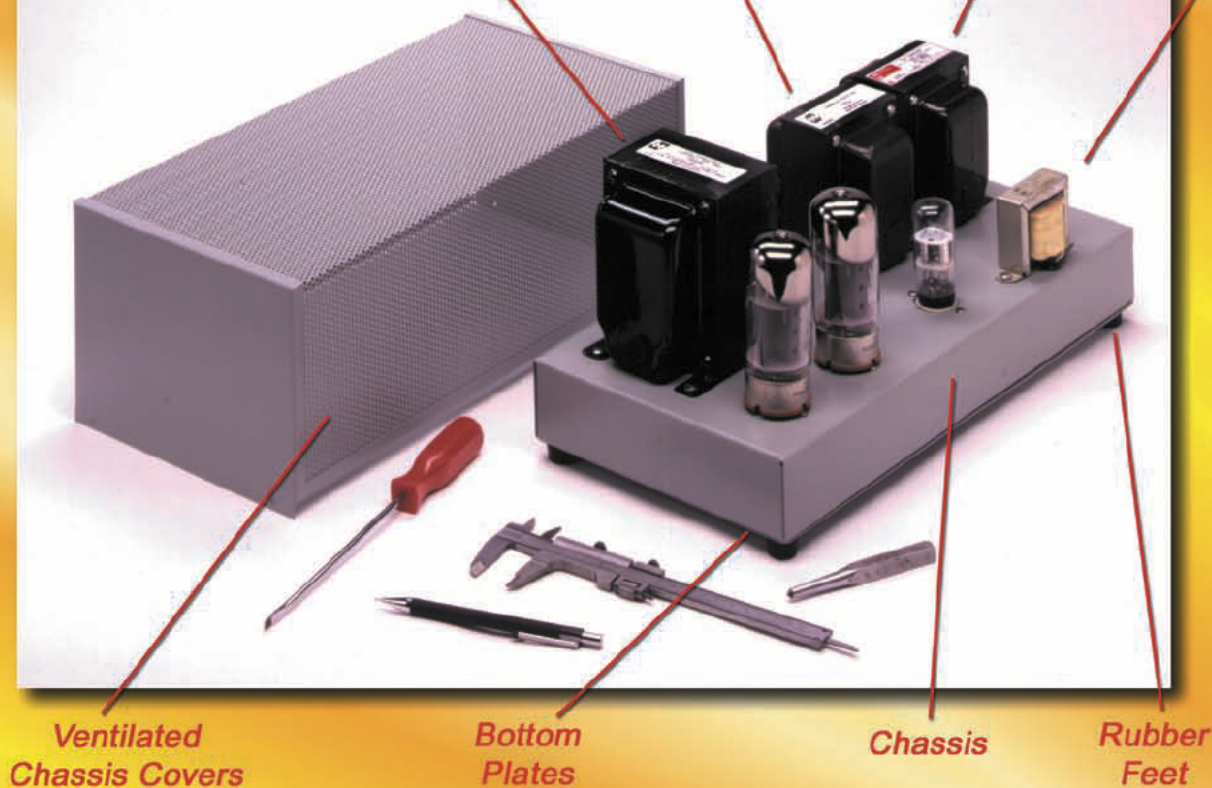
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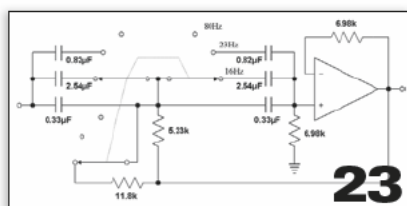
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# GOING DIGITAL, REVIEWS, AND A LONG LIFE

A warm thank-you is in order to all our subscribers to *audioXpress* monthly. We are now evenly divided between printed and digital versions. Please realize, all you paper lovers, that you are welcome to see the digital issues by merely supplying your e-mail address and a password. You can enjoy the digital edition a week earlier than the printed issue.

Please notice that the digital choice also preserves trees—our source of oxygen and the major way carbon dioxide is being taken out of the air. See Steven Johnson's really wonderful book, *The Invention of Air*, which profiles Joseph Priestly, who discovered the components of air, where it came from, and other items in the history of the breath we breathe. Your digital sub is a green move.

We note some comments on our article balance lately. In the future we will put less emphasis on consumer elec-

tronics reviews. The reviews we have do not convince advertisers that *aX* readers are a viable market for their products. Component manufacturers nearly all believe that you audiophiles don't buy their equipment, because you make your own. So our ad sales people have given up trying to convince them that you do not make your FM tuners, Blu-ray recorders, A-to-D converters, and other vital parts of your systems.

They also seem not to believe that many of you read *audioXpress* vicariously, for imaginative journeys through construction and testing of construction projects. This type of knowledgeable audiophile is, in my opinion, the best sort of customer. And apparently you as consumers have kept quiet about your purchases of hardware. We will, from time to time, do reviews of groundbreaking importance, and now we have room in the digital pages to do more extensive ones.

I invite all of you to consider writing up your construction projects and send proposals about them to me. Most of you seem to prefer *aX* articles to be hands-on hardware pieces involving side cutters and soldering irons. The other activity that would help us greatly is some evangelism about the joys of building audio as a craft. Converts would swell our ranks. You'd be telling most candidates about some talents they don't know they have.

Further, this sort of hands-on activity, according to recent scientific studies, reduces depression and contributes to longevity to those who are hobbyists of any kind, especially with manual activities involved. So for a serene and long life, get busy with your pet project. I urge you to respond to my suggestions at [ed@audioxpress.com](mailto:ed@audioxpress.com). Also, for other sidelights on how we're doing, go to [www.eddellsdecibels.com](http://www.eddellsdecibels.com).

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# The Wedge Slope-Back Speaker

These wedge-shaped speakers will fit right in with your sound system.



PHOTO 1: Finished Wedge pair, with the grille off on the right one.

**T**his system features the combination of a classic M-T-M system with a sealed woofer in an enclosure built from stock dimensional lumber. The sloping back reduces the profile somewhat and is not an unattractive addition to your home entertainment room.

The side-firing woofers allow you to use a large woofer while maintaining a narrow profile for the enclosure. The system is great for conventional hi-fi and can be used as the left-right speakers in a surround system. The 12" woofers may not need an additional sub-woofer for most surround-sound applications.

Each system incorporates two 6.5" drivers and a 1" dome tweeter in an M-T-M arrangement in the upper enclosure and a side-firing 12" woofer in the sealed lower portion of the enclosure. A simple 6dB three-way crossover handles the frequency separation. The drivers are not that critical and substitutions that meet the specs in the parts list will work just as well. Calculations showed that the woofer specified would achieve an F3 of 40Hz in the 1.6' enclosure with a Qtc of 1.5. The finished speakers are shown in **Photo 1**.

## CONSTRUCTION

Begin by laying out the cuts needed for the sloping side panels. They are cut from 36" x 48" x 3/4" one side finished birch plywood. On the plywood panel for the left enclosure, measure over 8"

from the top right of the panel and draw a line from that mark down to the left corner of the panel. Do the same on the plywood panel for the right enclosure, but measure 8" over from the top left of that panel. I used a clamping circular saw guide to cut the angled sides easily and accurately (**Photo 2**).

Remember that you will be making mirror images of the sides (**Photo 3**). Once the sides are out, cut the woofer holes. The size of the hole may vary with the woofer selected. The Goldwood woofer required an 11" hole. I used a Jasper Circle jig for all the driver holes, but a compass and jig saw will also do the job (**Photo 4**). I did not elect to cut a groove to flush-mount the drivers. If you want to flush-mount them, you must cut the mounting groove first before making the full driver hole. The Jasper jig is ideal for making these flush-mount cuts.

The front and rear panels can be hardwood or pine, as desired. I used clear grade pine for both front and rear and the inside partition. Cut the front panel to a length of 48" and then cut out the driver holes. Space the top driver hole 3" from the panel top and allow 2" between each driver hole edge. (This spacing may vary with the type of driver you use; just keep the speaker frame edges as close as possible.)

The 6.5" drivers required a 5 3/4" hole, and the tweeter required a 3 3/8" hole. Again, check the dimensions of



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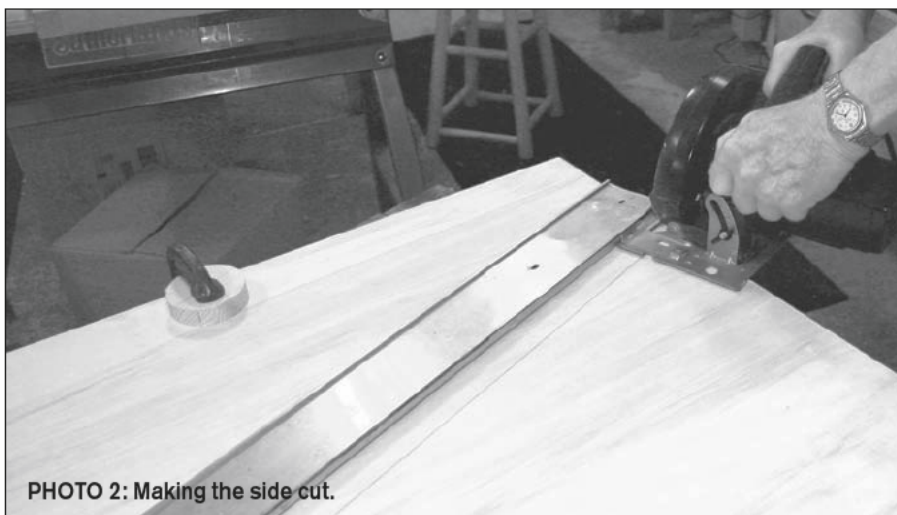


PHOTO 2: Making the side cut.



PHOTO 3: Wedge sides.

the drivers you use for correct hole size. The rear sloping panel is cut to 50.5". The internal tops are 6.25" × 7.25" and the bottoms are 7.25" × 22.75". I cut these from a 1" × 8" pine board. The enclosure partition is 14.75" long. Note the 70° bevel cuts to match the sloping back.

I made separate top caps and bottom foot pads from pine. The top cap is 9.25" × 9.5". The bottom foot plate is 11" × 25" and was cut from a 1" × 12" × 8' pine board. I routed the edges of these pieces with a Roman Ogee bit and attached them after assembling the main bodies of the enclosures. It is best to measure all the pieces for fit against the assembly as you go, because differences in the dimensional lumber can vary. The enclosure dimensions are detailed in Fig. 1.

## ASSEMBLY

With the panels cut to the correct size, assembly was not difficult. I used butt joints with the board edges first covered in Elmer's Wood Glue and then clamped them together. I then countersunk 2" drywall screws and drove them into the panels. With one side assembled, insert the center partition and attach the front and sides with drywall screws. Turn the partially assembled side over and carefully locate the center of the partition, sinking two screws through the side to secure it. Drill a ¼" hole in the panel for the tweeter and midrange wire leads. Cut a hole on the rear sloping panel to mount the speaker terminal cup.

Before attaching the final side panel, run a bead of silicone caulk along all the inside joints of the top,

front, rear, bottom, and the partition (Photo 5). Instead of using wood glue to secure the side panel, run a bead of caulking along the mating edges and then set the side panel into position and screw it in place. It is difficult to caulk the panel through the speaker holes and I have found that the silicone caulk will provide both a good seal and a good bond for the panels. Be sure to wipe off any excess caulk with a damp cloth.

Attach the bottom foot plates with drywall screws. Glue the external tops cap with liquid nails or nail with brads as desired. I also added a piece of ½" round to the base, but you can trim the enclosure as you desire. Fill in the countersunk screw holes with a wood filler matched to the finish you will use and then set the enco-



PHOTO 4: Cutting woofer panel.



PHOTO 5: Enclosure assembly.



tures aside for 24 hours. I applied 3/4" milled screen door lath as trim pieces on the plywood seams on the front and back. It makes an attractive element to the cabinet and is easy to apply with 1/2" brads.

## FINISHING

Sand all the surfaces of the enclosure taking special care to smooth the wood filler areas. Once the sanding is complete, you can stain the wood to your choice of color. I used a dark mahogany stain, followed 24 hours later with a

coat of polyethylene varnish. I rubbed the enclosures with 000 steel wool followed by another coat of polyethylene.

I stuffed the midrange sections of the enclosures with 8 oz of polyester pillow stuffing and lined the lower woofer sections on all sides with 1" of polyester blanket padding.

## CROSSOVER

I designed the crossover for 500-4000Hz crossover points and fabricated it on a pine board mounted in the bottom of the woofer chamber (Photo 6).

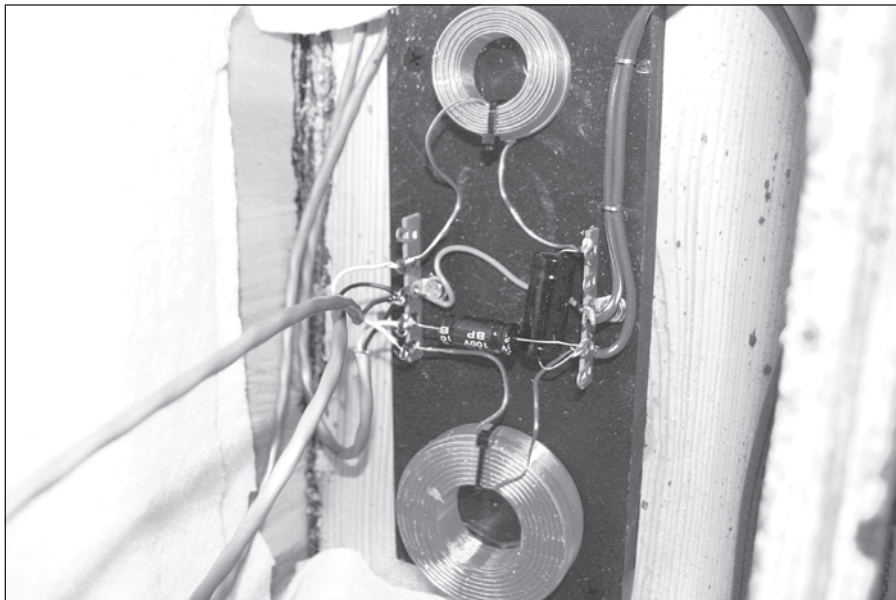


PHOTO 6: Crossover installed.



PHOTO 7: Wiring mids and tweeter.

## SCANSPEAK Discovery

Scan-speak is now offering a new line of products called Discovery. The Discovery line is designed to provide speaker builders with a Danish engineered and built product at economical pricing. Here are just a few of the items that encompass this expanding line of drivers. (8" in stock too, not shown)

The line will soon include more midrange choices and long stroke subwoofers with matching passive radiators.



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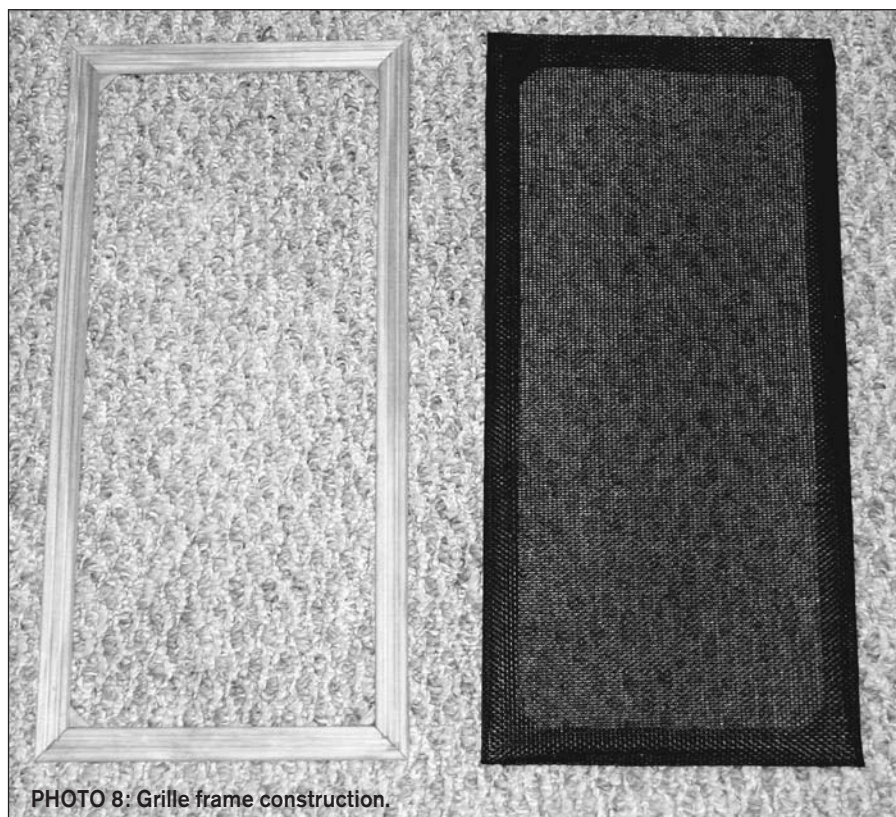


PHOTO 8: Grille frame construction.

It is wise to bench-test the speakers and crossovers before installation. This will reveal any defects in the drivers and careful listening will determine whether the crossovers are functioning properly.

Install the crossover in the bottom of the woofer section because this location is easy to access in the event of future repairs or changes. Run the wires to the upper M-T-M section through a 1/4" hole and then seal the hole with a bit of caulking. Wire the speakers' upper chamber first (Photo 7). The ones I used were each 12Ω impedance so I wired them in parallel. If you are using two 4Ω units, wire them in series.

I placed a bead of clear silicone bathroom caulk on the speaker mounting lip and used 1" drywall screws to fasten the speakers. The silicone ensures a good air seal and is easy to remove when changing drivers. Wire the woofer to the crossover and then the input to the speaker terminals. I installed a four-way cup-style speaker binding post on the lower rear of the enclosure and wired it to the crossover input.

