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Editorial

Looking Backward

Every craftsman and every enthusiast evolves. This is not an essay about Darwin, it is a call for reminiscence into the path your development has taken. I would guess that the general rule that things must grow or die applies to the audiophile just as much to any other person with a special interest.

Recently I have found myself exploring the lives of composers whose music I can readily retrieve from my music library. Even though I keep copies of Scholes and others handy near my listening chair and check basic biographical data about the great ones, I have begun to do some serious reading about some who interest me most: Mozart, Beethoven, Mahler, Berlioz, and Ives. (My spell checker is telling me it doesn't know who those last two are.)

In a similar way, my knowledge of audio began a long while ago, mostly from reading *High Fidelity* and its DIY offspring *Audiocraft*. *High Fidelity* came along in the summer of 1951. *Audiocraft* began publication in November of 1955. In its earliest issues *HF* reflected the reality that high fidelity could be a do-it-yourself enterprise. The appearance of *Audiocraft* confirmed that assumption—at least for the next three years until Charles Fowler's publishing company was sold to a larger corporation.

The Heath Company was probably the most important influence on anyone eager to pursue audio as a hands-on hobby. The original Radio Shack in Boston was another influence. It had a special department for high-quality sound in the 1950s. It included Klipsch; Radio-Craftsmen; Brook tuners, amps, and preamps; Bozak; Janszen electrostatics; Ampex tape recorders; and REL FM tuners.

My first amplifier was really a small public-address type with controls and 6V6 outputs—and it was cheap. It came as a kit by Heath and could be put together, point-to-point, in a few evenings. It worked perfectly the first time I turned it on. The lure and magic of building electronics all flowed from that first experience.

Over time, like most enthusiasts, I moved on to speakers by Briggs using Altec drivers, Electro-Voice's kit of parts for a Klipschhorn, and Bozak pairs of 12″ drivers with Wharfedale midranges and Janzen tweeters. Dynaco came along with a kit at a dollar a watt, and really splendid output transformers. As time goes by it all becomes more and more exciting, more complicated.

We built tape decks with heads from Nortronics and DIY circuit boards published in *Radio-Electronics*. Gordon Holt agreed to publish an experimental Stereo 70 project in *Stereophile* with full cooperation of Dynaco, who supplied the output transformers. Almost everything else in the project was military surplus.

As you can see, the progress was toward greater and greater complexity. This happens to most of us, as we leave the simple behind. The technology has also evolved, of course, and today your range of knowledge as an amateur must be significantly larger than it was in the simpler days of the 1950s, when many of us began our journey.

I suspect that a large part of the appeal of vacuum tubes lies in its simpler—and more accessible—technology. Not only have tube enthusiasts returned to the earlier forms of amplification, but they have skipped back over the later developments of pentode, push-pull, and capacitive coupling to formats of the '30s and '40s, before "hi-fi" was a movement.

Solid-state enthusiasts, particularly the professionals, were "glad to see the back of" the old, bulky, hot, and deteriorating-over-time bottles. Naturally, there will never be a resolution of the tube/solid-state discussion. We all have our reasons for our preferences. In these pages I trust we can recognize that and all of us remain reasoned and reasonable on the subject.

I encourage you to think back over your own journey as an audiophile. Some of what you have left behind may just be worth revisiting. Those of you who write for these pages might consider that some newcomers here could use your earlier experiences and your simpler projects as learning tools for their own growth. Everyone must begin somewhere, and our big, complex projects may be too far up the ladder for beginners. We welcome questions from beginners and also lists of your needs for better system response.

This magazine is, more than anything else, a central exchange for ideas, experience, and knowledge. Yours is, as always, welcome.—**E.T.D.**

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> *The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.*
>
> JOHN STUART MILL

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Audio News

■ DYNAUDIO AUDIENCE SERIES

Dynaudio introduces a new edition of its Audience Series loudspeakers, with a broader range of applications and improved performance. The new Audience series ranges from bookshelf models (the Audience 42 and the Audience 52) to floor-standing towers (models 62, 72, and 82). The new Audience 122C Center and floor-standing Audience 122 loudspeakers are complemented by additional models designed for home theater and lifestyle applications; in particular, the compact Audience 42C Center and the shallow-depth, front-ported Audience 42W wall-mount main/surround speakers. Contact J.B. Stanton Communications, (860) 542-1234, FAX (860) 542-0005.

■ 12-SE SUBWOOFER

RBH introduces the compact 12-SE subwoofer, which combines a discrete 300W amplifier with a 12″ proprietary metal cone woofer and is capable of reproducing powerful bass information below 30Hz. The 12″ aluminum driver is a front-firing design that vents out of a specially engineered, high-performance downward-firing port that eliminates "chuffing." The 12-SE also features a variable crossover (40-100Hz), switchable phase, and solidly built cabinet using MDF construction with internal bracing. RBH Sound, 967 N. McCormick Way, Layton, Utah 84041-7261, (801) 543-2200, FAX (801) 543-3300, www.rbhsound.com.

■ TENMA OSCILLOSCOPE

Tenma Test Equipment announces the introduction of their new Model 72-6800 20MHz Oscilloscope. It features a 20MHz bandwidth with dual channel and dual trace operation, 6″ rectangular CRT with internal graticule, high sensitivity of 1mV/division, automatic and external triggering, and special vertical and horizontal triggering modes for stable TV signal observations. The Model 72-6800 also includes two 10:1 oscilloscope probes and an owner's manual, and ranges from 10MHz to 200MHz. Available from the Tenma website at www.tenma.com, or from MCM Electronics, (800) 543-4330, www.mcmelectronics.com.

■ THE NEW 300B EH

Electro-Harmonix now includes a high performance 300B in its new EH tube line. Like the original Western Electric 300B, the 300B EH has an eight-course directly heated filament, 40W plate dissipation, classic large-shoulder tube glass, and is available singly or in precision matched pairs. The 300B EH's plate curves and static DC characteristics exactly match the WE originals, so you can drop them into any vintage or modern 300B amp. Electro-Harmonix, (212) 529-0466, FAX (212) 529-0486, info@ehx.com, www.ehx.com.

■ ICC UPDATE

ICC has recently released its updated list of entertainment, receiving, and special purposes tubes. Over 2,000 quality tube types include rectifiers, power tubes, premium 10-M series, matched pairs, voltage regulators, magnetrons, and many others. For more information, contact International Components Corporation, 107 Maxess Road, Melville, NY 11747, (516) 293-1500, FAX (516) 293-4983, www.icc107.com, e-mail: HG@icc107.com.

■ IVEX INSTANT QUOTE SERVICE

Ivex International announces a new instant quote service available with WinBoard PCB. Anyone seeking a quick price quote and a quick turnaround making their printed circuit board can now use WinBoard 2.25 for instant access to PCB Express on-line. PCB Express is available from ECD Inc., and offers a variety of express board manufacturing options and secure on-line transactions. Ivex Design International, 15232 NW Greenbrier Pkwy, Beaverton, OR 97006-5746, (503) 531-3555, FAX (503) 629-4907.

■ "K SERIES" SPEAKERS

Miller & Kreisel Sound (M&K) has introduced the "K Series," consisting of two satellite speaker models and a powered subwoofer, allowing for flexible system configurations to satisfy multiple size, performance, and budget requirements. The K-5 Satellite Speaker is compact (7 ⅛″H × 4 ⅛″W × 5 ⅛″D) and uses a full M&K phase-focused crossover network to feed its 1″ neodymium dome tweeter and 4″ cast-basket treated paper cone woofer. For more performance, the slightly larger K-7 satellite delivers greater output, power handling, and dynamic range, using a second treated paper cone woofer. Completing the K-series system, the K-9 powered subwoofer uses a special 8″ long-throw treated-paper subwoofer driver. The units are available with radiused edges and a black/titanium finish. Contact M&K at (310) 204-2845 or check out their website at www.mksound.com.

A Pair of Computer Speaker System Designs, Part 1

Who says smaller is better when it comes to computer speakers? These easy-to-build models are a little larger than you'd normally find, thus offering improved performance. **By G.R. Koonce**

Computer speakers are relatively small speaker systems built with shielded drivers so you may set them close to TVs and video monitors without causing picture problems. The speakers described here are bigger than the little plastic speakers that come with computers, offering the benefit of improved performance. These systems were originally developed as projects for the book I wrote with David Weems[1], but were omitted because of space limitations. A requirement for the projects was that construction be kept simple, making these systems as easy to build as a small box allows.

Computer speaker type A (CSA) is a two-way system in a small vented enclosure, using a 5″ (13cm) woofer, the Vifa M13SG-09-08. This driver has been used in closed box projects in *Speaker Builder* ("Simple High-Quality Computer Speakers," *SB* 8/97 and "Audio Video Revisited," *SB* 6/98) while CSA uses the larger vented enclosure for better bass extension. CSA (*Photo 1*) is the better of these two systems for low-level listening.

Computer speaker type B (CSB) (*Photo 2*) is a two-way system using the Eminence 615F2025 6½″ (16.5cm) woofer in a closed box that is small for this size of driver. The bass extension would be better with a vented design, but the closed box reduces the size and provides the ability to play loudly. Used with a subwoofer these boxes would make a full-range system, with the response of the CSB system used as the high-pass crossover. They sound good alone at reasonable levels, but are not the best choice for low-level use, as with a computer.