## SPEAKER GRILLES

The grille frame for the front panel is

made of milled 3/4" screen door molding with 45° mitered joints (Photo 8). Each joint is reinforced by a corner gusset made from four-ply illustration board. This makes a lightweight frame that, when attached with Velcro, does not resonate with the audio.

First, glue all the frame joints together using picture frame clamps. Then glue and clamp the corner gussets. Paint the frame flat black, and stretch and staple the grille material over the grille frame. I used a reinforced screen door material that is designed to be pet-claw proof. This black plastic coated nylon is open to all audio frequencies, while being rugged enough to protect cone drivers from accidents. It also allows just a hint of the drivers to show through for those who want to see as well as hear their audio creations.

The frame is attached to the enclosure with matching Velcro "sticky back" strips stapled on the corners of the frame and mating enclosure surface. For the woofer, you can purchase ready-made round frames from Parts Express (www.parts-express.com, part number 260-428), or do as I did. Lay the woofer face down on the pet screen



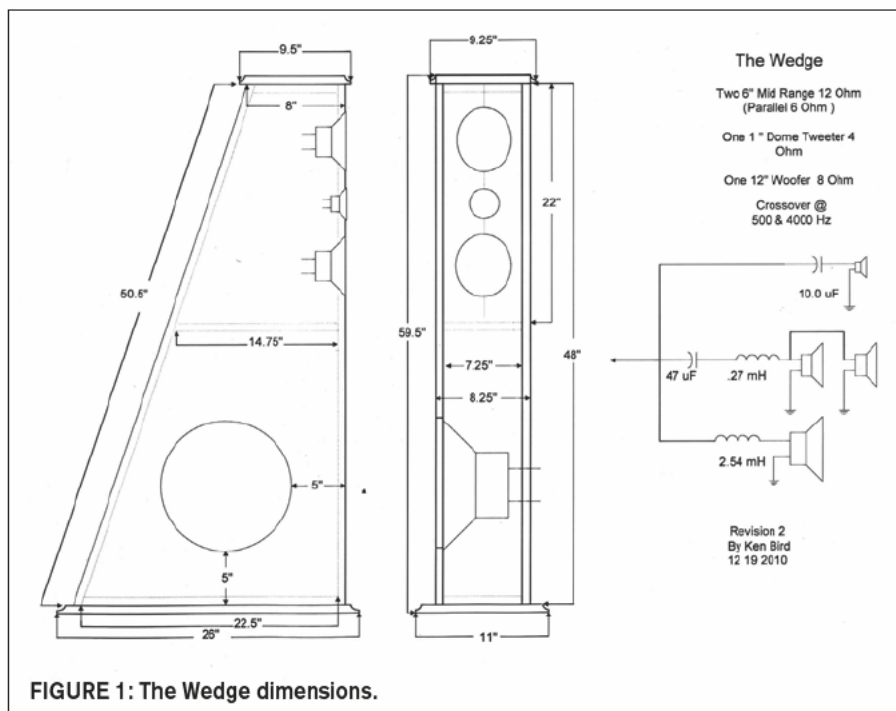
material and cut along the edge with a carton knife. Once you mount the woofer, just glue the grille to the speaker gasket using clear drying "tacky glue" as found in craft stores. There is plenty of clearance for the woofer's extreme excursions, and the woofer is protected by the screen material.

## SYSTEM TEST

One of my pastimes was astronomy and building telescopes. It was always a tentative moment when the first light was focused through the eyepiece and Jupiter would pop out clear and sharp with its multiple moons.

The same can be said of a home-built speaker system. Those first few moments with your favorite test record can be a real thrill or a "back-to-the-drawing-board" disappointment. This system did not disappoint.

The quality drivers performed to their specs, providing stunning imaging and solid bass without boom. The sloping back no doubt reduces internal reflections in the midrange, and the separate housing for the woofer prevents a muddy midrange response. The sloped back profile also gives the sense that the speakers are smaller than they really are. *ax*



# 7052PH

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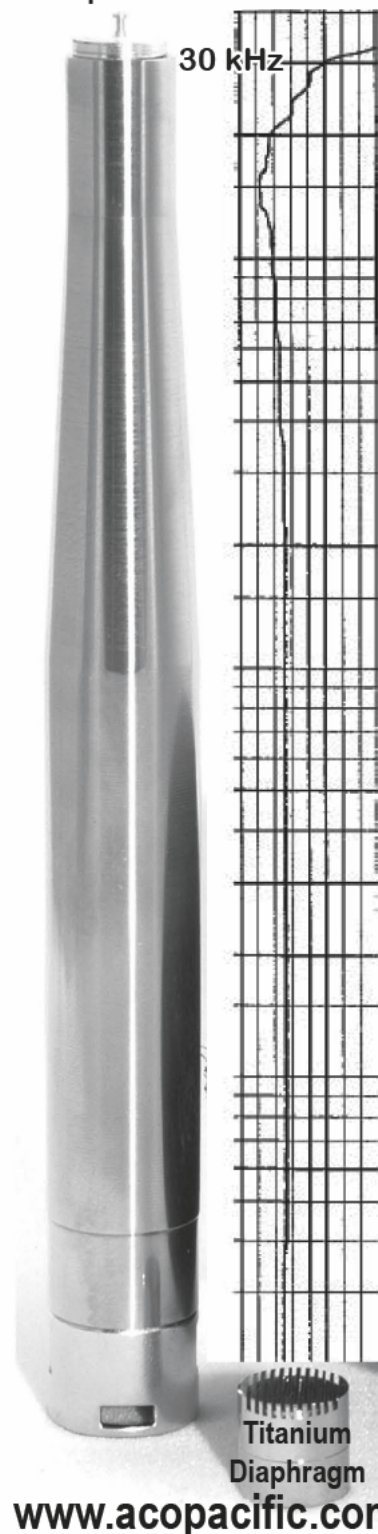
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- 1 each 1" x 12" x 8 (dimensional size) pine or hardwood for top caps and bottom feet
- 44' of 3/4" milled screen door lath for cabinet trim and speaker grilles
- 16 oz of "Polyfill" polyester stuffing
- 1 tube clear bathroom silicone caulk
- 1 tube white window caulking
- 48 1" drywall screws
- 100 1.5" drywall screws
- 1 pint Elmer's Carpenter's glue
- 1 square yard of fiberglass pet screen

### Speakers as below or equivalent to the specs listed

- 4 each 6.5" Parts Express 299-060 or equivalent 50W, 12Ω Response: 65-6000, Fs 68, SPL 86.5 Vas .26 Qts 0.72
- 2 each 1" dome tweeter Parts Express 299-938, 40W, 4Ω, Response: 1000-19000, Fs 1200, SPL 86.0
- 2 each 12" woofer, Parts Express 290-365, 140W, 8Ω, Response: 30-1500, Fs 33, SPL 87, Vas 3.42, Qts .89
- 2 each speaker terminals, Parts Express 260-311

### Crossover Components

Design goal: 500 and 4000Hz

- Two each C1 10μF Parts Express 027-428
- Two each L1 0.27μH Parts Express 266-810
- Two each C2 47μF Parts Express 027-448
- Two each L2 2.5μH Parts Express 027-836
- All wiring was with stranded 16 gauge wire.

# A New Tube Phono Amp

This high-quality vacuum tube unit features a "brick wall" that rf interference cannot penetrate.

**T**his phono amplifier has the most realistic sound reproduction of any of my phono amplifier designs. All my earlier phono amps avoided the use of negative feedback; this amplifier uses 6dB of feedback in the third stage. Applying feedback to this stage reduced the 1.5% distortion caused by the RIAA network to 0.66%. It simultaneously enhanced the realism of the sound. The improvement is outstanding!

This phono amplifier required as its main objective the ability to reject interference from cell phone, TV, FM, AM, police, fire transmitter station located within three-fourths of a mile, and five cell phone transmitters as close as 400". To achieve this degree of rejection, I modified the RIAA circuit architecture to successfully reject the

RF signals. This modification is applicable to all traditional RIAA circuits and is mentioned as a modification in my articles published in Oct. 2004, Apr. 2003, and Mar. 2001. The reason for this change is that I moved to the top 19th floor of a high rise and am surrounded by transmitting antennas at this location.

With a voltmeter hooked to the output of the phono amp, the pointer/signal was very active (not audible) and looked like a studio audio dB meter. With the modification of the RIAA circuit, the voltmeter pointer is stationary at zero. This feature is a real improvement for phono amplifiers and is very easy to implement. More details on this modification appear later in this article.

This phono amplifier offers "life like"

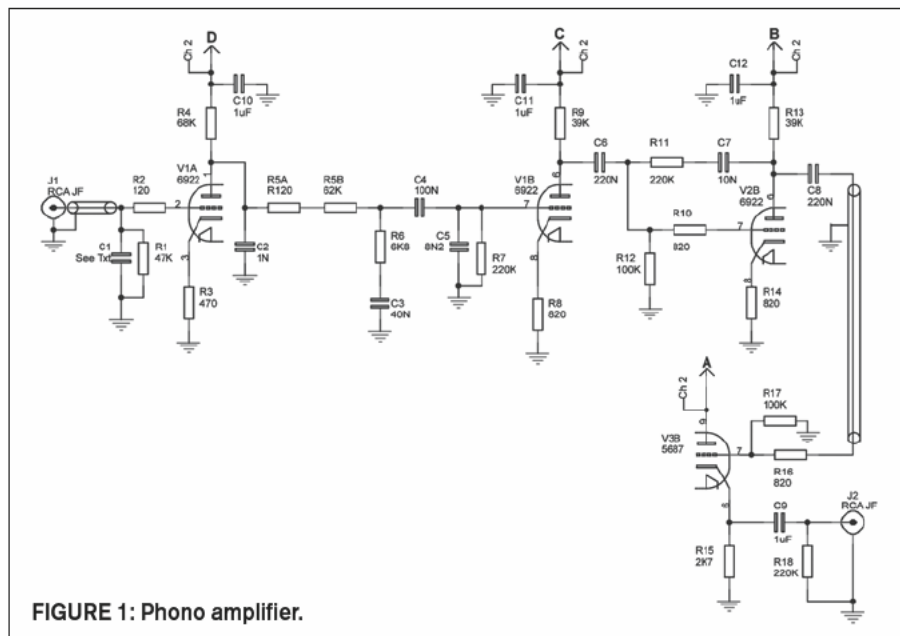
**TABLE 1 Distortion Measurements of Phono Amplifier at 1V Output**

|       |       |
|-------|-------|
| 20Hz  | 0.66% |
| 200Hz | 0.66% |
| 1kHz  | 0.65% |
| 5kHz  | 0.7%  |
| 10kHz | 0.7%  |
| 15kHz | 0.7%  |

Noise and Gain Measurement: The noise of the phono amplifier is circa 3.5mV at the output of the cathode follower stage with the input of the first stage 6922 open.

concert hall performance for a modest cost of \$130. It is a design featuring high-perveance tubes in the phono and line amplifier. The phono amplifier features three 6922s. It has an output voltage of 0.46V RMS (line amp) or 0.4V RMS (cathode follower) with an input voltage of 3mV/1kHz. Note that the phono amplifier in this article does not use high-mu 12AX7s, but instead uses linear, low noise, high-perveance 6922s with a mu of 33.

During the early days of the introduction of magnetic phono cartridges in commercial radios, the 12AX7s were developed to eliminate a medium mu tube and thus save the radio manufacturers money. The implementation of the 12AX7 was based on cost constraints and not technical excellence. Many manufacturers who employ the 12AX7 for costly hi-fi use do not seem to be aware that they adapted a low-cost concept in their product. To provide technical excellence in phono amplifier design and ensure a superior sound, vacuum tube phono amplifiers should be limited to tubes with a mu of circa 40. All low noise preamplifier tubes are designed with heaters having low heater current.



**FIGURE 1: Phono amplifier.**



**TABLE 2 RIAA Frequency Response of Phono Amplifier**

| RIAA Actual V RMS       | 9.2  | 7.1  | 4.5   | 2.6   | 1.4   | 1.0  | 0.74 | 0.39 | 0.2   | 0.14  |
|-------------------------|------|------|-------|-------|-------|------|------|------|-------|-------|
| Phono-Line Output V RMS | 9.8  | 7.8  | 4.8   | 2.6   | 1.3   | 1.0  | 0.78 | 0.48 | 0.24  | 0.14  |
| Frequency               | 20Hz | 50Hz | 100Hz | 200Hz | 500Hz | 1kHz | 2kHz | 5kHz | 10kHz | 15kHz |

**Note:** The output voltage of the audio signal generator is set for circa 75mV RMS with a 1kHz signal to obtain an output from the cathode follower of 1.0V RMS at 1kHz. The output signal level of the audio signal generator is not changed for the remainder of the frequency measurements.

## LINE AMPLIFIER AND CATHODE FOLLOWER

The line amplifier (Fig. 1) uses one 6922 with a  $\mu$  of 33. The 6922 has a flat frequency response of 20Hz to 20kHz and a distortion of less than 0.1% from 20Hz to 20kHz at 1.0V RMS. The 6922 voltage gain is 15 and output impedance is 5500 $\Omega$ . The plate voltage via 39k $\Omega$  resistor is 95V DC and the cathode voltage via 820 $\Omega$  resistor is 2.7V DC. The line amplifier stage employs 6dB of negative feedback via 220k $\Omega$  resistor and 10nF capacitor. I chose the value of the capacitor so it does not affect the low-frequency rolloff of the RIAA network. See **Table 1** for distortion measurements.

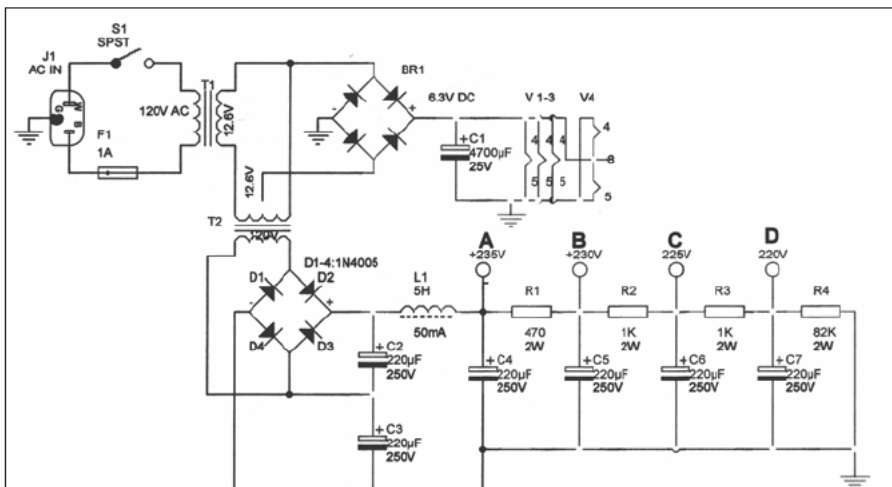
The cathode follower is a 5687 with a  $\mu$  of 16 but has no gain in a cathode follower configuration. The frequency response is flat from 10Hz to 100kHz and distortion is 0.2% at 1V RMS. The plate voltage is 235V DC and the cathode voltage is 12V DC via a 2700 $\Omega$  resistor. The low output impedance of the

cathode follower permits you to run a long audio cable to the power amplifier. The cathode follower is only required when you use a long run audio cable.

## VACUUM TUBE PHONO AMPLIFIER

The phono amplifier uses two 6922s in a non-feedback design with a passive RIAA network. The noise is circa 3.5mV and sensitivity is 3mV for 0.46V RMS (line amp) or 0.4V RMS (cathode follower) at 1kHz. The distortion of the first, second, and third stage is less than 0.15% from 20Hz to 20kHz with a 0.5V RMS output in each of these stages. These distortion measurements are made bypassing the RIAA circuit which produces 1.5% distortion. The 6922 third stage employs 6dB of feedback and reduces the distortion to 0.66%. I chose the 6922s for their low noise capability.

The first stage 6922 has Ebb of 220V DC, Eb of 55V DC, and Ek of 1.2V DC. The second stage 6922 has Ebb of 224V DC, Eb of 100V DC, and Ek of 2.6V DC.



**FIGURE 2: Power supply schematic.**

With 3mV, 1kHz fed to the input of V1, the output at V1 should indicate 55mV. The loss through the RIAA network drops the 55mV to 4.8mV at the input of V2. The voltage at the output of V2 is 38mV. The 38mV output of V2 feeds the third stage V3 and the output of V3 is 0.46V RMS. The output of the cathode follower is 0.4V RMS.

Capacitor C1 tailors the high frequency output of the magnetic pickup, and R1 provides the proper load to the cartridge.

The value of capacitor C1 is determined from cartridge manufacturers' minimum recommended capacitance value and by subtracting 100pF from that value. The 100pF value represents the 3" cable capacitance of the record player and input capacitance of 6922.

Note 1: I recommend you have an audio oscillator (optional) and vacuum tube voltmeter or DMM. This equipment is essential to ensure proper balance and gain is obtained throughout the phono amplifier circuit. The 6922s provide a very broad frequency response with a detailed and clean output.

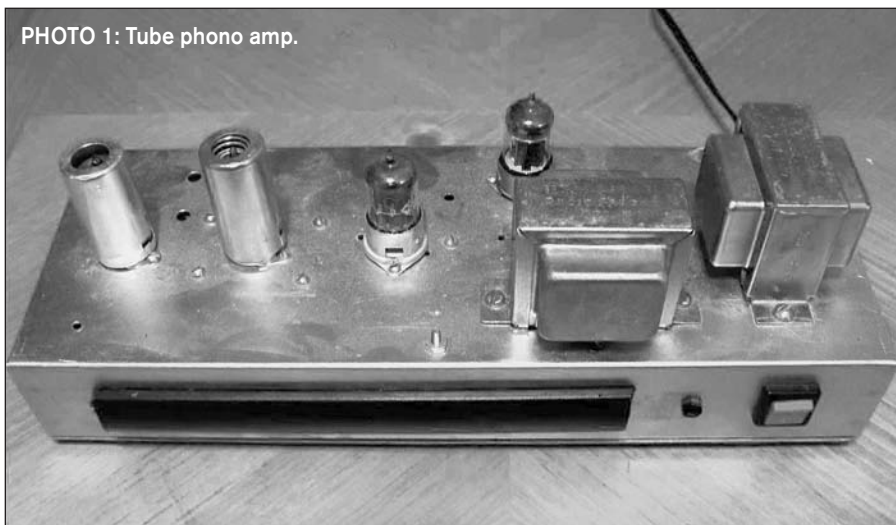
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PHOTO 1: Tube phono amp.



Note 2: Accuracy of the RIAA network is checked against the simulated output of the actual RIAA curve as shown in **Table 2**.

## RIAA MODIFICATION

The traditional RIAA circuit is modified to reject unwanted RF signals. This modification establishes a “brick-wall” that RF signals cannot penetrate. The modification is as follows: The 100nF coupling capacitor is moved from the plate of the first stage 6922 to the grid side of the second stage/RIAA network. This produces a carbon block between the plate and the R/C network. The frequency forming capacitor is split into two segments with a 1nF (0.001 $\mu$ F) capacitor connected to the plate of the first stage 6922 and a 10nF (0.01 $\mu$ F) capacitor connected directly to the grid of the second 6922. Make sure the 1nF and 10nF capacitor leads are short by using solder terminals.

The typical second stage grid resistor of 470K or 1M $\Omega$  is reduced to 220k $\Omega$ . This is the extent of the modification and the purpose of the change is self-evident. The output of the first stage and the input of the second stage are heavily bypassed for RF signals. Note that the same RIAA frequency curve is now maintained using a 1nF and 10nF capacitor, as opposed to the original design that used a single 12nF capacitor.

## POWER SUPPLY

The power supply (**Fig. 2**) consists of two 120V AC to 12V AC, 3A transformers, which are connected back-to-

back. The first transformer has one-half of the secondary winding connected to a full-wave, bridge-rectifier.

The output of the rectifier is 6.3V DC and is filtered by a 4,700 $\mu$ F capacitor. The output of the heater supply provides 6.3V DC to the three 6922s and one 6587 with a ripple of 0.32V RMS. The 12.6V AC output of transformer T1 is also connected to the 12.6V AC winding of transformer T2. The 110V AC secondary is connected to a full-wave bridge (D1 through D4) rectifier. Voltage doubling is accomplished by capacitors C2 and C3.

The choke L1 diminishes the switching transients that occur during the rectification process. Capacitors C4 through C8 are additional filtering devices and provide a low noise ripple of 0.6V RMS at the 235V DC output (the noise measurement was made with a DMM using shielded test leads). Resistors R1 through R3 provide de-coupling. The resistor R4 is the bleeder for the 235V DC output. Transformers T1 and T2 are only slightly warm after hours of operating time and cost only \$10 apiece.

## CONSTRUCTION

It is imperative that the phono amplifier (**Photo 1**) is an unpainted aluminum chassis to ensure that all components are properly grounded. Do not use a painted steel chassis!

The electrolytic capacitors are grounded at the negative band, except those connected in series. All other

components are connected to tube sockets. In the phono amplifier section, seven 5-pin terminal strips are required.

It is very important that the assembly be tailored around the implementation of the seven 5-pin and three 2-pin terminal strips (to ensure a good ground, interconnect the three 2-pin terminal strips ground lug with a separate ground wire). Also required are eight spade lugs. Using the terminal strips provides a very simple step-by-step construction process. It also permits easy changeability of parts. Building the control unit is a fun, easy project using the 5-pin terminal strips and a nightmare using a PC board. The RIAA network components are mounted on a single five-pin terminal strip.

It is important all audio runs approximately 3” or more use shielded cable and that the shield is grounded at the input end. The shielded cables at the input end should be grounded at the phono input jack. It will only be necessary to ground the shielded leads of the phono amplifier at the end nearest the input of the amplifier. The 5-conductor shielded cable that connects to the 6-position rotary switch must be grounded at the input end. Most grounds are made to the center post, 5-pin terminal strip. The use of chassis-mounted solder ground lugs for the cathode and grid resistors and heater grounds is also recommended.

The minimum test equipment required for building the phono amplifier and the control unit is a vacuum tube voltmeter or DMM, and an audio signal generator is optional.

## USEFUL HINTS

Building a preamp circuit can be very challenging, especially for first-time builders and 82-year-olds like myself. I provide the following useful tips, but I’m sure I missed many.

1. Have a well-lighted area and clean your eyeglasses and keep a magnifying glass nearby.

2. Use 40W pencil iron and 60-40 rosin core solder.

3. Use spade lugs generously when a component or heater ground is needed. Use 5-pin terminal strips for mounting components.



4. Make sure you don't have any cold solder joints and insert components through holes of tube socket pins.

5. Secure fasteners for lugs and terminal strips with a tight fit to ensure a good ground. Do not secure until all components are soldered to avoid a cold solder joint.

6. Tubes V4 and V5 use 2-pin terminal strips for ground and B+ connections. Capacitor C8 of power supply is located at V4, 2-pin terminal strip.

7. Capacitors C4 through C7 and R1 through R3 are secured to a 5-pin terminal strip. The 6.3V DC tube heaters wires and capacitor C1 are connected to a 5-pin terminal strip.

8. At the output of V6 a 5-pin terminal strip is used to secure capacitor C9, B+ and plate resistors and grounds of shielded cable.

9. When construction is finished, make sure all screws and nuts are tightened to ensure good grounding.

An improper solder joint or a defective tube socket with loose sleeves can cause a hum, howl, squeal, or microphonics. If this occurs, check all fasteners for snugness and discard screws that have stripped threads. Examine all solder joints under a bright light and using a pair of needle nose pliers, tug on all wires and components to ensure the solder joints are secure. This is the area that causes the most trouble when building preamps. Also, ensure components have space between each other and try to keep leads short.

When ordering tubes, you should purchase one extra 6922 and 5687. If an ordered tube is defective, this will simplify troubleshooting.

In conclusion, this phono amplifier has exceptional concert hall life-like sound and has excellent RF rejection capabilities. The line amplifier stage with 6dB of feedback provides realism to the music. This stage brings the music to life. The line amplifier stage is suitable for use with short audio cable runs. For long audio cable runs, be sure to add the cathode follower stage. For \$130 and some spare time, this is a project that can be very rewarding. The phono amplifier has been operating for five years with no problems. *ax*

**Table 3: Phono amplifier parts list** (double all parts, except C9-C11, J1-J4, V1-V4, X1-X4, W1)

These 7/8" diameter high-quality ceramic tube sockets provide sleeves that grasp the tube fins firmly. They are recommended for use with high gain preamplifier circuits. The cost with shield is \$2.25.

| Symbol            | Description                                                        | Vendor | Part no.    |
|-------------------|--------------------------------------------------------------------|--------|-------------|
| C1                | See text                                                           |        |             |
| C2                | 0.001μF, 5%, 600V, poly.                                           | AE     | C-PD001-600 |
| C3                | .04μF, 600V, poly.                                                 | AE     | C-SD04-600  |
| C4                | 0.1μF, 5%, 400V, 5%, poly                                          | AE     | C-PD1-400   |
| C5                | 8N2, 600V, poly                                                    | AE     | C-PD01-400  |
| C6, C8            | 0.22μF, 400V, poly.                                                | AE     | C-PD22-400  |
| C7                | 10N, 250V, Solon, Fast                                             | AE     | CFT1-250    |
| C9-C12            | 1μF, 250V, metal film                                              | RS     | 272-1055    |
| R1                | 47K, ½W, carbon                                                    | AE     | RA-47 K     |
| R2                | 120, ½W, carbon                                                    | AE     | RA-120      |
| R3                | 470, 1W, carbon                                                    | AE     | RB-470      |
| R4                | 68K, 1W, metal film                                                | AE     | RE-68 K     |
| R5A               | R120                                                               | AE     |             |
| R5B               | 62K                                                                | AE     |             |
| R6                | 6.8K, ½W, carbon                                                   | AE     | RA-6.8 K    |
| R7, R11, R18      | 220K, 1W, carbon                                                   | AE     | RB-220 K    |
| R8, R10, R14, R16 | 820, 1W, carbon                                                    | AE     | RB-820      |
| R9, R13           | 39K, 1W, metal film                                                | AE     | RE-39 K     |
| R12, R17          | 100K, ½W, carbon                                                   | AE     | RA-100 K    |
| R15               | 2.7K, 1W, metal film                                               | AE     | RB-2.7 K    |
| V1-V2             | 6922 (Sovtek)                                                      | AE     | 6922        |
| V3                | 5687 (NOS)                                                         | AE     | 5687        |
| X1- X4 T-6        | ½ ceramic tube socket,<br>9-pins with shield                       | AE     | P-ST9-162   |
| W1                | Single conductor, shielded                                         | RS     | 42-2371     |
| J1-J2             | RCA phono jack, female, gold,<br>large diameter housing, low noise | BE     | MTG12B      |

Note: These are the best jacks for low noise applications I could find.

#### PARTS SUPPLIER

RS = Radio Shack, [www.RadioShack.com](http://www.RadioShack.com)

AE = Antique Electronic Supply, [www.tubesandmore.com](http://www.tubesandmore.com)

ALEL = All Electronics, [www.allelectronics.com](http://www.allelectronics.com)

BE = Baynesville Electronics, [www.baynesvilleelectronics.com](http://www.baynesvilleelectronics.com)

Mouser, [www.mouser.com](http://www.mouser.com)

**Table 4: Power Supply Parts List**

| Symbol              | Description                          | Vendor | Part no.   |
|---------------------|--------------------------------------|--------|------------|
| T1, T2              | 120V AC to 12V, 3A                   | RS     | 373-1511   |
| L1                  | choke, 5H, 50mA                      | AE     | P-T155H    |
| BR-1                | 4A, 50V AC, PIV                      | RS     | 276-1173   |
| D1-D4               | 1A, 600V AC, PIV,<br>1N4005, 2 packs | RS     | 276-1104   |
| R1                  | 470, 2W                              | AE     | RF-470     |
| R2, R3              | 1K, 2W                               | AE     | RF-1K      |
| R4                  | 82K, 2W                              | AE     | RF-82 K    |
| C1                  | 4700μF, 25V, electrolytic, radial    | ALEL   |            |
| C2-C7               | 220μF, 250V, radial, electrolytic    | ALEL   | CR-184     |
| Fuse holder, 3AG    | in-line type                         | RS     | 270-739    |
| F1                  | 3AG, 1A, slow-blow                   | RS     | 270-1021   |
| TS                  | terminal strips (3-packs)            | RS     | 274-688    |
| S1                  | toggle switch, SPST                  | RS     | 275-694    |
| Chassis box         | aluminum, 13.5 by 5 by 2             | AE     | P-H1444-18 |
| Chassis cover plate | aluminum, 13.5 by 5 by 2             | AE     | P-H1434-18 |
| Rubber feet         | self stick, rubber                   | RS     | 64-2342    |
| J1                  | AC IN                                |        |            |

#### TEST EQUIPMENT USED

H-P distortion analyzer, 331A  
Heathkit sine-wave, square-wave,  
audio generator, IG-5218  
DMM, Radio Shack 22-168A  
Oscilloscope, Proteck, Model 6502

**Warning:** Dangerous voltages are present, exercise extreme caution when working on the control unit and never leave the control unit upside down when children are present.

# The Return of the Disappearing Loudspeakers

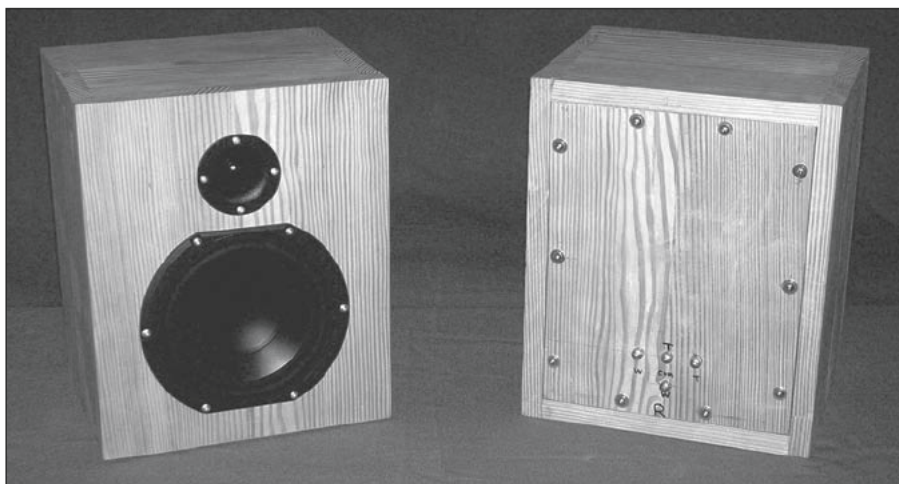


PHOTO 1: Cabinet front and rear view.

Having refined my stereo system to a pleasing level of fidelity, and with a Blu-ray player that sends out 7.1 channels of analog sound, I decided it was time for some surround speakers. Learning of this, my brother [Victor] brought his mini-monitor Disappearing Loudspeakers<sup>1</sup> over one weekend and gave me a convincing demonstration using Blu-ray movies. The best experience was the aural sensation of James Bond racing an Aston Martin DBS, accompanied by a snarling engine, screeching tires, and gunfire. That decided it—I would build a pair of THX surround speakers (Photo 1).

My design approach for the new surrounds was to make them a modernized version of the Disappearing Loudspeakers, giving structural issues added attention, and, of course, meeting the THX standard for surrounds. I would also build stands that were not adaptations, and impose my own choices on their features.

I would rely on Vic's collection of measurement, modeling, and crossover design software for the crossover part of the project. The cabinet design, driver impedance measurement, modeling, and all construction fell to me.

## THE DRIVERS

Vic researched the descendants of his original drivers and chose the Peerless 830874 for the woofer and Peerless BC-25SC06-04 for the tweeter. The woofer is a 6.5" polypropylene cone driver, while the tweeter has a 1" fabric dome with ferrofluid damping. Detailed specifications are available on the web ([www.tymphany.com](http://www.tymphany.com)). I ordered three woofers and five tweeters in case anything should go wrong; however the spares are still unused at the time of this writing.

## CABINET VOLUME

The THX specification for surround speaker frequency response is 80Hz and up, making sealed-box volume computation straightforward<sup>2</sup>. I increased the theoretical volume of 548 in<sup>3</sup> by 20% to allow for driver components and internal bracing, but no crossovers. Applying the "ideal" proportions of 0.8 × 1.0 × 1.25, I rounded up the box internal dimensions slightly to 8.75W × 10.875H × 7.0"D.

## CABINET MATERIAL

Aiming to improve upon past boxes made from MDF particleboard, I studied wood species in Table 4-3b of Reference 3. I rated potential candidates

based on the ratio of Modulus of Elasticity (the higher, the stiffer) divided by Specific Gravity (the lower, the lighter).

Next, I visited the best lumberyard in the area to size up the winning Coastal Douglas Fir, which they had in abundance. I purchased two pieces of 1 × 6 × 14 S4S, which measures ¾" × 5½" × 14", surfaced four sides. That gave me stock for two cabinets, and I intended to edge-splice boards for all surfaces (this improves appearance and reduces any tendency to warp).

I exploited the relative stiffness of this wood to resist deflections, and to have the lower mass store less vibrational energy and be damped quicker by the final choice of deadener.

A practical word of advice: This species of wood is very prone to splitting due to its pronounced grain, and makes painful splinters. All saws, drills, router bits, and so on must be as sharp as possible for clean cuts. More exotic woods in Ref. 3 are candidates, so you can make other choices.

## CABINET STRUCTURE

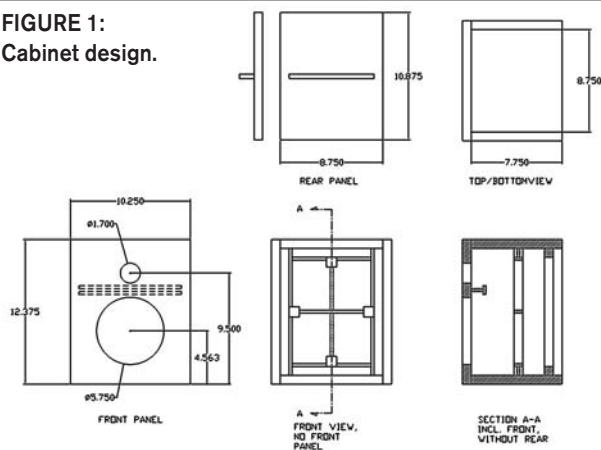
Small home-built speaker cabinets are great subjects for sophisticated construction, because very little more time and materials are needed to achieve truly high-end results.

The tweeter is mounted just above the woofer, but with enough front panel wood left between their cutouts to attach the sturdy cross-brace glued to the backside of the front panel. That cross-brace is made of a T-section with holes cut out of the web to minimize reflected sound bouncing back out through the woofer. The rear panel is removable (Photo 2).

Internal bracing is applied to all four front-to-back panels to improve their rigidity. This method, which has appeared in several articles in *aX* and *Speaker Builder*, works wonders.