All the drivers I used in these systems are available from various sources, with the exception that you can obtain the Eminence 6½″ woofer only from Martin Sound Products. All the air-core coils used in the crossovers came from Madisound.

SYSTEM-DEVELOPMENT APPROACH

I developed the enclosures by measuring the woofer Thiele/Small (T/S) parameters so I could determine the box size via a computer program; measured bass performance will be shown. I then measured the various drivers in a test baffle with Liberty Instrument's Audiosuite and exported their impedance and frequency response files. These files were then fed to a crossover modeling program (CMP.Exe) for crossover development. The concept of crossover modeling is described in "Modeling for Designing Passive Crossovers," *SB* 8/98, p. 20. CMP.Exe is included with our book.

I then implemented the systems in the test baffle and measured them to verify that the crossover performed as the modeling predicted. Finally I constructed the systems and completed the crossover design based on listening tests. I don't believe any amount of computer testing or design software can replace this critical listening and tuning part of the development. I modified the CSA's crossover during this listening period and will present both designs along with their merits.

ABOUT THE AUTHOR

G.R. Koonce is an electrical engineer who has enjoyed the hobby of designing and building audio equipment, test gear, and speaker systems. Professionally, he worked for over 30 years producing audio frequency equipment for military use. He is a member of AES and has been a contributing writer to *Speaker Builder* since 1981.

PHOTO 1: Completed CSA system.

PHOTO 2: Completed CSB system.

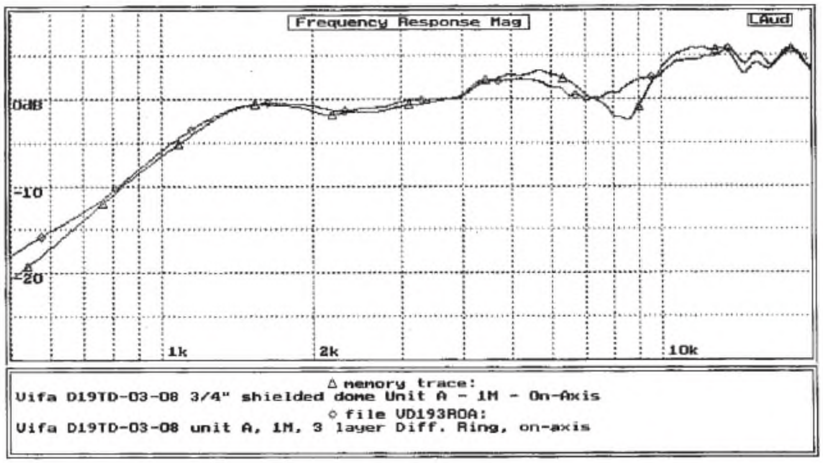

FIGURE 1: Vifa D19TD-03-08 on-axis response with (◆) and without (▲) diffraction ring.

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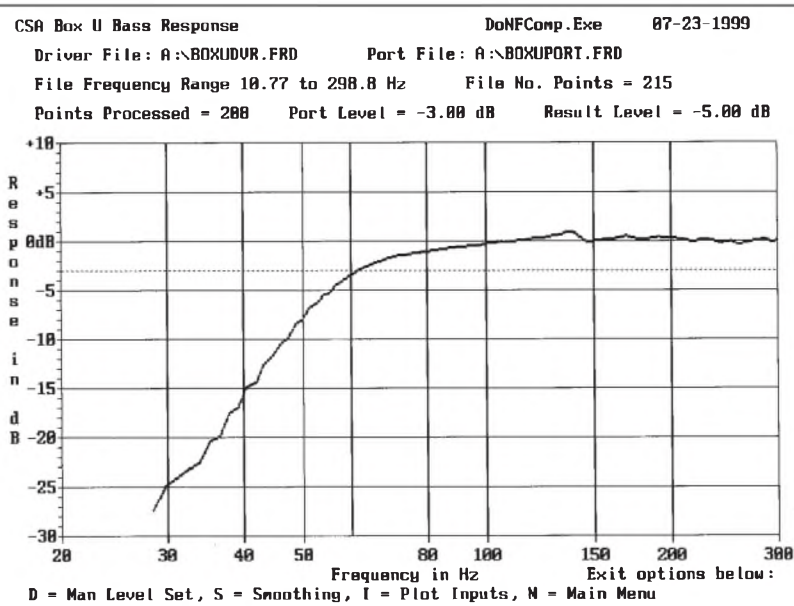

FIGURE 2: CSA bass response via near-field test.

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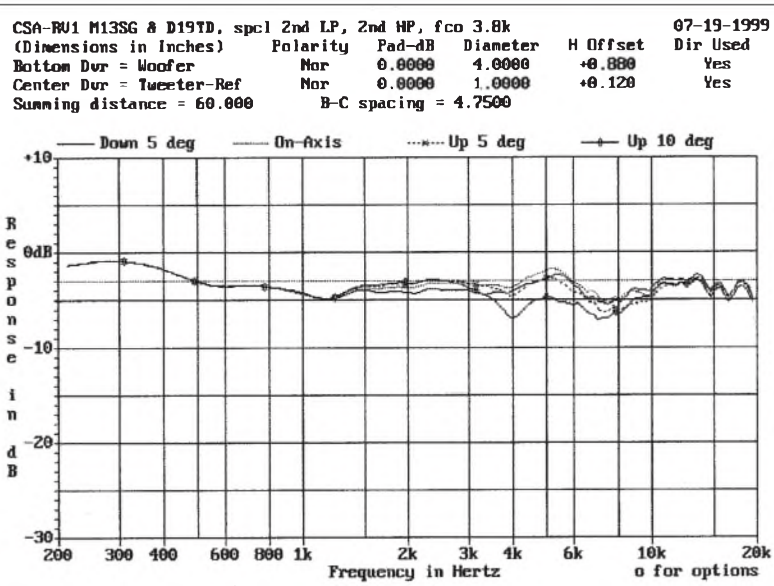

FIGURE 3: CSA-RV1 modeling results.

B-1762-3

DIFFRACTION RINGS

One requirement for all projects was that all drivers be surface mounted to remove any requirement for the builder's capability to rout them in flush. You mount the drivers by simply cutting the proper size hole in the box's front panel and securing the driver to it. When you surface-mount tweeters with a flat faceplate, the diffraction at the edge of the faceplate can cause severe on-axis response variations, adding up to a 6dB peak-to-peak response ripple. The response smooths as you move off-axis, but this ripple causes problems in setting the padding level and selecting the best crossover when modeling with the on-axis response.

A fix for the surface-mounted tweeter problem is to put a diffraction ring (DR) about the tweeter faceplate. You make this of multiple concentric rings of cardboard or similar material with diameters that increase as you move from the front of the tweeter faceplate to the front panel. *Figure 1* shows the measured on-axis response for the surface-mounted Vifa D19TD-03-08 tweeter used in CSA, with and without a diffraction ring, showing the major improvement this simple fix offers.

Some listeners have reported that their systems sounded smoother with the diffraction ring added, but my old ears were never sure they heard a difference. Surely with diffraction rings being so simple and cheap to make, it is crazy to omit them. I can't figure out why manufacturers don't ship their tweeters with snap-on plastic rings.

Sometimes the difference in acoustic phase center position between flush- and surface-mounting a tweeter makes surface mounting with a diffraction ring a better option. Building the diffraction ring for the CSA system is covered in the construction portion of this article, and you can find general development information on this topic in our book.

DEVELOPMENT OF THE CSA SYSTEM

The CSA system uses the Vifa M13SG-09-08 5″ (13cm) shielded woofer and the Vifa D19TD-03-08 ¾″ (1.9cm) shielded soft dome tweeter (it actually uses a canceling magnet) in a vented enclosure. The bass design indicated that a net box volume of about 0.27ft^3 (7.7 ltr) tuned to about 61Hz would produce an f_3 cutoff frequency of about 64Hz. I increased the size of the box to allow for the volume lost to the drivers, the crossover, the vent duct, and the bracing.

Figure 2 shows the measured response of one of the CSA boxes in near-field testing, indicating a final f_3 of about 63Hz. Modeling results for the CSA-RV1 system (*Fig. 3*) show that the optimum listening angle is just above on-axis with the tweeter, which uses a diffraction ring.

The front panel is tipped up 12° for floor-setting far-listening or desk-setting near-listening. The angle the bottom of the box makes with the horizontal forms the port diffuser and cannot be cut off in order to use these boxes on stands. The crossover schematic (*Fig. 4*) shows a strange-looking low pass for the woofer and a second-order high-pass with L-pad for the tweeter, wired with normal polarity. This low-pass network, which I call a "strange second-order," was developed by modeling to avoid the requirement of a Zobel, thus saving one large capacitor.

The woofer-to-tweeter crossover occurs at about 3.8kHz, just a bit below Vifa's minimum recommendation of 4kHz. The coil (L3) and parallel resistor (R2) in the high-pass serve to correct the tweeter's rising high-frequency response, clearly shown in *Fig. 1*.

The baffle test results (*Fig. 5*) verify that the crossover design of the CSA-RV1 system behaves as the modeling predicts. Implemented in listening tests, I found these systems have a laid-back sound that might be ideal for computer usage. Experimentation showed that an increased presence for a more forward sound was achieved by changing R1 from 3 to 2Ω, and R2 from 6 to 4Ω, resulting in system CSA-RV2.