**FIGURE 1:**  
Cabinet design.



**PHOTO 2:** Cabinet components before assembly.

## ASSEMBLY

My choice of glue in building the cabinets was yellow aliphatic carpenter's glue, clamped or weighted during assembly, using no fasteners.

**Figure 1** shows the cabinet design, while **Photo 2** gives a better idea where the bracing is applied to stiffen the panels.

The internal T-section bracing, made from strips ripped from the edges of leftover  $\frac{3}{4}$ " fir, is located about one-third of the way into the box from the rear. The cross-braces are attached with small shear plates visible in **Photo 2**.

After assembling all the components, I used a belt sander to even up the external joints and develop a uniformly smooth surface. This wood is a pleasure to work with and is pleasantly aromatic, which convinced me to leave it unfinished.

I lined the box forward of the bracing with lead sheet to damp vibration. I then lined that same area, plus the back, with 1" gray polyurethane foam bonded with spray adhesive and stuffed the interior with polyester batting to eliminate the frequently occurring boxy "cupped hands" sound. Other damping and stuffing materials are equally valid, so feel free to apply your own choices.

Each rear panel is attached with ten Phillips 2" long round-head wood screws with washers, because flat-head wood screws will split this type of Douglas Fir. The screw frame in the rear of each box is mounted with the grain aimed crossways to prevent the screws from splitting the pieces. Foam tape seals the frame to the back, and is squeezed very tightly by the ten screws.

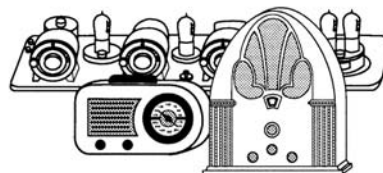
Mounting drivers in a box makes me nervous, because there is always a chance of slipping and punching a hole in a cone. In this project the drivers are mounted with square-drive wood screws ([www.mcfelys.com](http://www.mcfelys.com)), a welcome improvement over other types. I made a square driver out of a worn-out Phillips screwdriver using a bench grinder, then cleaned it up with a file. You can harden the tip by heating it bright-orange hot with a torch and quenching it in a tin can of oil.

## ELECTRICAL CONNECTIONS

Internal wiring is Teflon-insulated silver plated, with fast-on type fittings at the speaker ends and spade lugs at the interior cabinet feed-through screws. Those four rear-panel feed-throughs are pieces of  $\frac{3}{8}$ " OD brass bar, drilled and tapped for 10-32  $\times$   $\frac{3}{8}$ " round-head Phillips PEM screws (with integral lockwashers). I used a bench vise to press the chamfered feed-throughs into snug holes in the rear panels.

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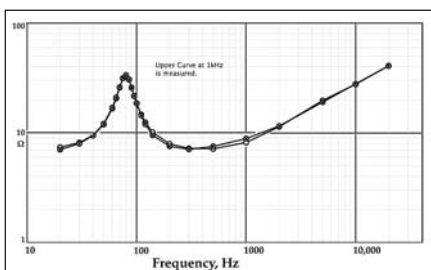


FIGURE 2: Woofer impedance magnitude.

## PRELIMINARY TESTING

The woofers were broken in with low-frequency sine waves while hanging in free air. I then measured driver impedances in-box using a B&K Precision function generator ([www.bkprecision.com](http://www.bkprecision.com)), digital VOM, and calibrated series resistor, plus calculator. Using Vic's software, I extracted the seven circuit element components (3 R, 3 L, and 1 C) which model the drivers, and e-mailed him the data.

Figure 2 shows the measured woofer impedance magnitude and the seven-element simulation impedance magnitude, and they match accurately enough for me! It also appears that I got lucky and hit the 80Hz box tuning. The tweeter impedance measurements (not shown) behaved similarly to those for the woofer.

## CROSSOVER DESIGN

Vic covers the crossover design process in the accompanying sidebar.

The woofer inductors aren't standard values. These were made by either removing or adding turns to inductors from Madisound ([www.madisound.com](http://www.madisound.com)), measuring their inductance periodically until target values were met within one turn.

Large capacitance values consist of combinations of non-polar electrolytics adjusted with film capacitors. Not ideal, but it seems to be working.

Vic's computer-optimized crossover

values always result in very precise performance. Because we have no way of measuring completed system frequency response any more accurately than the first measurements, we just listen to the surrounds both as main channels and as side-channel surrounds for cinema sound, and evaluate what we hear. So far, we found no reason to alter the crossover component values, shown in Fig. 3.

The crossover region is seamless, and the radiation pattern is very wide anywhere in front of the speakers. This makes aiming them as surrounds a very non-critical task.

## EXAMPLE STANDS

You can see "provisional" (meaning ugly) stands made to support the surrounds in Photos 3 and 4. They are composed of a large wooden disk-shaped foot, 5' of steel electrical conduit, an upper hollow platform/box containing the crossovers, and thick felt padding on top of the platform.

The upper and lower halves of the box are joined with piano hinges on each long dimension, with loose pins. By removing a hinge pin on either side, you can open the boxes. Sure beats unscrewing speaker panels to access the crossovers. Where the boxes close, I applied strips of foam tape to pre-load the hinges and eliminate buzzing.

Speaker wiring snakes down through the conduit and exits just above the foot through a grommated hole, while screws in tapped aluminum bars embedded in the feet provide terminals for wiring to the amplifier. The three major components slide together and apart—I need only open the boxes and disconnect spade lugs and everything is quickly reduced to three easily stored pieces.

I must confess I clumsily banged my head against the mounted surround speak-

ers (a speaker weighs 15.1 lbs) during initial setup and testing. This spawned the decision to always remove the speakers from the stands and put them on the floor when not in play—here in California I would eventually be very sorry if I didn't.

## TWEAKS

Once the speakers were playing on their stands, I made some tweaks to improve the sound. This consisted of replacing C1 for the tweeter and C1 for the woofer with better-quality non-polarized electrolytic capacitors from the Xicon BPH series (Mouser, [www.mouser.com](http://www.mouser.com)). The result is improved clarity and naturalness.

Builders are encouraged to experiment with their favorite components anywhere in the crossovers. Our findings, for example, are that inductors sound best having the lowest DC resistance, thus the iron cores. These surrounds don't play at volumes high enough to risk saturating the core.

## SONIC IMPRESSIONS

Test tones (from the Blu-ray player) were used to adjust the positioned surrounds in concert with the main speakers. A menu from the player allows each speaker to be attenuated in half-dB steps. When the system was finally checked out, some "challenging" movies were played, beginning with Digital Video Essentials.

The sonic personality of these surrounds ended up being essentially identical to my existing front channels, equipped with Focal midrange and tweeter drivers. This match makes the surround channel information merge seamlessly with the aural field from the main channels, and envelops the listener in a single continuum of sound. Echoes, decay, and other large-scale sound field effects

FIGURE 3: Crossovers.

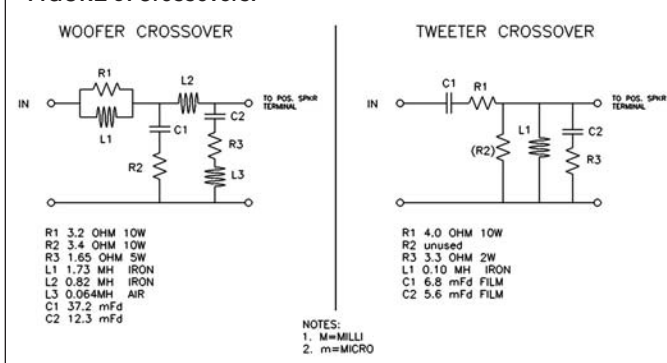
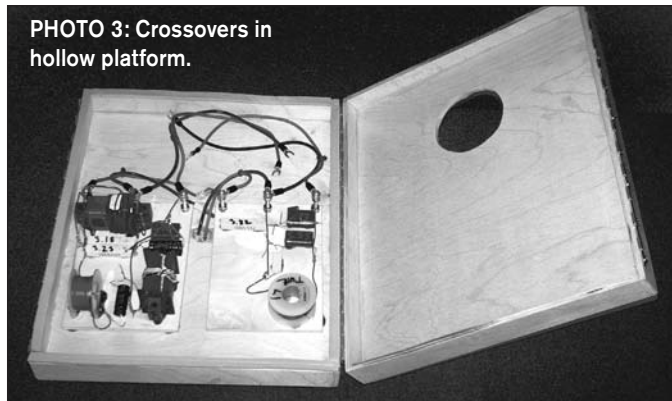


PHOTO 3: Crossovers in hollow platform.





are convincingly reproduced.

The movie background music sent to the surrounds is a continuous extension of the front-channel music. This may or may not be “real,” but it is very appealing. And yes, the Aston Martin driving experience is back with a vengeance, better than ever.

## CONCLUSION

Coastal Douglas Fir is performing admirably. At performance listening levels the surrounds tend to disappear into the sound field. That is the target use of these speakers, so my objective has been met. Thanks to the high quality of the drivers and the optimized crossovers feeding them, they sound very good even used as freestanding mini-monitors, where they image and detail beautifully. There they rightfully deserve a subwoofer, which is exactly how Vic uses his Disappearing Loudspeakers. *aX*



PHOTO 4:  
Surround on  
stand.

## CROSSOVER CIRCUIT DESIGN

By Victor E. Staggs

I measured the assembled surround speakers outdoors atop a ladder on Darcy's front lawn with the microphone at 1m, aimed midway between the two drivers. The test signal was a 10-second sine wave swept between 20Hz and 20kHz. **Figure A** shows the measurements, with the levels normalized to the woofer's low-frequency efficiency. Frequency resolution was limited to 1/12 octave.

Reflections from the surroundings cause ripples. The woofer's peak at 4kHz and ripples above there are the real thing, and reproduce the manufacturer's spec sheet pretty well.

Below 300Hz the tweeter measurement consists mostly of background noise, which is also true for the woofer below about 80Hz.

## FITTING THE MEASUREMENTS

**Figure B** shows the best fit to the woofer frequency response. At low frequencies the Thiele/Small model is used, and a collection of standard functions fits the rest of the measurement. The box diffraction signature consisting of a +6dB rise in amplitude above 200Hz is evident.

The target function for the crossover was an acoustic Linkwitz-Riley response centered on 2250Hz, the lowest practical crossover frequency for the tweeter. This yields good off-axis response from the woofer just below the crossover frequency.

The circuit topology chosen was that for the “Disappearing Loudspeaker”<sup>1</sup>. The tweeter network was simplified by omitting the resistor to ground in parallel with the inductor.

The woofer and tweeter input impedances were measured and fit with the Thiele/Small model as it appeared in Reference 4.

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## OPTIMIZING THE CROSSOVERS

The tweeter crossover was optimized by hand in software, using the impedance fit and the idealized Thiele/Small model of its frequency response, but having the measured acoustic efficiency. In this step I used the idealized Linkwitz-Riley low-pass response of the woofer. The resulting fit introduces a ripple of about +1dB in the summed response.

I optimized the woofer crossover using a nonlinear optimization program, feeding it the modeled woofer impedance curve and the analytical fit to its acoustic frequency response. The crossover was forced to produce a low-pass acoustic response at 2250Hz that yielded flat response when summed with the idealized Linkwitz-Riley high-pass response. In addition, I used a penalty function to keep the woofer input impedance, including the crossover network, from dropping below 7Ω.

The target function for these surround speakers leaves a 2dB rise due to the box diffraction signature. I tested this beforehand using the original Disappearing Loudspeakers on stands near the front wall, and listening to the effect of filters applied to music via software, and including a subwoofer. When the surround speakers are used near a wall, this lack of bass fills in. When used as main front left/right speakers, they will need to be closer to the front wall than most other speakers. This design choice was based on curves presented in Reference 5. When used as main front left/right speakers, they will need to be closer to the front wall than most other speakers.

Figure C shows the modeled woofer acoustic response

with the crossover in place, plus the target function (dashes). The tweeter response is the idealized Linkwitz-Riley fourth-order high-pass response.

Figure D shows the modeled woofer acoustic response as filtered by the crossover (solid line), summed with the idealized high-pass response of the tweeter, along with three parallel curves that are the target function and error limits of  $\pm 0.5$ dB. So far we do not have anechoic measuring facilities that have better amplitude resolution than this.

The input impedance of the completed speaker is 6.5-7Ω below the crossover frequency and 3.5Ω above the crossover frequency. There is little continuous energy above 2250Hz in most music and sound effects, so this should be an acceptable load for the amplifier.

The woofer's bass alignment matches the THX specification for bass management in A/V receivers and processors given in Reference 6. ■

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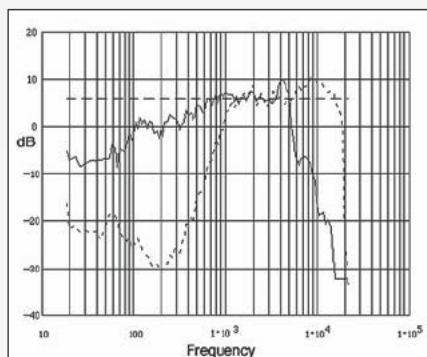


FIGURE A: Woofer, tweeter measured acoustic frequency response.

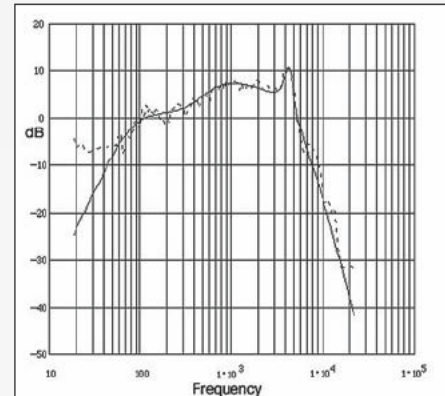


FIGURE B: Analytical fit of woofer acoustic response.

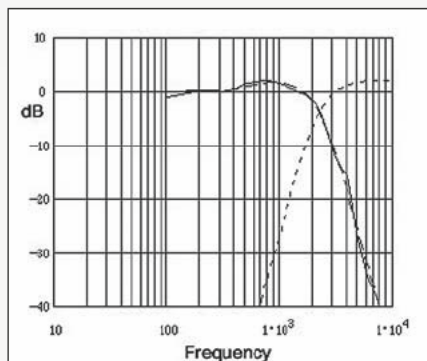


FIGURE C: Modeled woofer response, plus idealized tweeter, with crossover.

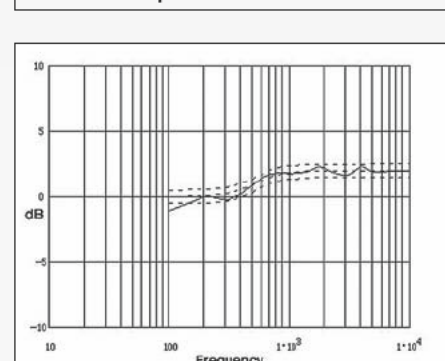


FIGURE D: Sum of modeled woofer, idealized tweeter with crossover.



# Infra Dig: Passive Preamps, Part 3

The author takes a look at phono preamp design, particularly high-pass filters to remove infrasonic signals.

In two previous *audioXpress* articles, I talked about passive EQ networks for phono preamps (RIAA and otherwise)<sup>1</sup> and discussed design constraints that arise in the classic two-amplifier design<sup>2</sup>. After a couple of years' detour into power supply designs, it's time to return to preamps.

When designing a phono preamp, it's important to remember that the amplification and equalization circuits aren't the end of the story. In the real world, it's also important to filter unwanted garbage from the signal.

In his ground-breaking JAES paper<sup>3</sup>, Tomlinson Holman demonstrated to my satisfaction the necessity for filtering infrasonic junk out of LP-derived audio. My experience has borne this out. Whenever I've switched infrasonic filtering in and out of a phono circuit, the music sounded cleaner with the filtering engaged.

Why should this be? Infrasonic crud can cause increased intermodulation distortion in power amplifiers—particularly tubed amplifiers—and sometimes in preamps, too. Large infrasonic signals, from warped records and cutting lathe rumble, can also drive woofers to great excursions, into regions where they are less linear. Filtering out these infrasonic signals lets both amp and speakers perform better.

It's also good to filter out infrasonics before recording the signal; these days, most of us who copy records are digitizing them, and digital recording systems pass infrasonic signals all too well.

## WARP FACTOR

Most arm/cartridge systems for LPs are (or should be) tuned for a fundamental resonance of about 10Hz. This allows

some mechanical filtering of some of the worst warp frequencies (clustering around 4Hz) without seriously affecting the lowest musical frequencies (42Hz on most classical and traditional pop recordings, lower on pipe organ, piano, and synth-heavy discs).

Holman found, however, that the poorly damped 12dB/octave high-pass filter formed by a cartridge and tonearm doesn't provide enough infrasonic filtering for optimum performance. Indeed, even such devices as the damping brush included with the popular Shure V-15V series of cartridges ([www.shure.com](http://www.shure.com)) allow a good deal of infrasonic material to pass through. In the real world, LPs still need a good electrical high-pass filter to sound their best.

That goes double for 78s, which can have some pretty bad warps, and taking a warped disc back to the store and exchanging it is not an option. In fact, 78s can sometimes demand considerably more filtering than LPs (which I'll talk about at the end of the article).

Holman recommended a third-order Butterworth filter (maximally-flat response) at 16Hz for best filtering without compromising the amplitude and phase response of musical signals. I prefer a more damped design; I like a Bessel filter, for optimum transient response.

## FILTER TIME

When designing filters, I do what almost everyone does—I open the pages of Don Lancaster's *Active Filter Cookbook*<sup>4</sup>. Lancaster lays out the basic theory and practice of simple filter design, then gives charts allowing you to pull one from column A, two from column B, and wind

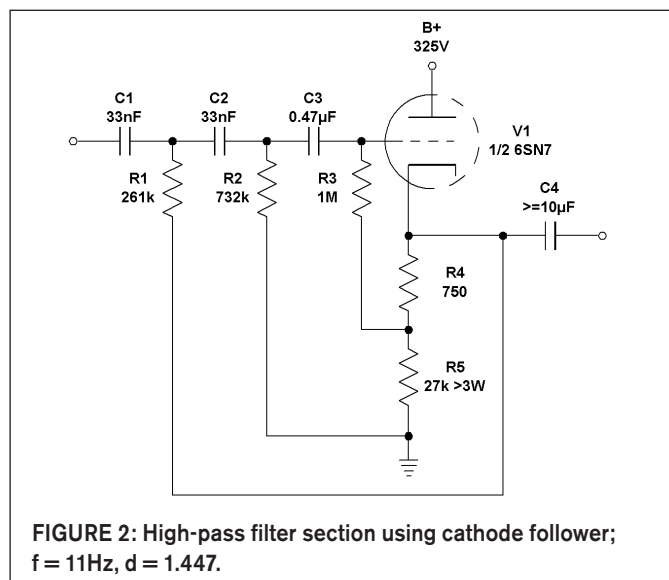
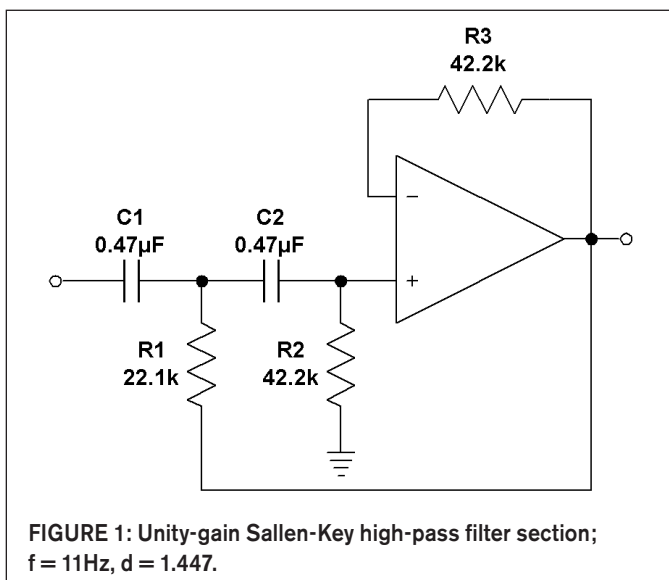
up with just about any filter you need.

One quirk of Lancaster's book is that he doesn't use the standard "Q" to describe the shape of a filter circuit. Instead, he uses the reciprocal,  $1/Q$ , which he calls "d" (for damping). I'll use his terminology; if you prefer to think in terms of Q, just push the "1/x" button in your mental calculator.

To construct a third-order (18dB/octave) high-pass filter, you need to cascade a first-order section and a second-order section. For any filter that's not a Butterworth type, the cutoff frequencies of the sections are different from the cutoff frequency of the composite system. If a third-order Bessel filter has a -3dB frequency equal to  $f$ , the first-order section should have a cutoff of  $f \times 0.753$ , while the second-order section should have a cutoff frequency of  $f \times 0.688$  and a damping factor  $d = 1.447$ . For a 16Hz filter these translate as almost exactly 12Hz and 11Hz, respectively.

The first-order section is simply a passive filter, implemented by an RC circuit. In a previous article, I suggested that in a passive preamp, the coupling capacitor between the first amplifier and the passive EQ circuit is not a good place to perform this filtering, as the EQ network's impedance will interact badly with the high-pass coupling cap and its associated resistor. Instead, I suggested making this coupling network as large as practical, or, when possible, DC-coupling the first amplifier (with or without a servo) and performing the first-order high-pass filtering elsewhere.

One place for that is at the input, where the cartridge meets the amplifier. A high-quality capacitor in this location



begins filtering out infrasonics at the earliest possible point, before they have a chance to generate nonlinearity in any active circuits. This follows the maxim, “The narrower you open the window, the less dirt flies in.” (For several reasons, this works better with FET-input and tubed circuits than it does with circuits built around bipolar transistors—a topic for another day.)

## SECOND ORDERS, PLEASE!

The rest of the infrasonic filtering is performed by a second-order filter section. Take a look at **Fig. 1**, which shows the most common type of active high-pass circuit found in audio equipment, the “Unity-Gain Sallen-Key filter.” Here it’s implemented using an op amp as the active engine. (It doesn’t need to be an op amp; an emitter- or source-follower, or even a vacuum tube cathode follower, can do the job. More about that later.) Most commonly, the two capacitors are set at equal value, with damping controlled by the values of the two resistors.

In this circuit, you first choose an “operating resistance”—an ideal resistance used in computing the cutoff frequency. If say, the two capacitors have a value of C and the desired cutoff frequency is f, then:

$$1 \quad R = 1/(2 \times \pi \times f \times C)$$

where R is in ohms, f in hertz and C in farads. If C = 0.47µF and f = 11Hz, then R = 30.8k.

R is not the value of the two resistors in the unity-gain Sallen-Key filter,

though. To calculate them, compute:

$$2 \quad R1 = R/g$$

and

$$3 \quad R2 = R \times g$$

What’s g (gamma)? In a unity-gain filter,

$$4 \quad g = 2/d$$

If d = 1.447, g = 1.382. In a filter that’s not unity-gain, g becomes a little more complicated, but I’ll get into that later, when I talk about tubed circuits.

For a filter section with f = 11Hz, d = 1.447, and an operating resistance R = 30.8k, you wind up with C1 = C2 = 0.47µF, g = 1.382, R1 = 22.3k, and R2 = 42.5k.

## BEARING THE LOAD

One question Lancaster’s book doesn’t cover is the load a Sallen-Key high-pass filter puts on the preceding stage. To find out, I set up a series of simulations in PC-ECAP; the results are shown in **Table 1**. It turns out the impedance is high at low frequencies, drops to a minimum just above the cutoff (–3dB) frequency, then rises to a plateau through the passband of the filter. In the worst case I tried, a 3rd-order Butterworth filter, the impedance minimum was approximately 0.89 × the operating resistance R, at a frequency of about 1.06 × the cutoff frequency. In the others, the impedance minimum was always greater than R.

This was something of a relief; I had worried that the impedance might dip to a level low enough to cause loading problems on a previous stage; with rea-

sonable care, this is easy to avoid.

I did these simulations using a typical op amp as a model, with very high open-loop gain (100dB) and low output impedance (1Ω), both reasonable assumptions at the low frequencies I’m working with. Just for grins, I reran the Butterworth simulation using an “op amp” with an open-loop gain of 10 and an open-loop output impedance of 5k—in other words, a bit like a 6SN7 tube. The amplitude response was no longer correct; the filter was considerably more damped, and as a consequence the –3dB point was higher. Clearly, you can’t just substitute a cathode follower in a circuit designed for op amps and have it perform properly.

**TABLE 1 INPUT IMPEDANCE OF SALLEN-KEY FILTERS**

| Damping                              | Minimum impedance | At frequency |
|--------------------------------------|-------------------|--------------|
| 1.000<br>(for 3rd-order Butterworth) | 0.89×R            | 1.06×f       |
| 1.414<br>(2nd-order Butterworth)     | 1.11×R            | 1.27×f       |
| 1.447<br>(for 3rd-order Bessel)      | 1.12×R            | 1.3×f        |
| 1.73<br>(2nd-order Bessel)           | 1.14×R            | 2.5×f        |

f = cutoff frequency, R = operating resistance

**TABLE 2 HIGH-PASS FILTER USING 6SN7 TUBE, DRIVING A 10K LOAD (GAMMA = 0.85×)**

| Damping                      | g     |
|------------------------------|-------|
| 1.447 (for 3rd-order Bessel) | 1.672 |
| 1.73 (for 2nd-order Bessel)  | 1.303 |



## LACK OF UNITY

How do you compute the response of a Sallen-Key filter using a buffer stage that's not unity-gain? If you know the actual gain of the active device, the equation for computing damping is:

$$5 \quad d = (2/g) + ((1 - \text{gain}) \times g)$$

(If gain = 1, then 1 - gain = 0, and the equation reduces to Eq. 4.)

I set up a 6SN7 as a cathode follower with a 10k load, a 27k cathode follower resistor, and a 750Ω bias resistor; the gain was 0.85x. It turns out to be easier to plug values into Eq. 5 and use it to compute d than vice versa. The results are summarized in **Table 2**.

It's easy to compute similar figures for other tubes: Measure the gain into the impedance it will be expected to drive, then plug it into Equation 5. A spreadsheet helps.

The values in this example were optimized for a 10k load. Some professional sound cards are set up to have a 10k load, balanced; that means, usually, a 5k load on each leg of the input. When the load changes, the gain of a cathode follower changes a bit, too. The 6SN7 described above has a gain of 0.83x with a 5k load. Not a big change; if you stick with  $g = 1.672$ , then  $d = 1.480$ .

Will that make a big difference? Not really; at 27Hz (low note on a piano) the response error will be about 0.07dB. Of course, if you know the load will be 5k, you can optimize for that.

Plug these numbers into the 2nd-order section for a 3rd-order Bessel filter, designed to drive 5k. As noted, if the overall 3rd-order cutoff frequency is 16Hz, the cutoff frequency for the 2nd-order section should be 11Hz, with  $d = 1.447$ . If you let  $C1 = C2 = 33\text{nF}$ , then the operating resistance is (ideally) 438.4k. Because  $g = 1.672$ , R1's ideal value is 262.2k and R2's is 733.1k. The nearest E96 values are  $R1 = 261\text{k}$ ,  $R2 = 732\text{k}$ .

**Figure 2** shows the final high-pass filter design, but what's C3 doing there? It turns out that you can't really use the grid resistor (in this drawing R3) as R2. It's bootstrapped to the output, which means its effective resistance is higher than the actual value of the resistor—how much higher is a function of the tube's open-loop gain. This varies from sample to sample, and changes with

time, so the effective value of R3 isn't predictable enough to use it in a filter. Adding a separate R2 from the grid to ground, of course, would mess up the DC bias point.

Instead, I've added C3. It isolates R2 from the grid, so the bias won't be affected. If the rolloff defined by C3 and R3 is sufficiently low, it won't affect the filter response appreciably; setting  $C3 = 0.47\mu\text{F}$  and  $R3 = 1\text{M}$  the -3dB point would be 0.34Hz, without bootstrapping, and about 0.03Hz with bootstrapping. (If I wanted a lower cutoff frequency, I'd increase C3 rather than R3; according to the 6SN7 datasheet, 1M is the maximum recommended grid resistance, and that figure is typical of audio-frequency triodes.)

Note that feedback resistor R1 comes directly off the cathode, rather than after the coupling capacitor. I can get away with that because its other end connects to the junction between two capacitors, with no DC path to anywhere else (provided the caps aren't leaking!) As long as C1 and C2 have adequate voltage ratings, this hookup will work fine—and it takes the output capacitor out of the feedback loop.

## HIGH CUTOFFS

So far I've assumed that you want to chop off only infrasonic material, leaving audible signals alone. Sometimes, and particularly with 78s, that's not true; there's rumble in the low audible frequencies, and you'd rather chop that off.

One possibility, of course, is to do the chopping in the computer, assuming that you're digitizing your recordings in order to clean them up and burn them onto CDs for convenient listening. That's easy; programs such as Adobe Audition ([www.adobe.com](http://www.adobe.com)) include clean high-pass filters which offer the choice of Butterworth (maximally flat), Chebyshev (maximum steepness with a peak at cutoff), and Bessel (best damping) responses. I use the Bessel, as I've mentioned before.

Sometimes, though, you want to do the chopping in the analog domain, for live listening sessions. If that's the case, then the way to do it is to alter the response of the high-pass filter in the preamp.

I've found that, for 78s, I can get along with high-pass filtering at a



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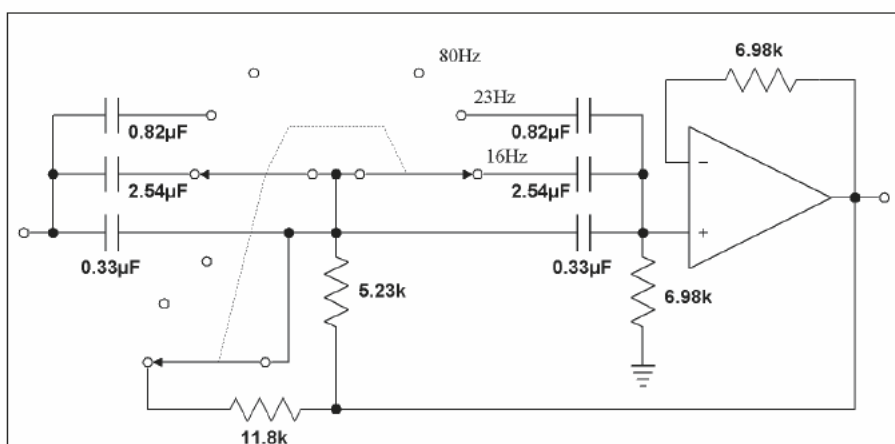
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**FIGURE 3:** Op-amp-based high-pass filter section with switchable frequencies. The switch, for a stereo preamp, is 6-pole, 3-position.

couple of additional frequencies. For acoustical 78s transferred flat, I find a 12dB/octave Bessel filter at 80Hz chops off low-end rumble from the cutting lathe without affecting the signal appreciably. (A typical acoustical 78 has a steep rolloff below 200Hz, so there's very little musical bass to affect.) Electrical 78s need more room; recordings featuring guitar accompaniment (such as early country music and country blues) typically work with a high-pass filter at 23Hz, again 12dB/octave Bessel. Recordings featuring piano or string bass, on the other hand, require the full low-frequency audio bandwidth, with a 16Hz cutoff.

**Figure 3** shows an IC-based high-pass filter with switchable frequencies of 11Hz, 23Hz, and 80Hz, with  $d = 1.447$  at 11Hz (befitting a section of a 3rd-order Bessel filter) and  $d = 1.73$  at 23Hz and 80Hz (at these frequencies, the section is functioning essentially as a 2nd-order Bessel filter, although slightly affected by the 12Hz passive rolloff elsewhere in the preamp).

The design, at first glance, looks unduly complicated. Why not simply design the basic circuit for 11Hz, then raise the frequency and change the damping simultaneously by adding parallel resistors to R1 and R2?

The reason is DC. If the amplifier has FETs on the input, that's not a problem; the input current will be very low, and the amplifier's DC offset won't vary when the value of the resistor running from the non-inverting input to ground changes. But FET-input amplifiers typically run dirtier

than bipolar-input amplifiers when used in the non-inverting mode. That's not an issue for low-level signals, but at the output of a phono preamp with 34dB of gain at 1kHz, you might see a +18dBu signal on a really bad scratch, and it's never a good idea to aggravate the annoyance of a scratch by adding distortion.

Therefore, I prefer to use a really clean amplifier for the high-pass filter; the cleanest amplifiers are bipolar-input, which make input bias currents an issue. The AD797 from Analog Devices ([www.analog.com](http://www.analog.com)) is probably the cleanest amplifier on the block in this application<sup>5</sup>, but it typically has an input bias current of 250nA.

Let's say I built a high-pass filter with  $C = 0.47\mu\text{F}$  and  $f = 11\text{Hz}$ ,  $d = 1.447$ , switchable via parallel resistors to 80Hz,  $d = 1.73$ . I'll spare you the math; R2 for an 11Hz filter is 42.5k, while it's 4.9k for an 80Hz filter. Assuming R3 remains constant, the input bias current from an AD797 would cause the output DC level to jump about 9mV when the parallel resistor was switched in or out. That's a pretty audible pop, about 30dB below nominal signal level.

The AD797 isn't the only clean amplifier around. You could use a Linear Technology LT1468 ([www.linear.com](http://www.linear.com)), which typically has a much lower input bias current, 10nA. Using that amplifier, the jump in DC levels would be only about 0.38mV, or about 58dB below nominal.

That's better performance, of course, but I still don't like having DC on the

outputs. There is a workaround: a resistor switched in parallel with R3. At that point, though, you're back up to needing six poles of switching for a two-channel preamp. That's no simpler than **Fig. 3**, which has the added advantage that the input impedance changes much less when the frequency is switched. That means the load on the preceding stage doesn't change appreciably—which is a good thing.

You can implement the switchable high-pass filter with a cathode follower, of course (I'll leave the calculations to you). While vacuum tubes have virtually no input bias current to worry about, the minimally changing input impedance of this design becomes much more important. Tubes, after all, care a great deal about what impedance they drive, and since a cathode follower high-pass filter is almost certainly preceded by another tubed stage, the relatively constant input impedance is a really good thing.

(There is, however, another issue with tubes: Will the cathode follower drive its following load cleanly? That's a whole different kettle of worms, which I'll address, after more lab work, in a future article—editor permitting!)

## WRAPPING UP

I hope this article gives you the sense that while designing high-pass filters for phono preamps isn't a trivial exercise, the subtleties are manageable, even if the active devices are tubes. It's worth doing the work to design these stages correctly, because they can add a great deal to the musical enjoyment available from LPs and 78s—which is what it's all about. **ax**

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\*Anechoic correction may have limited resolution at bass frequencies (20Hz - 200Hz), where room measurement and correction can be used.

## PHONO SOLUTIONS

I just received your December issue. The first time I have seen it—excellent. I am a hobbyist but not really electrically expert. I want to build the strobe light highlighted on page 20, but the picture looks as though there is a capacitor in the circuit not listed on p. 21 (under the resistor in the photo). Can you advise if this is the case and, if so, what the value is?