Modeling results showed that CSA-RV2 had a boost in the region of 2k–4kHz and a slightly rising tweeter response above 10kHz in comparison to the CSA-RV1 response. System CSA-RV2 does not appear as flat, but does sound good if you prefer a high-presence system. A removable back panel permits listening to the system each way before you decide.

DEVELOPMENT OF THE CSB SYSTEM

The CSB system uses the Eminence 615F2025 6½″ (16.5cm) shielded woofer and the Audax AW025M1 1″ (2.5cm) shielded soft dome tweeter. Remember that this woofer is available only from

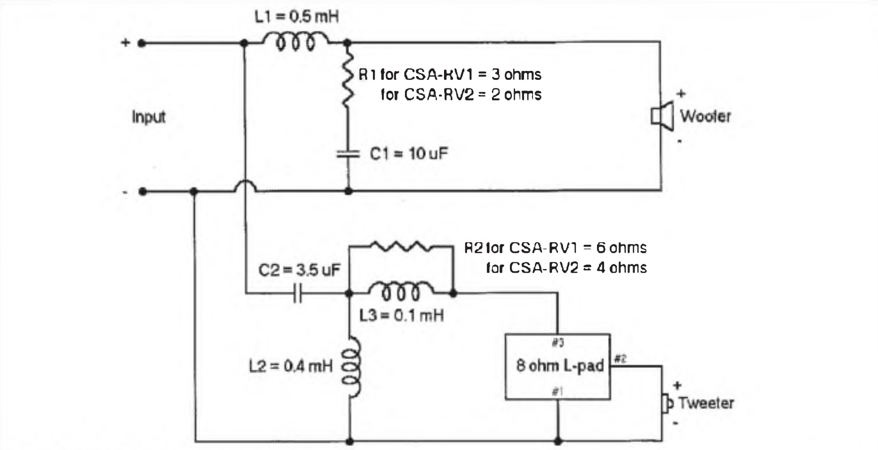

FIGURE 4: CSA crossover schematic.

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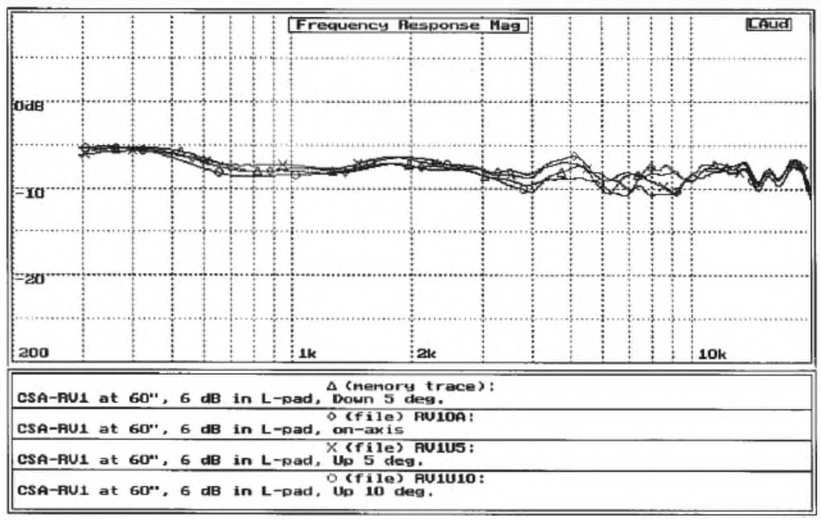

FIGURE 5: CSA-RV1 baffle test results.

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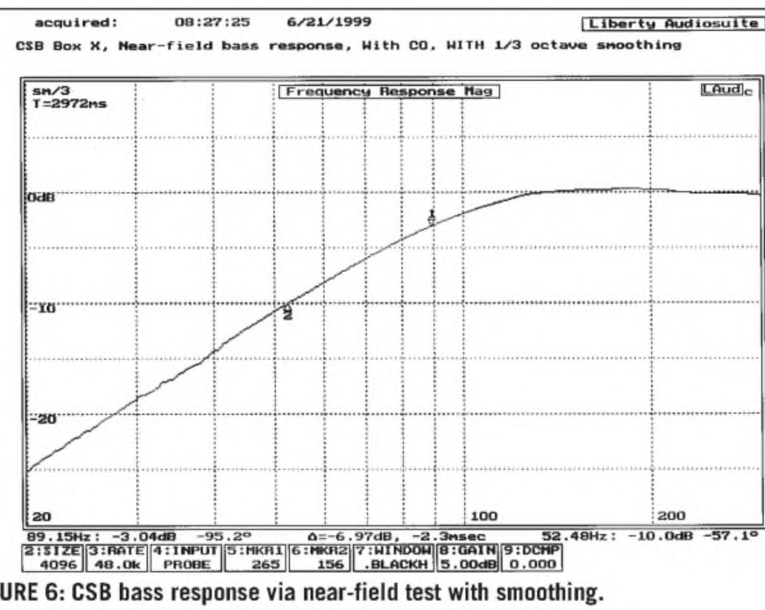

FIGURE 6: CSB bass response via near-field test with smoothing.

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Martin Sound Products; it is shown in their 1999-2000 catalog as Martin #2736. I obtained the minimum box size by using a closed-box design, which, however, makes construction tedious, since the large woofer nearly fills the box.

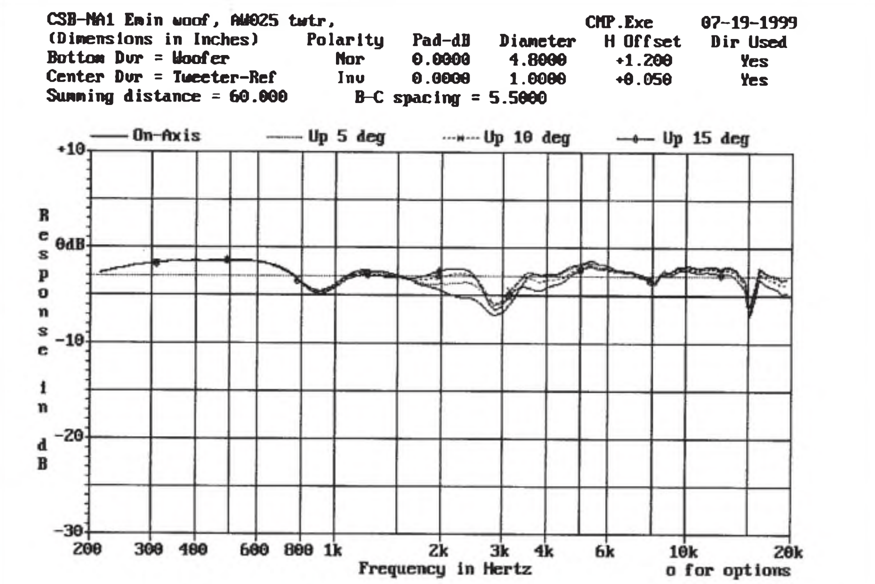

FIGURE 7: CSB-NA1 modeling results.

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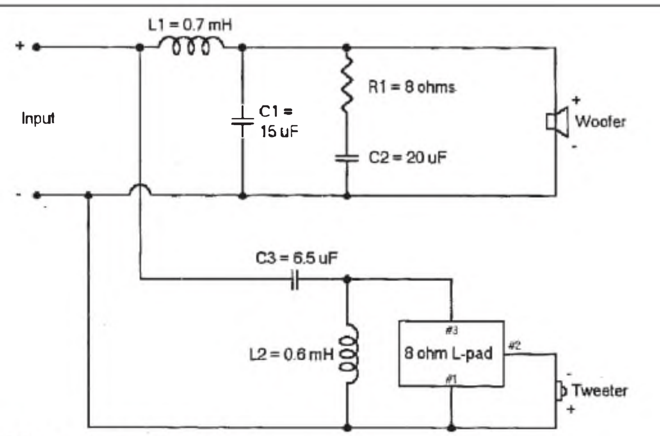

FIGURE 8: CSB crossover schematic.

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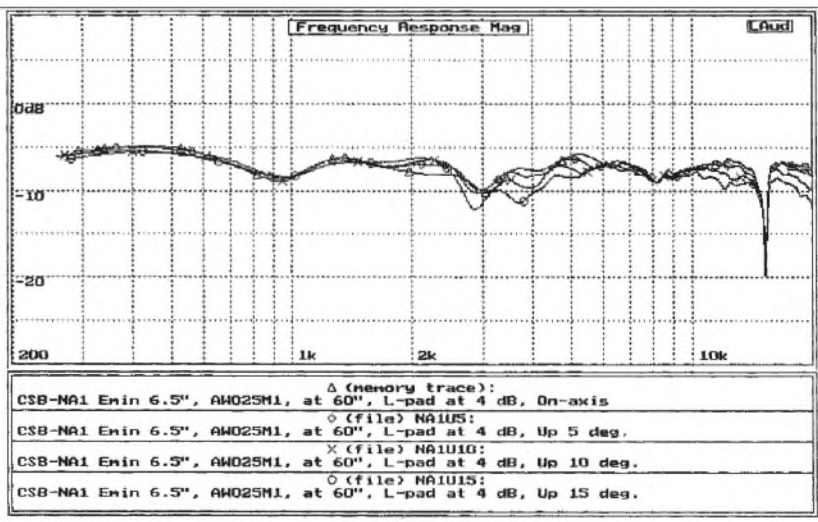

FIGURE 9: CSB-NA1 baffle test results.