I also would like to build the speed control (p. 22) but am more a kit-guy than a total DIYer (from scratch). Any way to get parts lists or a completed circuit board?

John Bratten  
shelbcoop@gmail.com

### Ron Tipton responds:

Thank you for your comments and questions. What you see in Photo 7B under the resistor is another resistor; two resistors in parallel to get 165Ω.

### Joel Hatch responds:

Thank you for your interest in making the

Variable Frequency Synchronous Motor Controller. I have included the VFSMC parts list (Table 1). At this time, I'm not set up to make boards. If there is enough interest, I will provide *audioXpress* with the printed circuit board file. Good luck!

## READER REACTION

I have been a happy subscriber to your magazines for a long time and will continue to subscribe as long as breath, hearing, and funds allow. It is unfortunate that there are not longer-term subscriptions available now. I thoroughly understand the pressures on publishers of all kinds.

I have enjoyed listening to recorded music since I was about three and have been involved with the audio industry in one capacity or another since the age of 20. I was a practicing musician for a little while and would have continued if not for twists and turns in life.

I have been using computers since 1988 and (only) if necessary will accept the magazine in PDF format in the fu-

ture. In any case I am happy to keep the connection open.

Andy Newhoff  
red box design  
red.box@sympatico.ca

I have noticed, with pleasure, that your appreciated editorial is back in *audioXpress*. I hope you will continue because I find it gives readers a good idea of your day-to-day fight editing *audioXpress*.

I am personally convinced that the digital version of *audioXpress* is the best economical response to the world's publishing situation. Costs are reduced and the quality of the document, for the end user, is quite similar to the normal printed edition. In a changing world, we must adapt ourselves and our concepts permanently if we want to stay alive. This is a terrible rule that none can transgress.

Pierre Touzelet  
Vélizy, France  
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**TABLE 1: PARTS LIST FOR VPI TURNTABLE VARIABLE FREQUENCY SYNCHRONOUS MOTOR CONTROLLER**

| Part ID | Part Value   | Part Description                                                                 |
|---------|--------------|----------------------------------------------------------------------------------|
| C1-8    | 0.22 $\mu$ F | EPCOS B32559C224K00 or WIMA MKS20.22/63/5                                        |
| C9      | 1 $\mu$ F    | Kemet T330C475K035                                                               |
| C10     | 6.8 $\mu$ F  | Kemet T330C475K035                                                               |
| R1      | 61.9K        | Vishay Dale CMF5561K900FHEK                                                      |
| R2-5    | 36.5K        | Vishay Dale CMF5536K500FHEK                                                      |
| U1      | LTC1799      | Linear Technology LTC1799 IC oscillator with resistor set                        |
| U2      | MM74HC393    | Fairchild Semiconductor MM74HC393N dual 4-Bit binary counter                     |
| U3      | MAX7480      | Maxim 8th order lowpass Butterworth switched cap filter                          |
| U4      | TDA7396      | ST Micro 45W - 2 $\Omega$ bridge car radio amplifier with diagnostic facility    |
| U5      | LP2951       | On Semiconductor Low Power Low Dropout Voltage Regulator                         |
| VAR1    | 5K           | Front Panel potentiometer - Surplus Sales of Nebraska, similar Bourns 6639S-1502 |
| VR1     | 50K          | Bourns 3266W-1503LF variable trimmer potentiometer                               |
| W1-2    | FQ-1         | Wire connection pad 0.100 dia with 0.043 dia hole                                |
| W3-4    | TRANS-1      | Wire connection pad 0.100 dia with 0.043 dia hole                                |
| W5      | 120VAC-1     | Wire connection pad 0.100 dia with 0.043 dia hole                                |
| W6      | 120VAC-2     | Wire connection pad 0.100 dia with 0.043 dia hole                                |
| W7      | V+ 18        | Wire connection pad 0.100 dia with 0.043 dia hole                                |
| W8      | GROUND       | Wire connection pad 0.100 dia with 0.043 dia hole                                |
| T1      | Transformer  | Acme Electric/Amveco toroidal transformer 62050 (Digi-key TE-62050-ND)           |

#### AC cord

- 2-prong GFI plug from hairdryer
- thrift shop (no vendor)

#### Housing

- Radio Shack 350W AC to DC power inverter 22-155

#### Adapter board

- Proto-Board adapter for SOT-23-5 (to SIP) model 33205
- Phillips flat head machine screw 8-32-5/8 (McMaster Carr 91771A196)
- Phillips flat head machine screw 8-32-1 (McMaster Carr 91500A199)
- Machine screw hex nut, type 316 SS, 8-32 (McMaster Carr 90257A009)
- Front Panel knob - Surplus Sales of Nebraska, similar Eagle Plastic Devices 45KNO14-GRX

## Contributors

**Edward T. Dell, Jr.** (Editorial, p. 6) is editor/publisher of *audioXpress*, *Voice Coil*, and *Multi Media Manufacturer* magazines.

After a 40-year career in engineering and product management in consumer, industrial, and telecommunications electronics, **Ken Bird** ("The Wedge Slope-Back Speaker," p. 8) retired to his native Missouri, and now pursues his interest in building speakers and tube amplifiers, along with model railroading, amateur radio (W0KLB), and digital photography. His website is [www.klbird.com](http://www.klbird.com) and he can be found on Facebook.

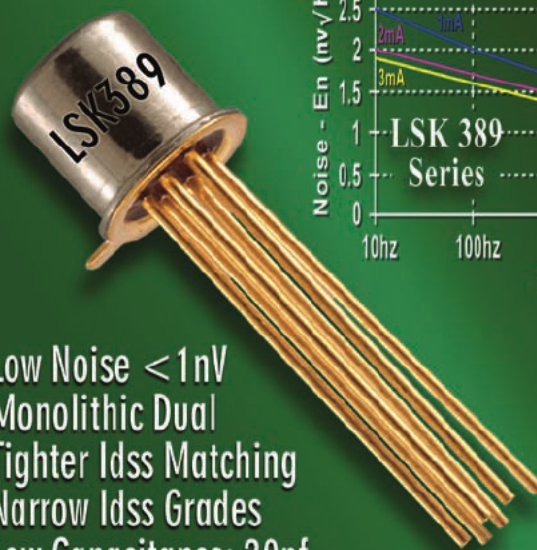
**Joseph Norwood Still** ("A New Tube Phono Amp," p. 14), retired from the electronics industry, designs affordable high-quality audio amplifiers. He thoroughly enjoys this hobby, which rewards him with many pleasant interchanges with dedicated, resourceful audiophiles.

**Darcy Staggs** ("The Return of the Disappearing Loudspeakers," p. 18) has been interested in electronics and its magic since building a crystal set in grammar school. This elementary model was later equipped with the first commonly available transistor amplifier, a GE 2N107. A college education (MS, engineering) was followed by work on the Apollo Lunar project (simulation of the electrical power system), war gaming, management software for Swedish IBM, weapons system simulation, and successfully raising two sons, who also love good music.

**Victor Staggs** ("The Return of the Disappearing Loudspeakers," p. 18) graduated from University of California Berkeley in 1963 with a BS in Engineering Physics, and worked in the nuclear defense industry from then until 1981. Along the way he obtained an MA in Mathematics from U.C. San Diego, and later an MS and Ph.D. in Communication Theory from the same institution. During this post-defense era, he has worked as a computer sales technical consultant, occasionally doing consulting jobs in audio and electroacoustics. He is now doing private research in phase equalization by digital signal processing. He has also done private research in theoretical acoustics, nonlinear optimization, and speaker design.

**Paul J. Stamler** ("Infra Dig: Passive Preamps, Part 3," p. 23) is a recording engineer/producer, musician, and technical writer; he also hosts a radio program, "No Time to Tarry Here," featuring traditional folk music and related stuff. He has delighted in 78s since he was a boy, when they were still being made.

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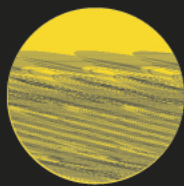


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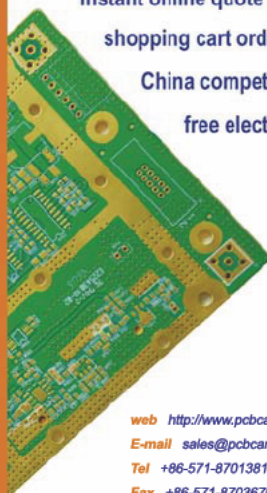
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# AES Nuggets from San Francisco

What's the mood of the audio industry, as analog equipment appears on the market in growing numbers?



PHOTO 1: A chance AES chat: John Curl (right) making a point, while Richard Small (middle) and Bob Cordell (left) seem skeptical.

With 14,000 attendees and some 300 exhibitors, the 129<sup>th</sup> AES convention was not as large as those of a few years ago, but the mood was upbeat and the gloom of last year was gone. The trend toward smaller booth sizes leveled the playing field, emphasizing technology rather than hype and dazzling displays. New convention programs such as Product Design Track and the expanded Live Sound, Broadcast/Streaming, and Game Audio Tracks drew interested audiences. Most exhibitors were quite pleased with both the turnout as well as the buying power of the attendees.

## WHAT GOES AROUND

You'd think that with the push to digital media, we'd seen the last of the dreaded

turntable and tape machine wow and flutter. Not so, says Ronald Quan (New Techniques for Evaluating Audio Amplifiers via Measuring for Induced Wow and Flutter and Differential Phase Distortions; Ronald Quan, Convention Paper 8194). He presented a measurement method that looks at the amplitude and phase modulation of an amplifier's transfer function as it traverses the signal cycle. Especially near the crossover point in push-pull amplifiers, there are relatively large discontinuities that give rise to these types of modulation. And, says Ronald, what do you know: it looks just like analog wow and flutter!

I had a chance to talk about it with a few persons whose judgment I value, and they reinforced my own view. The study did not uncover phenomena that

were not already known, and it did not provide new methods or tricks to build better amplifiers. Nevertheless, it provided a novel perspective on what goes on in our circuits, and as such will help to understand its causes and effects. Recommended reading.

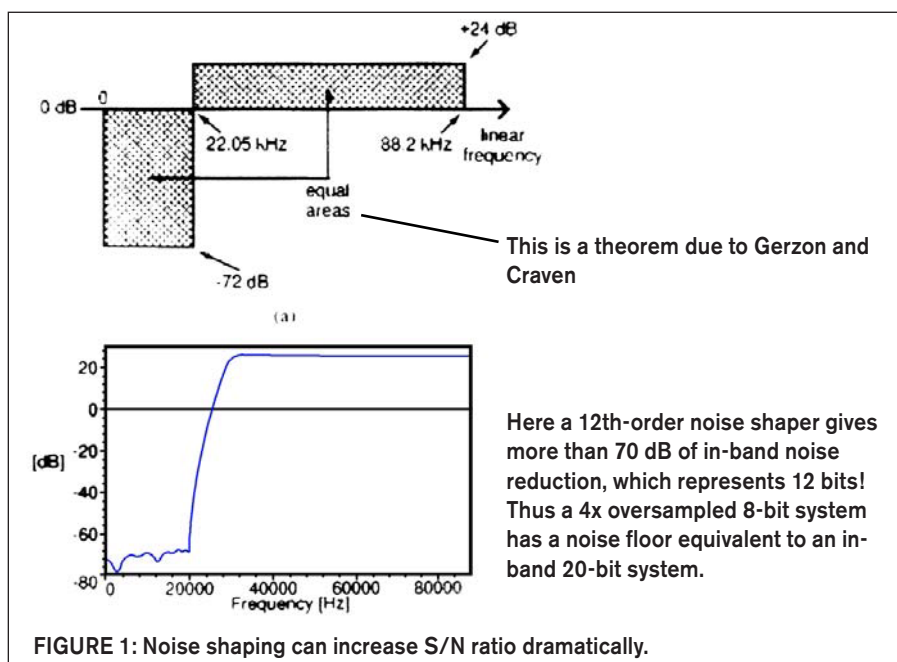
## ANALOG IS EVERYWHERE...AGAIN

In a previous report I borrowed this slogan from Analog Devices ([www.analog.com](http://www.analog.com)) to talk about jitter in analog circuits. This time it is more physical: literally, analog was everywhere.

While years ago there was a clear push toward digital workstations for anything from large-scale monster studio desks to the garage-recording set running on a PC, now I saw many, many pieces of recording equipment based on analog electronics. Stand after stand of racks with limiters, mike preamps, distribution amps, mixers, what have you, looking exactly like their forebears of 40-50 years back, including faithful replicas of the knobs and meter fascias. The circuits were vintage as well, replete with tubes, input- and interstage transformers. "Thermionic Culture," one large sign said, and that's exactly what it was.

## MIKES, MIKES, MIKES

In parallel with the analog equipment rebirth, there was an equally pervasive



proliferation of condenser microphones with a decidedly vintage look. It seems they become larger and larger, with some exceeding 10" in length. One manufacturer, Shure ([www.shure.com](http://www.shure.com)), even organized a live listening test.

A vocal/guitar artist did her thing while moving from one mike to another every half minute or so, to give the audience a chance to appreciate the particular "sound" of each mike. I'm not sure how useful it was, what with all the show background noise around you. But this (and other slogan signs such as "thermionic culture") seems to indicate that people readily agree to and accept the coloration of the sound by their equipment, and even use it as part of their artistic work. That's fine; an artistic expression should be whatever the artist decides it should be. But I still think I will design my own reproduction equipment to be as transparent as possible: minimum distortion, no coloration, what goes in needs to come out unaltered.

I don't think I should be second-guessing the musicians or the recording engineer. But what about tube stereo amplifiers that color and change the character of the sound? Is listening to

your music through tube reproduction equipment like looking at an original painting through a colored window, like retouching an original painting because you like it better your way? Maybe. And maybe you, as the consumer, are thus becoming a participant in the artistic performance. Food for thought!

## NEW OUTPUT DEVICES FOR ANALOG POWER AMPS

Interesting news from TT Electronics at Semelab ([www.semelab.com](http://www.semelab.com)): They are bringing out some new, high-performance output devices for audio. The bipolar range is extended with complementary pairs MAG633X and MAG941X. I saw a preliminary datasheet which showed  $h_{fe}$  versus  $I_c$  curves that were ruler-flat out to almost 10A. On some devices the Safe Operating Area (SOA) extends out to 100V at 2.6A, which is huge.

They also are extending their range of lateral MOSFETs for audio in the ALFET family. An interesting device is the new ALF08NP16V5, an N- and P-channel MOSFET in a single 5-lead TO-247. These devices can be run without a source resistor because of their negative tempco, without risk of thermal

drift. Worth taking a look at. <http://www.cfmediaview.com/click/click.aspx?db=6&obid=128033108&msgid=8538&linkid=1&redir=http://www.thatcorp.com/digmicpre.html>

## DIGITAL DOMAIN LISTENING TESTS

I always try to attend John Vanderkooy's presentations; although sometimes they're over my head, I always pick up interesting things. This year he talked about controlled listening tests comparing music signals in the digital domain. This can be useful for comparing sample rate converters, A/D converters, even digital interconnects. Switching between different signals in the digital domain rather than later in the analog domain after D/A conversion has advantages. For starters, if you compare signals in the digital domain, you can do away with the always suspect A/B switching boxes and relays or time-consuming cable switching. Selecting different digital streams can be done perfectly transparent to the bit streams.

Almost in passing, he also showed a drawing of the effects of noise shaping on digital signals. This technique, in combination with oversampling, lets you move noise from the audio band into a band higher up, where it is inaudible (Fig. 1). In this way, an 8-bit converter with 4x oversampling and 12-bit noise shaping can give a noise floor equivalent to a 20-bit system! Having heard noise-shaped systems many times, and understanding on an intellectual level how it works, I still haven't been totally able to get rid of that irrational "but this kind of processing *must* sound bad" feeling, so I fully understand the natural aversion people sometimes have to this type of signal manipulation. (This does tie in with Eric Valentine's presentation below.)



## COMPRESSED MUSIC LISTENING TESTS

This was a tutorial about “iPod listening.” We listened to various music excerpts at different compression ratios and schemes. It was very revealing listening to the difference between the original and the compressed version—in other words, listening to what was thrown away. Many times, the audience was amazed to learn how much was actually thrown out and how relatively good the compressed version nonetheless sounded. A tribute to the crafty use of our knowledge and understanding of how we actually perceive music.

In an A-B test, the differences between original and compressed were glaringly obvious, especially regarding loss of stereo imaging, loss of transients, lack of bass solidity. However, listened to in isolation, the compressed versions were not that horrible at all. There was also a clear difference between compression schemes. To me, AAC sounded better than MP3; WMA

removed generally less bass but more transient information. To make matters more complicated, these characteristics changed with different types of processed music.

## WHAT CAN WE REALLY HEAR?

A rare encounter: a record maker from Los Angeles, Eric Valentine, doing a tutorial on comparative listening tests. His quest for more insight began when he started to wonder whether these superhumans could really hear the difference between mike cables, different CD-R brands, SCSI drives, or FireWire replays? He trod a path which many of us have travelled: trying to find out why clear audible differences in sighted tests vanish when transitioning to a blind test.

As many of us have found, there’s precious little formal research available on this in audio, and just like many of us, he discovered research done in the area of (wine) tasting, where it was found that even with the same wine in

different bottles, the appreciation differed greatly depending on the label and the experts’ appreciation of that wine label. So far, nothing new.

Then he discovered research on the influence of non-sensory information on sensory perception, notably by Samuel McClure of the Neuroscience Department at Stanford. McClure did fMRI scans on people being subjected to blind and non-blind taste tests. His scans showed that the brain is modulating or filtering the sensory information being received to make it better match a particular expectation. That means that if someone has a particular expectation about what he/she is to experience, the brain will modulate the incoming information to match that expectation and *make the actual experience different*.

It’s not a matter of fooling yourself, or being deluded. You actually *do* experience a better taste for that wine! As Eric said, “This completely freaked me out.” Yet, after digesting this information and studying how and when he

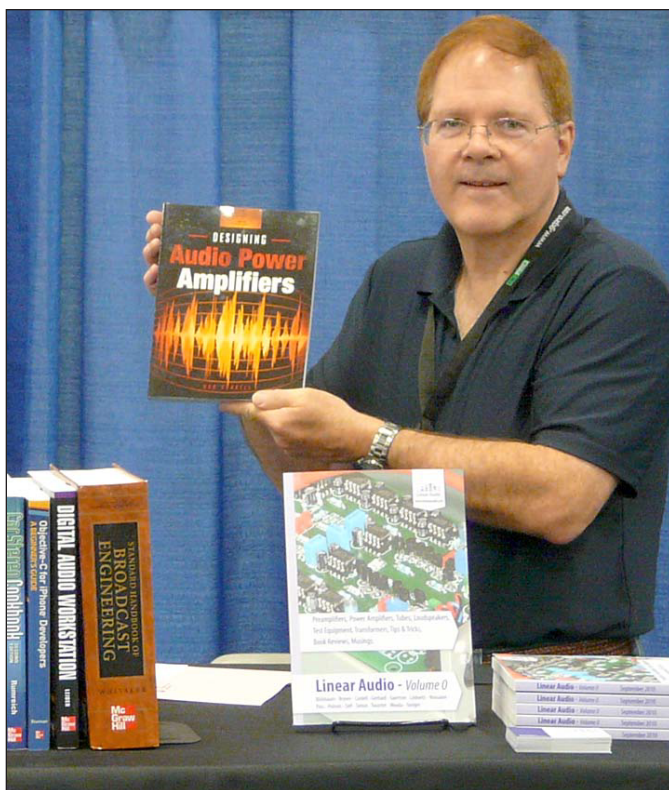


PHOTO 2: Bob Cordell proudly presented his book *Designing Audio Power Amplifiers*.



PHOTO 3: Ever more, and ever larger mikes.



and others could “hear” dramatic differences between equipment that vanished in blind testing, he says he is now much better in predicting when he will be able to continue to hear differences during the transition from non-blind to blind testing.

As an example, he noted that he is very familiar with NS10 monitor speakers, and this familiarity enabled him to discriminate between two types of capacitors in blind tests. Yet when turning to ATC monitors, which have more sub bass and with which he was much less familiar, he could no longer reliably hear the capacitor differences.

Eric is someone who hand-built his own mixing console. He literally knows each and every hand-picked component in the signal path, and is convinced that his console is the very best that can be built, and that it allows him to make better-sounding recordings. How does that jive with what was described above? Interestingly, he maintains that what gives him the edge is his belief that his console will produce the best recordings possible and *that* is a very strong motivator to work as hard at it as he can. So, in the end, hand-selecting these components and hand-building his console *does* help him make better recordings.

Extending the analogy, you could say that hand-selecting components and building your own amp makes for a better listening experience for the audio DIYer. Isn't that what we knew all along, whatever the issues with sighted and blind tests are?

Eric proceeded to do some blind testing with headphones, but with 50+ noisy people in the room, I didn't believe it would be really “blind,” so I left it at that. However, I will reveal his results of listening tests done at his AES booth.

The test was between two converters, a Lavry and an Mbox2 pro, with a price ratio of close to 1:10. Booth visitors could listen to them through head-

phones and were asked which one they favored and why. At the sighted test, 80% chose the more expensive and shinier Lavry over the Mbox. When the test was done blind, 78% favored the Mbox. The reasons given for the preference were also interesting. The arguments for preferring the Lavry in the sighted test were very similar to those for favoring the Mbox in the blind test (although one listener actually mentioned the “more shiny” Lavry as the reason he/she liked it more in the sighted test!).

This all begs the question: Why

bother with controlled testing at all? If I enjoy my choice, and you enjoy your choice, isn't that what counts? True, but on the recording side, huge amounts of money are spent to buy the best equipment available for the budget. Whatever the consumer uses to listen to the recording, it's important to get the basis right, to produce recordings that are transparent, uncolored, with low distortion. Objective, realistic evaluations are important for that, and controlled tests are a good tool to help make the right buying decisions.

PHOTO 4: Retro-analog was everywhere.





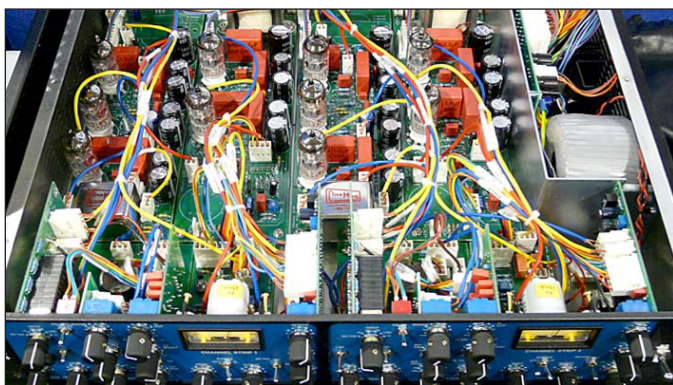


PHOTO 5: "Thermionic Culture" but with up-to-date manufacturing and packaging technology.

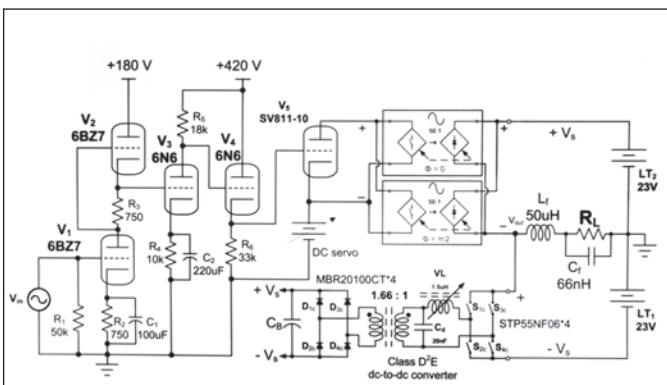


FIGURE 2: Thinking out of the box.

## OUTSIDE THE BOX DEPT

Yes, that output transformer in the tube power amp is the weak point and it would be nice to be able to replace it with something better and cheaper. But to replace it with a switching converter? That was exactly what a student from Korea did. A prime example of thinking outside the box, big time. Interesting as an intellectual challenge, see Fig. 2.

As I mentioned before, this was not the largest AES convention ever. But it was a convention with a positive, can-do mood with innovations, new products, and new trends. I was fortunate to share a booth with Bob Cordell, who presented his major tome, "Designing Audio Power Amplifiers," which enabled me to meet many old friends, make new ones, and present my own new publication

"Linear Audio."

With music, in whatever shape or form, forming an ever-larger component of our social and cultural life, those who produce that music and its distribution and reproduction cannot help but continue to play a major role. The AES has taken the downturn in stride and is brimming with energy and new ideas. This convention was indeed a bridge to the future. *ax*

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# Headphone Testing

## Part 1: The Basics

As more and more loudspeaker engineers find themselves employed in the fast-growing headphone market, either through company diversification or changing jobs, it is important that the unique challenges of testing headphone packages are fully understood. Many of the characteristics that make for a good in-room listening experience with a loudspeaker—good frequency response, low distortion, no rub & buzz or loose particles, and so on—also apply to headphones, and many of the principles of loudspeaker testing apply. However, there are some major differences and additional issues that need to be considered. These include couplers and associated correction curves, acoustic seal, fixturing, and additional tests such as L/R tracking.

In this article I discuss the issues that are common to testing all types of headphones. In part 2 I will address the specific needs of special cases of headphones such as Bluetooth and USB headphone testing, noise-canceling headphones, and maximum SPL measurements to prevent hearing loss.

### SIMILARITIES AND DIFFERENCES

First, I will describe the similarities in testing loudspeakers and headphones. The setup essentially consists of an electroacoustic measurement system, some kind of ear simulator containing a reference microphone, and the de-

vice under test. A test signal is sent to the transducer (headphone), which in turn is measured by a reference microphone in a coupler.

The basic measurements made on headphones are very similar to those made on loudspeakers. These include frequency response, phase (polarity),

distortion (THD and rub & buzz), and impedance. In both cases, the test signal is usually a swept sine wave, and the level can vary. Some set the drive level to achieve a certain sound pressure level at a given frequency; others choose the level that equates to 1mW of power. Certain products may neces-

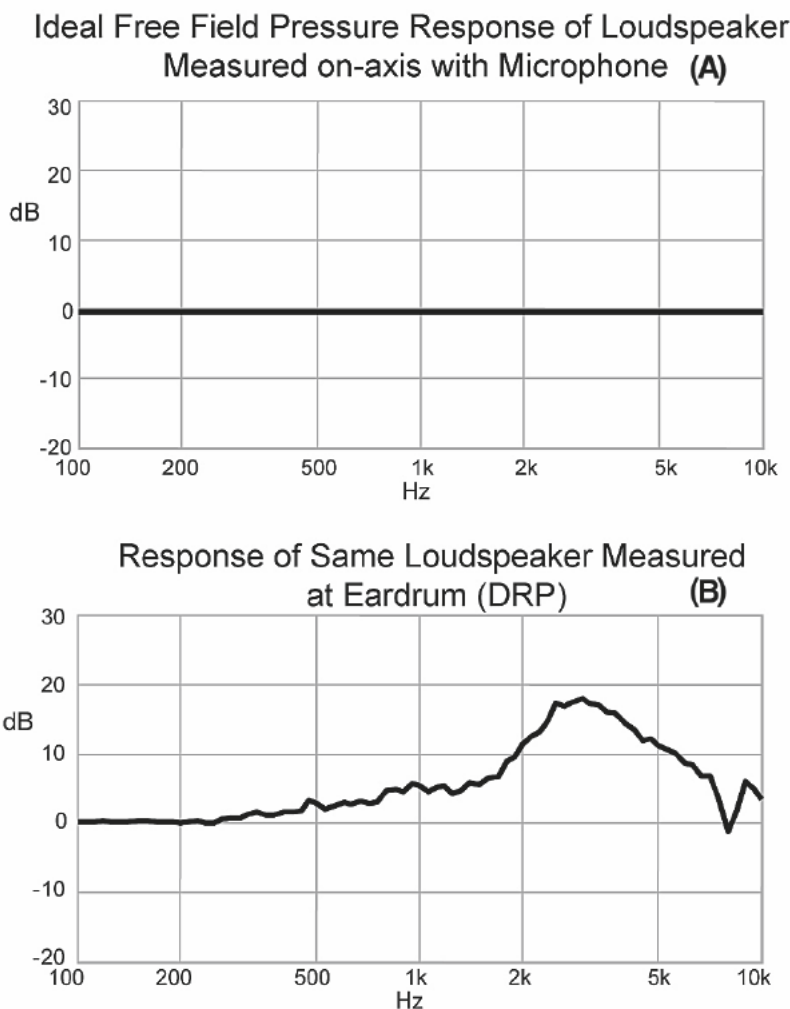


FIGURE 1: Difference in response between measuring with a microphone in the free field and measuring at the drum reference point.



sitate testing the frequency response at one level and performing a second, higher level test for distortion.

The primary difference in the test setup between a loudspeaker and a headphone is the method in which the transducer interacts with the microphone. Whereas loudspeakers are tested in open air, a headphone or earphone must be presented with an acoustic load that simulates the human ear. It is common to compare the left and right channel frequency response. Large differences at certain frequencies can be very audible in a stereo device, even though the individual responses may be within specification. Sometimes, electrical characteristics such as crosstalk may also be measured.

Before beginning to test headphones, you need to consider two major factors: correction curves and the acoustic seal. These both have an effect on the frequency response, and the latter also affects the repeatability of measurements.

### COUPLER CORRECTION CURVES

Loudspeaker engineers are familiar with the ideal frequency response for a loudspeaker measured in the free field being a flat line (**Fig. 1A**). For headphones, however, this is not the case. Headphone measurements are taken at what is known as the Drum Reference Point (DRP)—a point representing the human eardrum (**Fig. 2** shows where this is on a Head & Torso Simulator (HATS)). If you were to measure the same loudspeaker that produced the flat free-field response curve in **Fig. 1A** at the DRP, the frequency response would look like **Fig. 1B**. In other words, for a headphone to sound like a loudspeaker with a flat frequency response, it must produce a frequency response curve as shown in **Fig. 1B**.

This frequency response curve is a correction curve, or transfer function

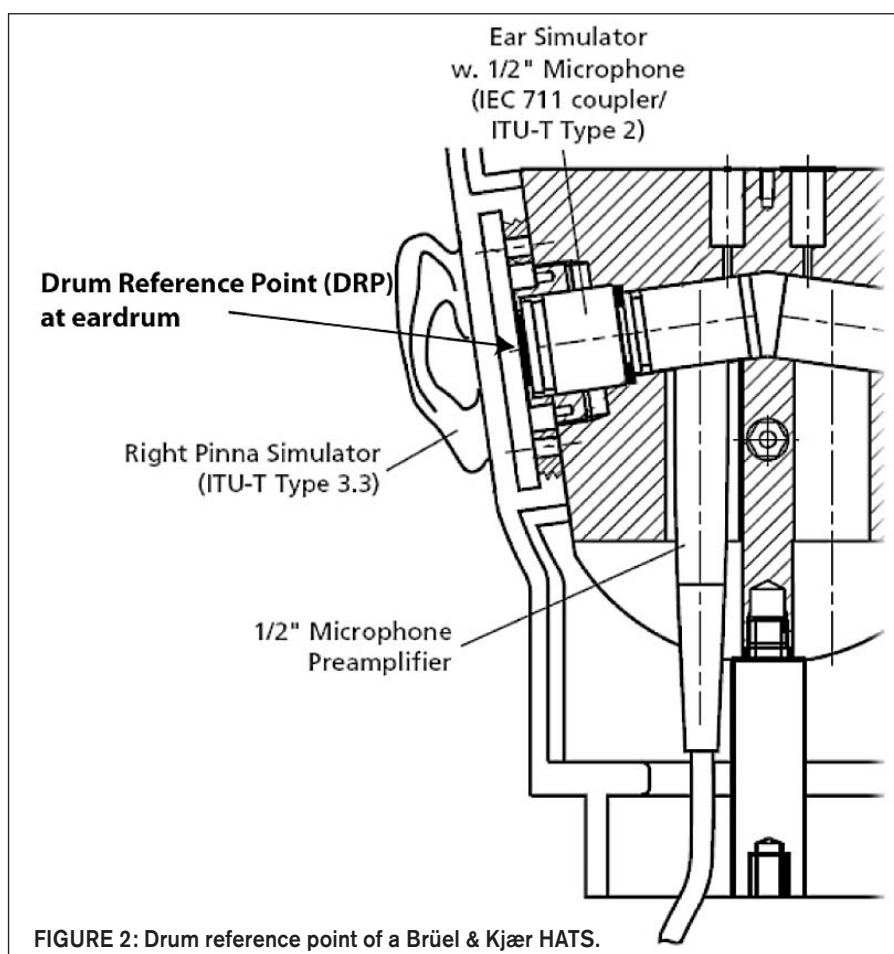


FIGURE 2: Drum reference point of a Brüel & Kjær HATS.

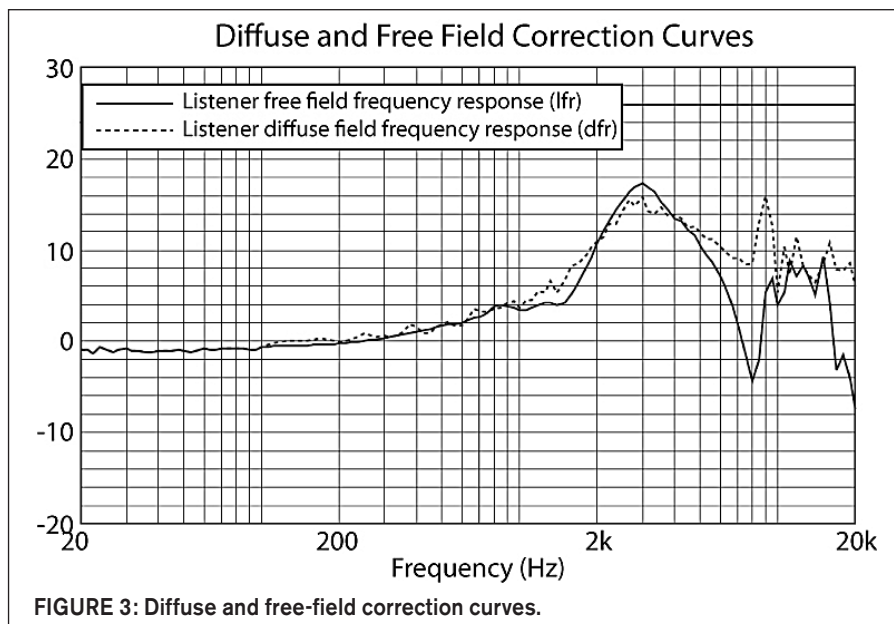
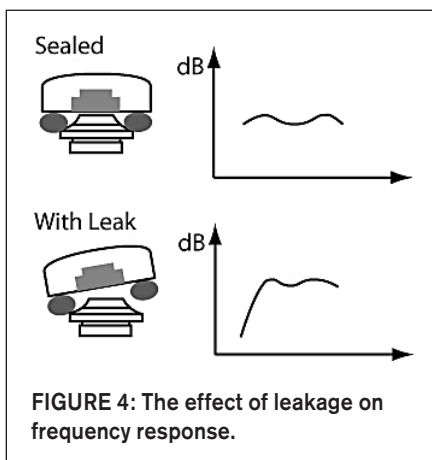


FIGURE 3: Diffuse and free-field correction curves.

that represents the effects of the head, torso, pinna, ear canal, and ear simulator. To further complicate matters, different correction curves are applied according to whether your measurements are made in the free field (an-

echoic room) or diffuse field (reverberation room) (**Fig. 3**). For the most part, as with loudspeaker measurements, the free field is used. Typically when making measurements, the subtraction of the correction curve from



the actual measurement can be carried out in your test software, so that your output frequency response is shown as the familiar straight line.