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The bass design showed that a net closed-box volume of 0.175ft^3 (4.95 ltr)—assuming lined walls but no stuffing—would yield an f_3 cutoff of about 90Hz. I increased the box size to allow for the drivers, the internal crossover, and bracing, resulting in the near-field bass response of a complete system (*Fig. 6*), with f_3 at about 90Hz.

You could extend the bass to f_3 at about 62Hz on this system by using a vented box of about 0.25ft^3 (7.09 ltr) net, tuned to about 53Hz. This would mean increasing the gross box volume by about 0.075ft^3 (2.13 ltr), plus the loss to the port duct and any additional bracing. Note that the vented-box version (not tried) would result in some decrease in ability to play loudly. Be sure not to change the driver spacing on the front panel, or you may need a new crossover design.

Figure 7 shows the CSB-NA1 modeling results for this system indicating that the optimum listening range is from on-axis to up angles relative to the tweeter location. The design tips the front panel an additional 12°, putting the listener on a good axis for distant floor-standing listening or near desktop listening. To use these speakers on stands, placing the tweeter below ear level, cut the side boards flush with the bottom board. The tweeter used with CSB has a tapered faceplate and does not require a diffraction ring.

The crossover schematic (*Fig. 8*) indicates a second-order lowpass with Zobel for the woofer, and a second-order highpass with L-pad for the tweeter, which is wired in inverted polarity. The crossover from woofer to tweeter occurs at about 2.8kHz. The baffle test results (*Fig. 9*) verify that the crossover design works as the modeling predicts. No change was made in the crossover circuit during listening tests.

BASIC CONSTRUCTION

Both CSA and CSB use the same construction technique, a design that has a maximum tolerance for errors in cutting the pieces, but neither of these projects is simple to build. With small enclosures, tolerances are tight in placing items inside the box, and care is needed to prevent physical interference problems.

The boxes are designed so that the grille frame is part of the enclosure, and the grille cloth staples right to the front edge of the boxes. Although my approach was to cover the boxes with stick-on vinyl, you can apply a finish to these boxes in various ways. After they are properly prepared, apply the vinyl and staple the grille cloth over it. Finally, cut 5/8″ (1.59cm) half-round strips to fit the grille opening, cover them with the same vinyl, and attach them around the front edge with small brads. This simple approach achieves a good-looking box with the color and style of the vinyl you select (*Photos 1* and *2*).

The construction of these two projects is so similar that if you plan to build either one, I recommend you read through the sections and look at the photos for both projects. A reasonable general construction agenda for either one is as follows:

1. Cut out all the pieces. By cutting all pieces having a common dimension with the same saw setup, the box will fix together even if a dimension is in error. The top is the only major piece whose dimensions depend on the actual thickness of your particleboard, so you should cut it extra wide for later fitting. It is also best to make the front fill board of CSB extra tall for later fitting.

2. Cut the needed driver holes in the front panel and the hole for the port in the bottom board for CSA. File a radius on the back of the woofer hole—only in the areas between the driver frame struts—to permit these small woofers to "breathe" properly. Drill the holes for mounting the drivers and for the front-to-back (F-B) dowels. The tweeter holes require notches for the wire terminals.

3. Mark the location of the front panel on both sides of the bottom board. Measuring from the underside of the board, mark the positions of the nail holes in the front panel, taking care that the 1 5/8″ (4.13cm) nails do not reach the woofer hole. Experience has shown that you should avoid 1 1/8″ (2.86cm) from the end of a board to prevent splitting the particleboard. Drill nail holes through the bottom board with a #50 (0.07″) drill to ensure the nails go in straight.

Using glue, attach the front panel to the bottom board, being careful to face the "front" of the panel correctly. Secure the front panel flush with both sides of the bottom board and set back the proper distance from its front. Verify that the panel is standing at right angles to the bottom board. Make sure the front panel is not "rotated" on the bottom board by seeing that it is at right angles to the sides of the bottom board. Proper assembly of the front panel and bottom board will ensure simple assembly of the remainder of the box. Allow the glue to dry on this assembly.

4. Using the front-panel/bottom-board assembly as a template, lay out and mark the locations of the assembly on both the inner and outer surfaces of the side boards. From the outside, mark the positions of the nail holes again making sure the nails will not penetrate the woofer or tweeter holes. Note that CSB has a front fill board, the location and nail positions of which should be marked on the sides and bottom board. Drill holes for these nails from the outside face, with the exception of those for the front fill board on CSB, which should be drilled from the top side of the bottom board.

5. Glue one side board and clamp it to the front-panel/bottom-board assembly so that the front panel is flush at the top, and the bottom board is flush at the front. Make sure the front panel maintains a constant setback from the front edge, and nail it onto the side. Glue and clamp the second side at the top and bottom of the front panel, making the same points flush, as with the first side.

Next, stand the box up and see that it sits solidly on a flat surface. If it rocks, remove the top clamp and rotate the second side until the box sits properly. Reclamp the top, verify the two flush points, and attach the second side. Set all your nails driven into the sides below flush to allow sanding. Allow the glue to dry on these assemblies.

6. Next fit the back so that it sits in place while you fit the top. Cut the top carefully to fit the box. If you have a router, the best approach is to install the top with about 1/32″ (0.08cm) excess on each side, and trim it flush with a 1″ (2.54cm)-long flush cutting bit. Mark the top, drill nail holes in safe locations, and then glue and nail the top so it is flush with the front edge of the sides. Set all top nails below flush. For CSB, cut the front fill board to fit properly and install it. Allow the glue to dry.

7. Install the side-to-side dowel, being careful to get it positioned correctly, since clearances are tight. Set the dowel and mounting blocks in position and make sure they do not interfere with positioning the nominal 1″ × 1″ wood strips (actually ¾″ by ¾″ [1.91cm]) that mount the removable back. Now glue and install the side-to-side dowel. Driving nails inside these narrow boxes is difficult, so you may prefer to use brass screws (as I used for CSA) or simply glue and clamp. Again, positioning is critical.

8. Install the 1″ × 1″ strips all around the back panel at the proper depth so that the back with its foam-tape gasket will end up flush with the back of the box. Make the vertical strips full height, and cut the horizontal strips to fit between the vertical pieces. Install the strips with aluminum nails or brass screws because of their proximity to the crossover parts. Be sure the nails or screws will not interfere with the positions for the screws used to retain the back.

Glue in the horizontal 1″ × 1″ strip to terminate the front-to-back dowels so it is flush with the other strips. Along with holes for the front-to-back dowels, file notches in both sides of this horizontal strip to accommodate a #18 zip cord for crossover wiring. Make these notches with their outer ends set in about 1″ (2.54cm) from the inside edge of the box and facing the back panel. Glue in the ½″ (1.27cm) dowels, making sure they are below flush at each end. Now fillet all seams to be sure the box will be airtight.

9. Fill the nail holes set below the surfaces and sand the boxes as desired. Depending on your finishing technique, you may do other work at this point. If you plan to use stick-on vinyl, be sure to vacuum the surface of the boxes after sanding so that the vinyl will stick properly.

10. When all seams are properly sealed and all the glue has dried, you should set the back panel into position, first marking the positions of the 1″ × 1″ strips from the inside to show where the crossover parts may be located. Next flip the back over and mark the strip positions on the outside to show where you will drill the holes for mounting the back. Use flat-head screws and countersink them flush with the back.

Install the fiberglass inside the box and place the foam tape on all the 1″ × 1″ strips, but not over the notches for the crossover wires. Cut a hole in the foam tape around each back screw to keep the screw from tearing up the tape.

11. Construct the crossover on the inside of the back panel and wire and install the drivers. Install the woofer with the wire terminals at the top. Install the back panel, slowly on a closed box to avoid woofer damage. Complete the front panel with a diffraction ring on CSA and front-panel damping material on both box types. The systems are now ready for testing. Drape the grille cloth over the front to allow setting the proper tweeter level for your application. A decision on the final crossover for CSA should be made now.

12. Finally, cover the boxes with stick-on vinyl, wrapping it over the front edge and for a couple of inches onto the back. Also cover the front fill strip on CSB and the port diffuser area of CSA. Carry the vinyl around the bottom so that the box sits on it and keeps it from coming loose. Staple the grille cloth to the front of the box to keep it tight. Next, miter-joint 1/8″ (1.59cm) half-round to form a frame about the grille area. Wrap these strips with the same stick-on vinyl, folding it over the ends of the strips. Install these strips with small finishing brads, being careful to avoid the grille-cloth staples. This completes the construction. Next month we'll complete this project by assembling the units. ❖

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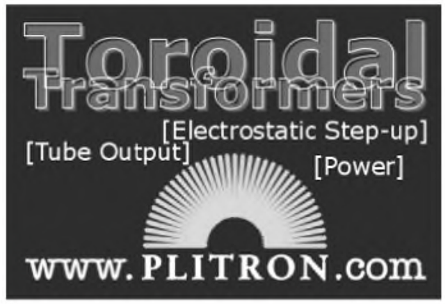

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The Cure for the Common Dynaco MkIII

What can you do for an ailing MkIII? This author prescribes the R_x for this venerable amp. **By Kara Chaffee**

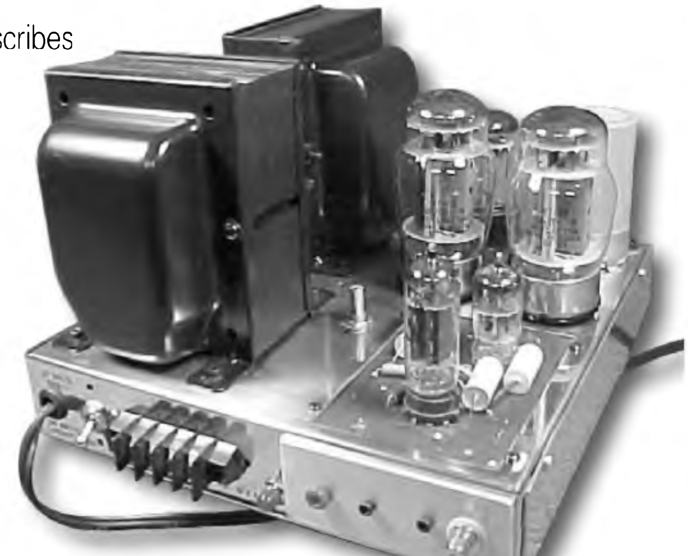


PHOTO 1: Sample Dyna MkIII.

A friend brought me a pair of Dynaco MkIIIs, taken apart, in a box. The person who had given them to him figured out how to take them apart, but never quite figured out how to put them back together again. My friend wished to rebuild them and asked me what could be done to make them special. I thought for a minute, handed him the 6AN8 (which was rolling around in the bottom of the box with the mouse turds and dust bunnies), and said, "Put this on the (nearby) railroad track two minutes before the Freight Special arrives, then we'll talk about a proper driver for this amplifier."

THE HISTORY

The MkIII was one of the most successful power amplifiers ever sold (*Fig. 1*). It's estimated that several hundred thousand were built by the time the model was discontinued in 1977! Although I am no fan of the 6AN8, Dyna's choice of this driver made sense, when you consider the fierce price competition of that era's marketplace and the need for a kit that was simple to build. From today's perspective, however, given the excellent quality of the Dynaco transformers, the minimalist drive circuit could use some improvement.

If you follow the vintage market, you know that the MkIII does not command the prices of the Marantz, McIntosh, or Eico monoblocks. The relative abundance of MkIIIs is a factor, and I also believe the drive circuit has kept the model's price down. Actually, this is a good thing.

With a little Yankee ingenuity, you can fix the circuit and take advantage of those excellent transformers at a discount. As an inexpensive basis for a modification project, the MkIII is stellar. The result is a superb amplifier with honest-to-goodness tube rectification (*Photo 1*), which you don't get from modern tube equipment manufacturers

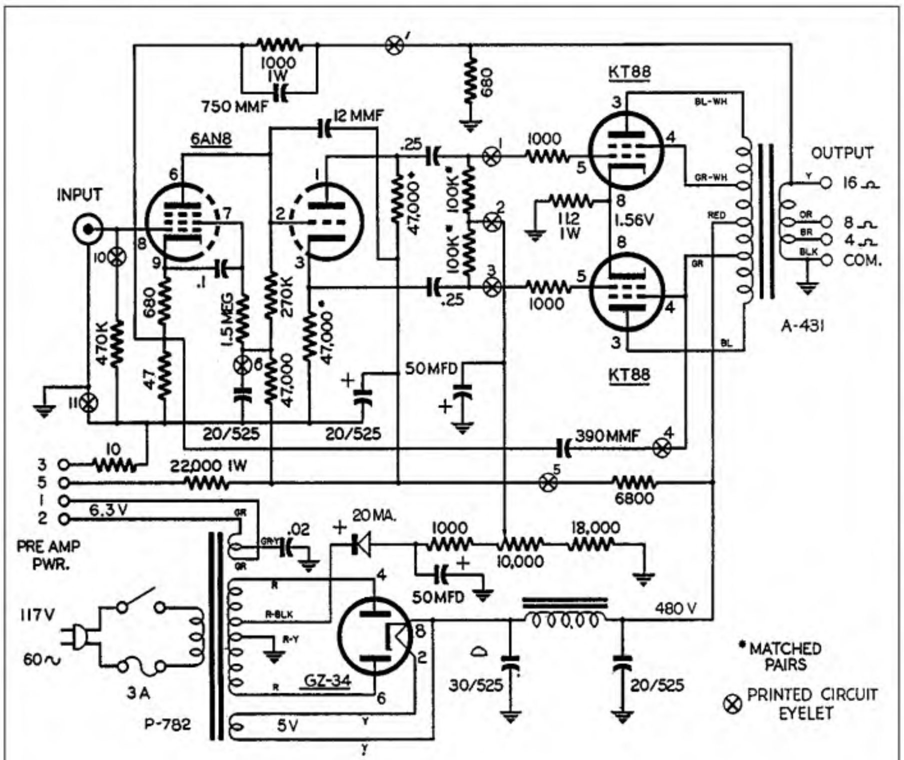


FIGURE 1: Original Dynaco MkIII schematic.

ABOUT THE AUTHOR

Kara Chaffee earned a Bachelor of Science degree from the University of California and subsequently was a manufacturing engineer at a sintered metals plant. Kara briefly owned a small company manufacturing valve train parts for 12-cylinder Ferrari engines, and for the past several years, has been doing the electrical and mechanical engineering for the deHavilland Electric Amplifier Co. (1701 Santa Rosa Ave., Santa Rosa, CA 95404; (707)527-5000, FAX (707)527-5009).

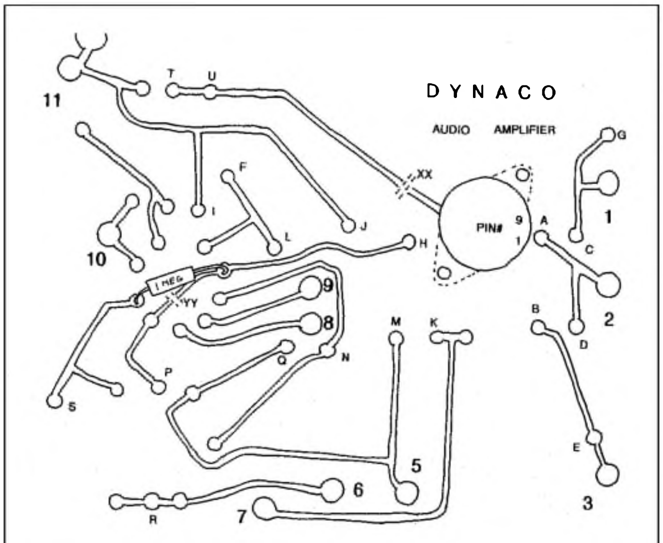


FIGURE 2: Circuit board layout.

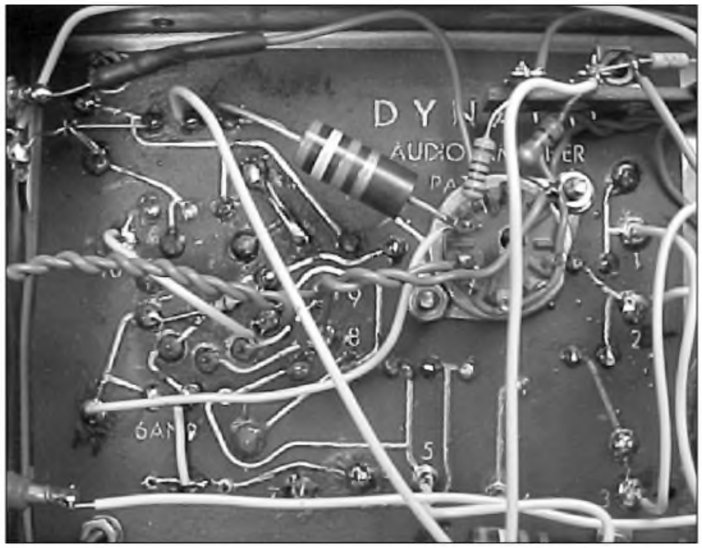


PHOTO 2: Circuit modification.

unless you spend big bucks. Just ask yourself why the top models of an amplifier line have tube rectification!

THE CURE

The solution, or cure, is to build a new drive circuit. By using the existing traces in a new way, and adding a 9-pin socket on the board, you can build an excellent two-tube drive circuit. The modification costs about $20 and consists of parts that any old radio codger would probably give you for free.

First of all, the new circuit is all triode. One half of a 12AT7 amplifies the signal coming in. This tube is directly coupled to a 6CG7/6FQ7 tube, configured as a Schmitt-type phase splitter. Another name for the Schmitt circuit is "long-tailed pair." (My cousin Zeke caught wind of the long-tailed pair name and would sing with his banjo, "Oh I singed my hair on a long-tailed pair, I singed my hair on a long-tailed pair!")

This circuit is similar to many classic Eico and Marantz designs. A typical Schmitt phase splitter can be seen in the Eico HF-60 amplifier. One half of the 12AT7 is not used. I chose not to build a Williamson-style driver here because the Schmitt type uses fewer components and is easily adaptable to the existing board.

The modification actually ends up with fewer components on the board than the original Dynaco! I do not recommend paralleling the two triode units of the 12AT7. My general design practice is to send the signal through as simple a circuit as possible.

THE TUBES

Many tubes could have been used for this modification. I chose the 6CG7/6FQ7 because it is a fine driver tube. It is fairly low mu and quite linear. Many designers, including myself, regard it as one of the best 9-pin miniature triodes.