Another issue that needs to be addressed when testing headphones is the acoustic seal, or leakage. Realistic headphone measurements (using a HATS or other ear simulator) have a certain degree of leakage because the headphone does not fit tightly to the pinna. This has an effect on the frequency response, with a demonstrable loss at low frequencies (Fig. 4). Although realistic, it affects the repeatability of measurement. In the R&D lab, this is compensated by repeating the measurement multiple times, removing and repositioning the headphone between each measurement and averaging; on the production line different couplers and fixtures are used to offer a more repeatable seal—I explain these in more detail below.

## DIFFERENT TYPES OF HEADPHONES

Before looking at test configurations for headphone testing, I will examine exactly what I am measuring. Headphone is a broad term that covers several different designs of product, each with its unique testing challenges.

Headphones fall broadly into four categories (Fig. 5): circum-aural (a large cup that completely surrounds the ear and pinna), supra-aural (an earpad that sits on the pinna), earbuds (also known as supra-concha), where the transducer rests at the entrance to the ear canal, and in-ear, where the sound port sits inside the ear canal. Although the measurements that need to be made are the same for each of these, the fixturing—a critical part of the test setup—is different for each type.

Figure 6 shows a headphone test setup (suitable for all types of headphones) using a PC- and soundcard-based mea-

surement system. Although you may, of course, also use hardware-based systems, a soundcard offers more than sufficient accuracy for testing headphones, and such a system is usually less expensive. As you can see, the only part of this setup that will be unfamiliar to those accustomed to loudspeaker measurements is the “black box” that represents a device that simulates the ear (and replaces the microphone in a loudspeaker setup).

## WHAT'S IN THE BLACK BOX?

The equipment in the black box can be a multitude of things, depending on whether you are testing in the R&D lab or on the production line, your budget, and the type of headphone you are testing. Essentially it will be the microphone-containing device you have selected which simulates the human ear with some degree of accuracy, and appropriate fixturing to ensure repeatable results with the particular headphones you are testing. This is perhaps the most complex part of this setup, and the one that poses the most challenges to engineers. In order to select the most appropriate ear simulation device, the first question is whether the test should be a realistic simulation of actual use (commonly performed in development) or a highly repeatable test capable of differentiating defective units from good ones (production).

When developing a product, it is desirable to have a means of measuring under conditions that the end user will experience. A head and torso simulator provides this level of simulation, because it is equipped with artificial ears, which mimic the acoustic characteristics of the human ear, as well as artificial pinnae, which mimic the way a headphone would fit on a human. The fit of headphones and earphones can vary from person to person and even from one use to another, and can drastically



change the user's listening experience.

Testing on a head and torso simulator can reveal this variability between fit and acoustic performance. This lack of repeatability is realistic and useful to understand how fit impacts the sound quality. Engineers will commonly take several measurements and average them to account for it. Head and torso simulators are expensive (upwards of \$20,000), so some R&D laboratories use less expensive (but with the expected performance trade-offs) alternatives such as cheek

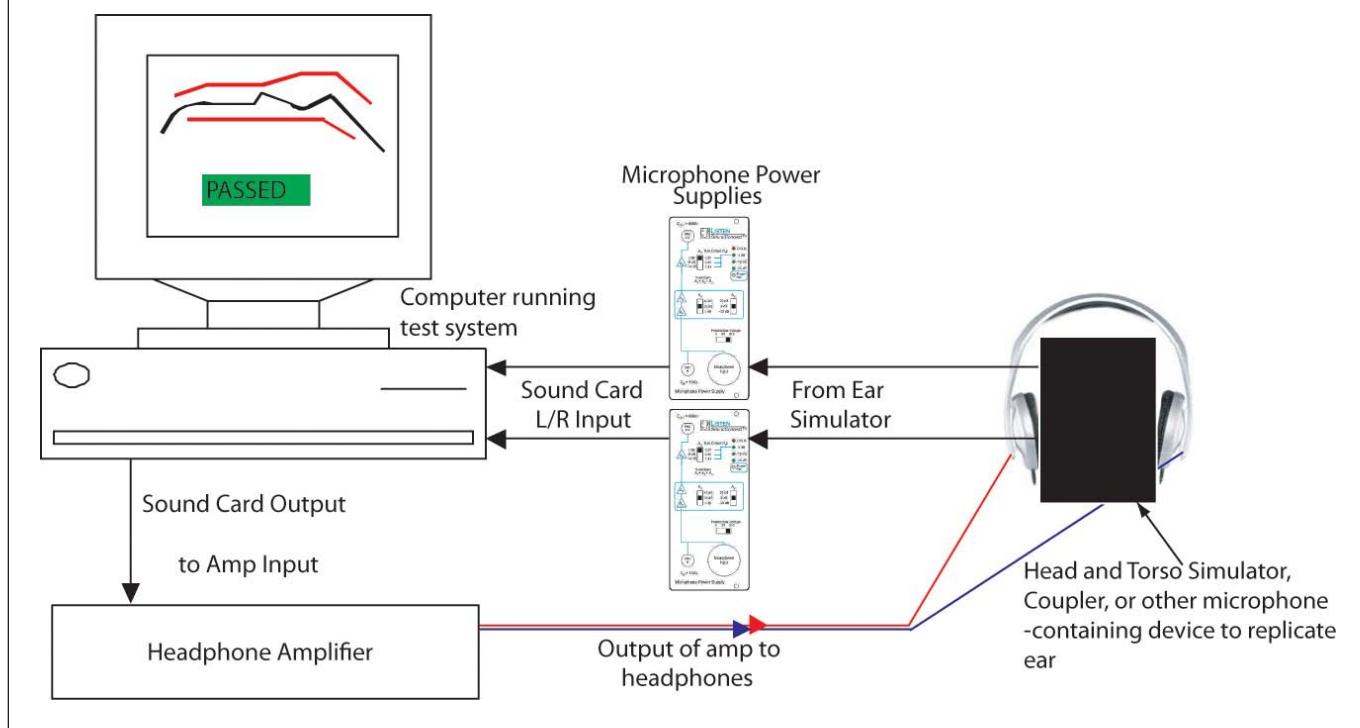
and ear simulators, couplers, simplified pinnae, and so on.

Head and torso simulators are not suitable for production line use for two reasons. As discussed, the fit may be slightly different each time, which makes it very difficult to achieve repeatable results on the production line, and also the cost is prohibitive. This is overcome by designing a product with a known response using simulators (such as the Head & Torso) that accurately replicate a human, and then ensuring that all products coming off

the production line match that ideal product. This means that different test methods can be used.

In production and QA it is important to use a fixture that is highly repeatable and can produce consistent results. An acoustic coupler, which is essentially a metal chamber that replicates the ear canal, is most commonly used. Couplers vary significantly in their complexity and cost. The most commonly used production coupler—a 2cc coupler, which is simply a 2 cubic centimeter cylindrical chamber—approximates the

**FIGURE 6: Headphone test setup.**



**FIGURE 7: Acoustic couplers.** In order of increasing complexity and cost, from left to right: 2cc coupler, IEC 711 coupler, pinna with simplified geometry, and a Head & Torso Simulator (courtesy of Brüel & Kjær and G.R.A.S.).



ear canal impedance.

More complex (and therefore more expensive) is an IEC 711 coupler, which has multiple internal chambers that more accurately replicate the acoustic impedance of a human ear. Although the characteristics of this coupler are close to the human ear in the lower frequency ranges, above 8kHz it is not well defined.

No manufacturer has produced a coupler that offers a truly accurate representation of human hearing above 8kHz because human ears vary more at high frequencies. For manufacturers who want to go one step further than a coupler, a pinna with simplified geometry is available (Fig. 7). Although not shaped like an ear, it is built to allow some degree of replication of the acoustic leakage that would occur with a real ear.

Even with the simpler geometry of a coupler, it is still difficult to achieve repeatable results, and most production line applications rely on custom fixturing to offer greater repeatability of mounting for a controlled seal. Usually a fixture is custom-built for a specific product to ensure repeatable attachment to the coupler. For circum-aural and supra-aural headphones, some sort of clamp is used to apply a consistent pressure, which aids repeatability. Earbuds and in-ear earphones typically use a gasket of some sort to produce a seal, which produces more accurate low-frequency response and has the added benefit of reducing the influence of ambient noise on the test.

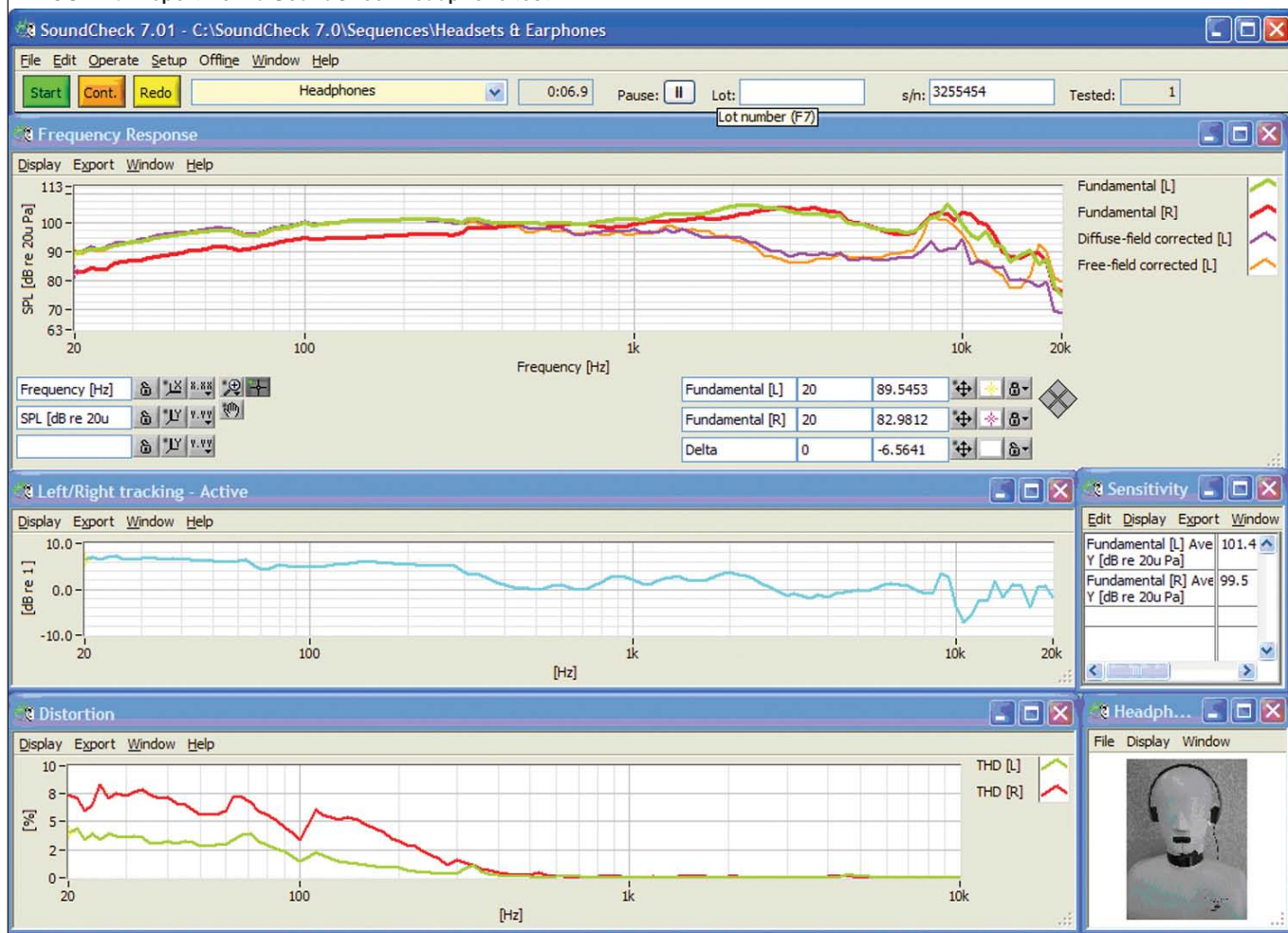
## HEADPHONE TESTS

Now that you understand the test setup, I'll examine what you typically

measure with a headphone. If you are buying a new test system for testing headphones, you should consider what and how you want to test before purchasing, because it has a bearing on the number of inputs and outputs you will need on your soundcard or test system. If you are testing a stereo pair simultaneously, you will need a two-channel soundcard just for the acoustic tests. If you want to also simultaneously measure electrical characteristics such as impedance, you will need four channels. If you already have a two-channel system, you can measure impedance sequentially rather than simultaneously, although it will be a little slower.

A typical basic R&D test using a head and torso simulator for a stereo headphone pair would consist of playing a 1/12 octave stepped sine sweep from 20-20kHz and measuring the

FIGURE 8: Report from a SoundCheck headphone test.





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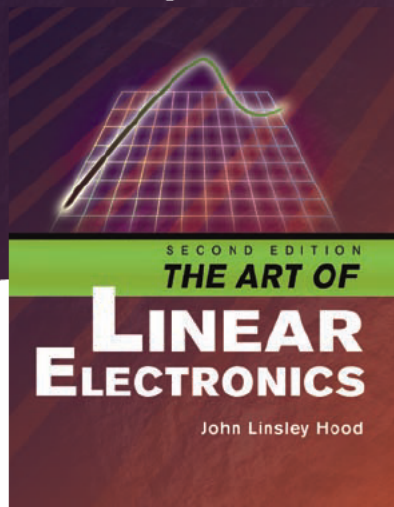
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harmonic distortion and fundamental frequency response. Correction curves would compensate for the free field response from 0° incidence to the nose to the eardrum. You can then use post-processing to compare the left and right earphone responses and show the difference curve, both for magnitude and phase.

With a four-channel analyzer, you can expand this test with the addition of a couple of reference resistors to measure impedance of the headphones at the same time. Other options for more detailed testing would include measuring intermodulation and difference frequency distortion, or measuring maximum SPL using the distortion level to set the upper limit. Naturally, each company wants to test its products slightly differently, and most test and measurement systems offer the ability to customize tests.

A typical production line test would

be similar, but you would need to apply a custom compensation curve, based on the coupler and fixturing. On the production line, impedance, if measured, would typically be done simultaneously for speed reasons. **Figure 8** shows an example test report for a R&D headphone test carried out in SoundCheck, a soundcard-based test system that is widely used both in the R&D lab and on the production line for measuring headphones.

## CONCLUSION

IEC 60268-7: “Sound system equipment—Part 7: Headphones and earphones” is probably the most comprehensive standard on headphone testing. It discusses about how to classify earphone types; e.g., supra-aural versus supra-concha describes various couplers and acoustic ear simulators, defines free-field versus diffuse field conditions, and explains how to test headphones.

For example, it specifies the standard test level to be 94dB SPL at 500Hz or 1mW at the earphones’ rated impedance and defines how to measure the rated impedance. A copy of this standard costs approximately \$200, and may be purchased from the IEC website ([http://webstore.iec.ch/webstore/webstore.nsf/Artnum\\_PK/43714](http://webstore.iec.ch/webstore/webstore.nsf/Artnum_PK/43714)).

This article has covered the basics of headphone testing, focusing on those issues that are applicable to all types of headphones. In part 2, I will discuss special cases of headphone testing, specifically Bluetooth and USB headphone testing, noise-canceling headphones, and maximum SPL measurements. These are each complex subjects requiring the use of different test signals and compensation for time delays, dropouts, and other characteristics uniquely inherent to these special types of headphone. *ax*



**FIGURE 9:** Mark Brezinski, editor-in-chief of Headphoneinfo.com, part of the Reviewed.com family of product review websites, uses the setup described here with a head and torso simulator to carry out objective measurements of the audio quality of many different brands of headphones. The audio test forms part of a complete battery of tests which also examine noise cancellation and sound leakage. They also conduct additional tests that provide readers information on the comfort, usability, and design of headphones (credit Richard Baguley/Reviewed.com).

## CONTRIBUTORS

**Jan Didden** (“AES Nuggets from San Francisco”) built his first OTL amp with 807 tubes 35 years ago. He has built speakers, preamps, and tape recorders, but is most interested in power amps, especially using error correction as discussed by Hawksford. Many of his projects have been published by *audioXpress*. Now retired from a career with the Netherlands Airforce and NATO, he tries to complete all those half-finished projects accumulated for lack of time. He now also has the time to travel to interesting audio events and interview audio luminaries. His projects are documented on his Linear Audio website, [www.linearaudio.nl](http://www.linearaudio.nl).

**Steve Temme** (“Headphone Testing: The Basics”) is Founder and President of Listen, Inc., a manufacturer of software-based measurement systems for testing electroacoustic and audio electronic products. Prior to founding Listen, he worked for many years as an acoustic test and measurement applications engineer for Brüel & Kjær, and also as a loudspeaker design engineer for Apogee Acoustics. He has authored numerous papers on acoustic testing and lectured extensively throughout the world. [www.listeninc.com](http://www.listeninc.com).



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