I have observed that tubes usually can be classified as Yin tubes and Yang tubes. Don't laugh. This is actually useful. Beyond textbook engineering there is the issue of the sonic character of a particular tube.

"Yin" qualities are sweetness, detail, and transparency. "Yang" tubes are strong sounding, forward in tone, and perhaps more opaque. The 12AT7 is definitely Yin. It has sweetness and transparency that is a perfect complement to the 6CG7/6FQ7, for just the right balance of Yin and Yang.

Someone will invariably ask about using a 12AU7 instead of a 12AT7. Please feel free to drop a 12AU7 right into this circuit. Although the circuit is not optimized for 12AU7, the resulting operating point is not too far off, and you will get a fair taste of the 12AU7 sound. But remember, only when the Yin and Yang of the electricity is properly balanced will the Gates of Crystal Happiness open!

ELBOW GREASE

Here are some guidelines for the circuit board modification and installation:

1. Before unsoldering leads going to the board, mark each wire with the original Dynaco eyelet number.

2. Unsolder the leads and remove the board.

3. Remove everything except the existing tube socket. Make sure you heat the solder joints well, until components move freely, before pulling. The old Dynaco traces on the board lift or crack very easily. Take care unsoldering the 390pF mica capacitor (the one that looks like a domino). You will use it later. I recommend replacing even the

TABLE 1
CIRCUIT BOARD COMPONENTS AND CONNECTIONS

100k ½W	A–B
100k ½W	C–D
220nF/600V	eyelet #6-E
220nF/600V	F–G
100nF/600V	H–I
1M ½W	solder directly between PC socket pins 2, 7
wire jumper	J–K
43k 2W	L–M
18k 3W	N-eyelet #7
39k 2W	Q–P
wire jumper	P–R (on trace side)
wire jumper	PC socket pin 3 to eyelet #10 (solder small loop directly around PC pin)
eyelet #1	signal out to pin 6, KT88 (original wire marked #3)
eyelet #2	B-from bias pot
eyelet #3	signal out to pin 6, KT88 (original wire marked #1)
eyelet #5	390V DC connection to board
eyelet #11	ground
pad U	top of 39k 2W plate resistor, 12AT7
pad T	245V DC connection to board
pad S	signal input from pin 6, 12AT7
RCA jack	center terminal RCA jack to pin #7, 12AT7

Note (1) Original Dynaco eyelets referred to as "eyelet #3" etc.
Note (2) Pads lettered as shown on *Fig. 2.*

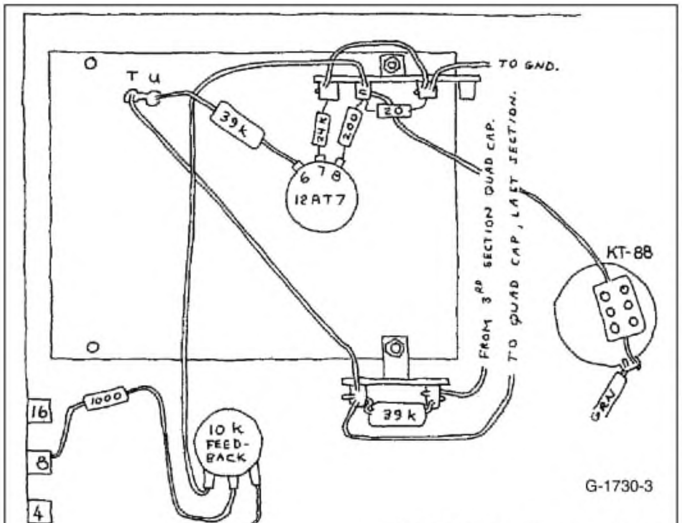


FIGURE 3: Feedback pot and 12AT7 socket details.

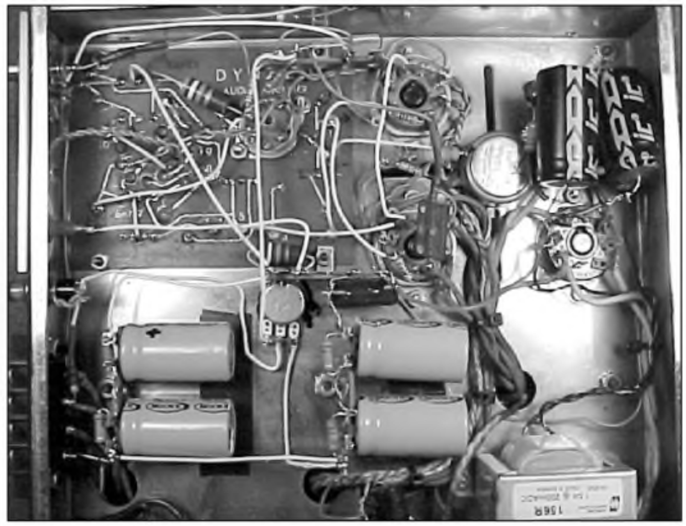


PHOTO 3: Component placement.

two 100kΩ grid resistors, R14 and R15, with 1% metal films.

4. Now here is the tricky part. The board is old brittle phenolic, and you are going to drill a hole in it. Select a 9-pin socket that has its mounting flanges on the top. The object here is to cut as small a hole as possible and to have only the plastic part of the socket go through the board. Refer to *Fig. 2* and *Photo 2* for the approximate location of the hole.

Note that the socket hole must not disturb the pads marked "H," "A," and "B." The trace that runs under the socket location is cut at "XX" and is of no consequence. Be aware of where the mounting screws come through also.

I used a Unibit stepped drill to cut the holes in my boards, but you may consider drilling progressively larger holes with a series of drills. The material actually drills decently, but back the board up on a piece of wood and be careful of the drill grabbing. I have indicated the orientation of pins #1 and #9 (*Fig. 2*), but if you have a socket that needs to be oriented differently, it can be made to work.

5. Mount the components and the new tube socket (*Fig. 2*). Components go on the topside of the board unless otherwise indicated. Put the caps on last so you don't stab them with hot leads as other components are added to the board.

Trim the lead on the 43k resistor so that it goes through pad "P" a little long (¼"). A jumper under the board will need to be soldered to the extra lead length. *Table 1* is a list of board connections, including which pads go with which components and leads.

6. Now that you have assembled the new board it is ready to install. Note, we have not mounted anything to the new socket or dealt with the new feedback circuit.

7. Reinstall the circuit board in the same orientation as the original.

8. Mount a three- or four-terminal tag strip on the same screw that secures the corner of the board where it says "Dynaco." Mount the components for the 12AT7 (*Fig. 3* and *Photo 3*). I placed the 11.2Ω cathode resistor between the rightmost terminal and the grounded terminal because I prefer to tie as many ground points together as possible, but it can be left where it originally was.

9. Remove the original screen feedback wire that went between pin #4 of the lower KT88 and eyelet #4. Solder the 390pF mica capacitor to this same pin, #4, of the KT-88. The solid green screen tap lead from the output transformer goes to this pin also. Splice it to the other side of the 390pF mica and heat-shrink the joint (*Fig. 4*). Eliminate the 680Ω resistor between the 16Ω tap and the COM tap, and leave the COM tap grounded.

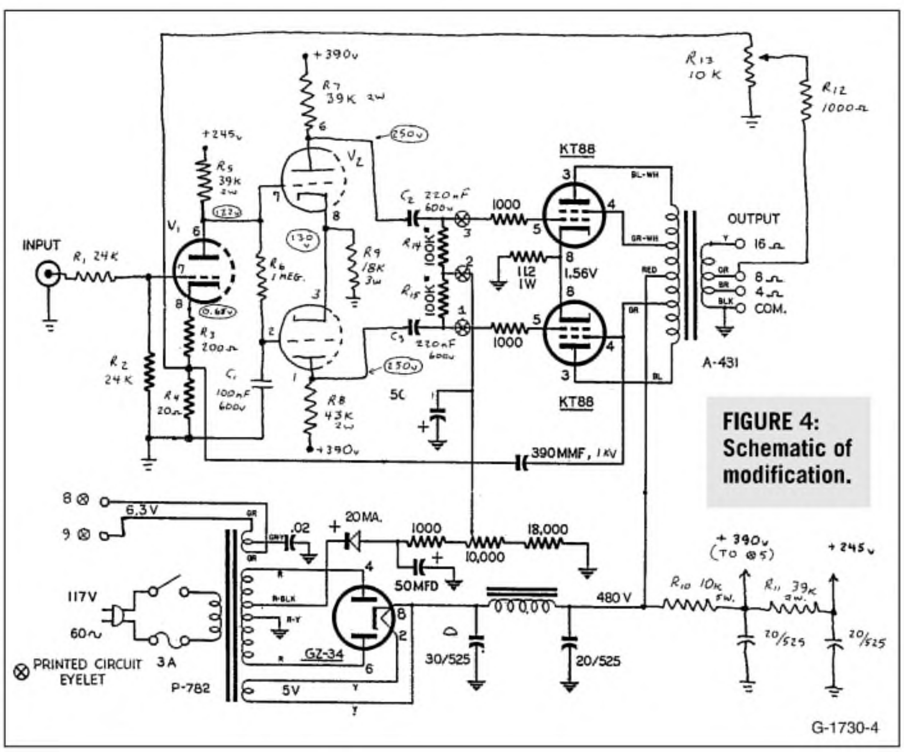


FIGURE 4: Schematic of modification.

TABLE 2
PARTS LIST

R1, R2	24k ½W	Metal film
R3	200Ω ½W	Metal film
R4	20Ω ½W	Metal film
R5, R7, R11	39kΩ 2W	
R6	1M ½W	Metal film
R8	43kΩ 2W	
R9	18k 3W	Metal-oxide film
R10	10kΩ 5W	Wirewound or metal oxide
R12	1000Ω ½W	
R13	10kΩ potentiometer, ½W	
R14, R15	100kΩ ½W	Metal film
C1	100nF/600V	Film capacitor
C2, C3	220nF/600V	Film capacitor
V1	12AT7	
V2	6CG7/6FQ7	

Supplier notes:
All resistors are available from Mouser Sales. Sockets, hardware, and NOS tubes are available from Antique Electronic Supply Co.

10. Remove the 6k8 resistor from the quad cap and replace it with the 10k 5W.

11. Mount a two-terminal tag strip on the board mounting screw near eyelet #3. R11 (39k) is mounted here (*Fig. 3*). Refer to *Table 2* to make remaining wiring connections.

FEEDBACK KARA'S WAY

I prefer to use a variable feedback scheme in this amplifier. Refer to *Fig. 3* for the layout. The primary advantage of variable feedback is that it allows you to voice your amplifier with your particular speakers (and ears).

I use a quality vocal ensemble recording at a moderate level as a reference. As you become familiar with the feedback control, you will notice a critical point where the vocals cease to be "yelly," and yet retain a vivid quality. You will also notice a nice balance between the vocals and instruments. Too much feedback subdues the musical dynamics needlessly, and too little gives you an amplifier that barks.

Setting the feedback by ear is not as sloppy a technique as you might imagine. I have set the feedback by ear many times on this amplifier, and it is surprising how close the "by ear" setting is to the optimum setting for a 10kHz square wave when set on the oscilloscope. If you do not wish to use the variable feedback, eliminate the pot and the 1kΩ resistor. Run a 1.87kΩ resistor between the 8Ω speaker tap and the junction of the 200Ω/20Ω cathode resistor of the 12AT7.

ACKNOWLEDGMENTS
To Chester K. Farrow, my electronics teacher who had an infectious love of tubes and an inherent distrust of transistors, and Norman Koren who got me to cut holes in my Dyna 70 to let the evil spirits out, and let another tube socket in.

I retained the 390pF mica cap for two reasons. First, I already have one, and second, it functions well where my ear says to set the feedback pot. Feel free to use larger caps, up to 500pF, especially if you have electrostatic speakers. If you use a different cap than the original 390pF, get a 2kV rating. Remember, one side of that cap is on the hot side of the output transformer.

Normal operation of the feedback pot decreases the gain as you rotate the pot clockwise. If you turn clockwise on the feedback pot and the music is getting louder, and then the amp begins to howl, you probably forgot to cross the leads going to the KT-88 grids. The phasing of the modified circuit is opposite of the stock circuit. My chassis already had a hole in it where a pot could be mounted. If you don't wish to drill the chassis, you can mount a small pot on a tag strip inside and set the feedback with the bottom off.

INPUT SENSITIVITY

The new circuit has plenty of gain; R1 and R2 form a voltage divider (*Fig. 4*). *Table 2* shows the parts list. I did this so full power could be obtained with 3V P-P input. I shrink-wrapped R1 into the lead coming right off the RCA jack.

If the residual noise floor of your preamp is too high, consider a smaller value for R1 and a larger value for R2. These resistors also set the input impedance for the amplifier, and if you are using a preamp such as a Dynaco PAS-3, it would be wise to set the total of R1 plus R2 at 470kΩ, due to the high output impedance of the PAS-3. A good rule of thumb is to keep the load around ten times the output impedance of the source.

CAPACITOR MAGIC

Coupling caps do make a sonic difference. I used the little AXON film and foil caps in these amplifiers. They are sonically good and are inexpensive.

Hovlands, or the Rel-Cap PPFXs, are also nice. Although C2 and C3 are called out as 220nF, 100nF will work fine as well. If you purchase a cap such as a Hovland, the 100nF will fit on the board with a lot less struggle.

RESISTOR MAGIC

I like the cheap metal films, which are noticeably more transparent than carbon composition resistors, and if the amplifier has a basic good tone, you do not need to load the amp up with carbon to smooth it out. Experiment here and you will be rewarded.

I definitely do not sign on to the program that the only worthwhile resistors are old carbon comps that came from the Navy radio that was recovered from the B-17 bomber that was dredged up from the lake. The best amplifiers I know of use a mix of resistor types. A place where carbon is usually retained is the 1kΩ grid resistors on the KT-88s.

POWER SUPPLY

So far I have not provided much description of the power supply. In my friend's amplifier, the quad caps were gone, so I built capacitors out of pairs of 450V or 500V electrolytics in series. Be sure to run 270kΩ balancing resistors across each cap if you stack them, and remember the upper capacitor case is at one-half of the total voltage. The other reason to beef up the voltage rating of the power supply capacitors is that if you run any other rectifier besides a 5AR4 in the amplifier, the voltage across the power-supply caps can easily exceed 600V before the other tubes on the chassis warm up.

IS 200Ω ENOUGH?

The 200Ω cathode resistor for the 12AT7 may seem a little strange, but there is a reason for this. The object is to get some current flowing without having the standing voltage at the plate rise too high. This voltage is directly coupled into the 6CG7/6FQ7, and sending 100 to 120V along to the next stage nicely balances the voltages in the phase splitter.

(to page 89)

Digital Class-D Subwoofer Amp, Part 1

Here's a close-up look at the pros and cons of true digital amplifiers. **By Thomas O'Brien**

In the "old days" of audio, amplifiers were purely analog (tubes, then transistors) and inefficient, wasting power by generating heat. The wasted power was a concern, especially for portable and miniaturized audio systems, so Class-D was used to conserve power. Most Class-D amps use pulse-width modulation (PWM) to convert the incoming audio to a pulse train, although there are several variations of how this is done.

Class-D is the same technique used in classic motor drivers, but modern high-speed electronics are required to run at appropriate frequencies for accurately driving a loudspeaker. There are benefits aside from efficiency when using the Class-D approach. Elimination or reduction of some types of distortion is possible, and good transient response leads to a reputation for accurate sound.

In a Class-D amplifier, a modulator converts the input signal to a pulse stream. The modulator output is a low-power signal. High-power circuits (called the "output stage") amplify this signal, and the high-power output drives a speaker through a passive filter. The filter removes high-frequency noise from the signal driving the speaker.

ABOUT THE AUTHOR

Thomas J. O'Brien attained a BSEE at Drexel University in 1992, and is currently pursuing an MSEE at Villanova University. He co-founded Sycom Technologies in 1993, producing personal digital voice recorders. He joined Quadrant International in 1997, designing DVD-player and Internet-TV hardware, and began working for InterDigital, Inc. in 2000, designing ASICS for cell phones. Thomas has pursued audio electronics on his own for 15 years, and has ten years of experience designing analog and digital Class-D audio power amplifiers. He also holds a patent, granted in August 2000, in the area of digital audio amplification. For more information about digital amplifier companies, check out www.digitalamp.com.

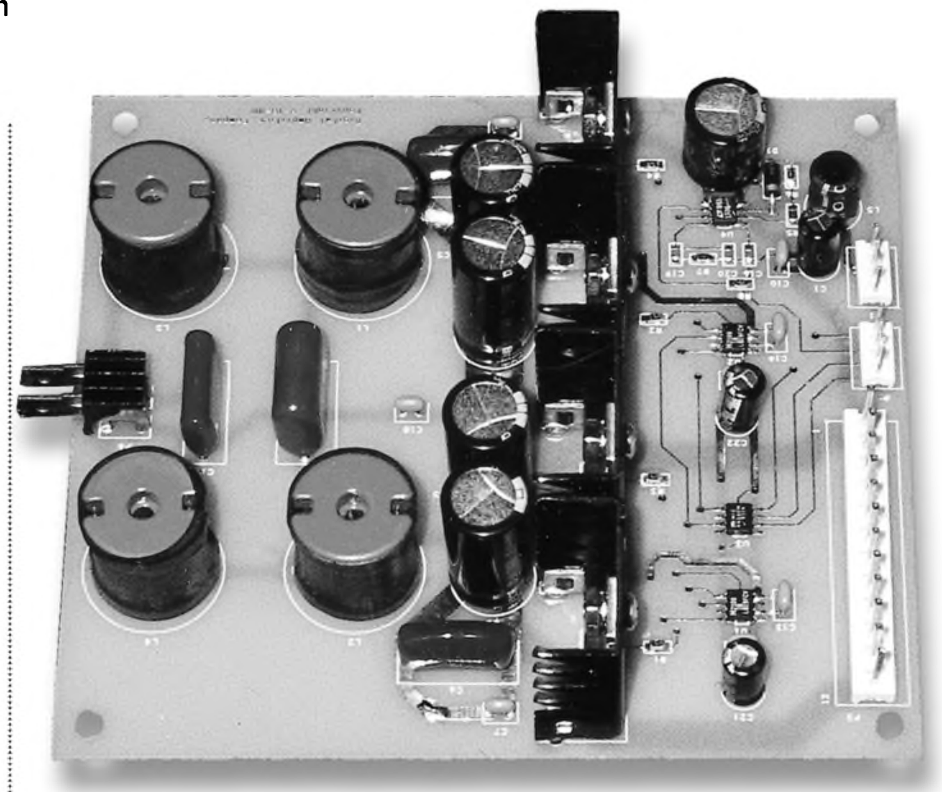


PHOTO 1: Modulator board.

DIGITAL CLASS-D

Although it may seem reasonable, the "D" in Class-D doesn't stand for "digital"; it's merely a configuration designator, as in Class-A, Class-B, Class-AB, and so on. Class-D amplifiers came to be known as "digital amplifiers" because they output a two-state signal (although some Class-D amps use more states), as do digital logic circuits. The output of a Class-D amp drives the load through a filter, removing the high frequency "carrier." However, most Class-D amps deal only with analog signals, and the output is not discrete-time (as in a computer); these amps are called analog Class-D amps.

It is difficult to achieve high performance with analog Class-D because analog modulators are inherently noisy. At least one level of analog feedback is usually employed to lower distortion to reasonable levels. Amplifiers requiring large amounts of feedback are sometimes viewed as poor designs, and are susceptible to instability.

The term "digital amplifier" is also used as a marketing tool. "Digital" has come to represent high quality in mass-market audio. For example, the word "digital" has even been applied to headphones. This can be confusing to the typical consumer.

Digital Class-D amplifiers, often called "true digital amplifiers," convert digital audio data (pulse-code modulation (PCM), usually from CD or DVD) to

PWM with logic circuits; this process is discrete-time. Converting the digital signal to analog before amplification is not necessary. The process of converting the signal from analog to digital or vice-versa adds artifacts to the audio. Also, digital Class-D amps do not require analog feedback to reach high-performance levels, as do typical analog Class-D amps.

On the qualitative side, digital Class-D is the most direct connection from a digital audio source to a loudspeaker. This directness provides very transparent sound, and some believe it to be the best way to hear a CD. Folks who swear by tube amps would rather hear the recording on LP instead, anyway.

Digital Class-D amps can achieve outstanding performance levels, including low distortion and high dynamic range. However, in some applications, performance is slightly compromised to lower the price of this technology by reducing signal processing in the modulator or by using cheaper FETs, FET drivers, inductors, and capacitors. The output stage transition time must be very fast and the output impedance must be very low in order to minimize distortion. The modulator must produce very accurate PWM with low noise, as well.

It is difficult, but not impossible, to build a digital Class-D amp with high performance and reasonable cost. The cost of digital Class-D is slightly higher than analog Class-D. Digital Class-D requires a better output stage because analog Class-D can correct for output stage problems with feedback, and digital Class-D doesn't have feedback.

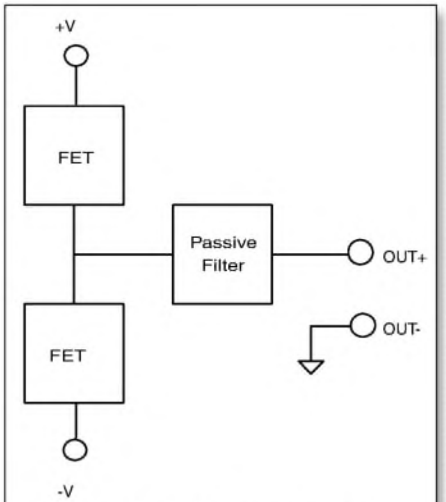


FIGURE 1: Half-bridge configuration.

ADVANTAGES

Digital Class-D amps do not need "calibration" or adjustment for DC offset because the offset can be digitally fixed near zero. In fact, digital Class-D amps are very repeatable in production because their behavior is almost completely defined in the digital domain, and it doesn't vary as much over age, temperature, and component variations as with analog Class-D amplifiers. Also, digital modulators don't rely on tight component tolerances, super-low noise components, and precise PCB layout, as do analog modulators.

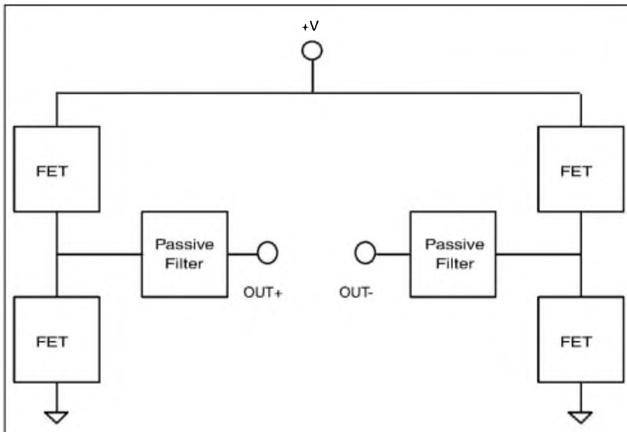


FIGURE 2: Full-bridge configuration.

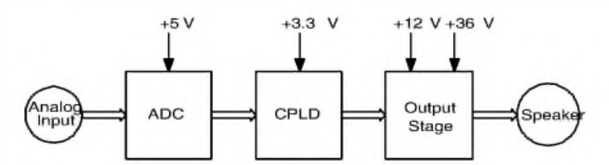


FIGURE 3: SA1 system configuration and required voltages.

A crystal oscillator digitally clocks a Class-D amp. Crystal oscillators provide the timing for accurate pulse widths and dead-time delays (see section on dead-time delays later in this article). In an analog Class-D amp, the pulse widths and dead times are determined by analog circuits and are more prone to noise than is a digital Class-D amp.

Digital Class-D also benefits from operating in the digital domain because circuits like equalizers, filters, and effects are often implemented digitally with no need to convert to analog.

Amplification does not add a level of complication to systems running DSP on the signal, and sometimes can be performed in the amplifier itself. A simple example is volume control. Many "integrated amps" have simple selection switches and a volume control. In this case, tone controls or equalizers are performed outside the amplifier. Selection switches are loss-less in the digital domain, and volume controls simply scale the audio data and are easily built into a digital Class-D amp.

In addition to eliminating the need for feedback, digital Class-D enjoys the same benefits as analog Class-D. There is no zero-crossing distortion. Efficiency is high, so there's little wasted heat; therefore, heatsinks, power supplies, and drive electronics can be very small without loss of quality or output power.

OVERCOMING DISCRETE-TIME LIMITATIONS

In a continuous PWM system (analog Class-D), the average output voltage is continuously variable because the pulse width is also continuously variable. In a digital Class-D system, the PWM is a clocked output, meaning that the pulse width must be a multiple of clock cycles.

For the pulse width to have 16-bit accuracy, you would need 65536 available pulse widths. At a pulse repetition rate (switching frequency) of 200kHz, this would require 76ps clock cycles (a 13GHz clock!). Because this is impractical (at least today), you must use a slower clock.

Typical Class-D PWM switching frequencies are between 200kHz and 500kHz. Digital PWM clocks range from 10MHz to 100MHz. For example, with a 50MHz clock and 200kHz PWM, you get 250 available pulse widths. This is less than 8-bit resolution!

The trick here is to exploit the fact that the switching frequency is much higher than the intended frequency

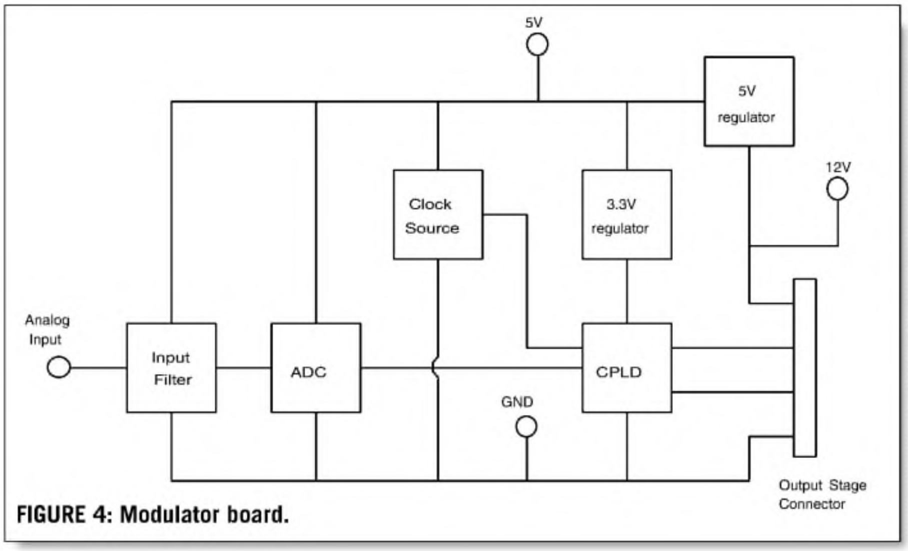


FIGURE 4: Modulator board.

you are reproducing. By selecting several different pulse widths in a repeating pattern, you can produce average pulse width values between the actual ones. For example, if the pulse width had 100 allowed positions (from 0% to 100%), and an average of 50.5% pulse width is desired, the output pulse width would alternate between 50% and 51%. The algorithms that produce the pulse widths, effectively increasing the resolution, vary in complexity and performance.

RFI CONCERNS

Like any switching amplifier, Class-D generates high voltage and current skews during output state transitions. This in turn produces RFI (radio-frequency-interference) that must be contained. In order to keep the Class-D amp from radiating this interference, it is important to enclose the output stage in a shield.

Typically, this shield is the amplifier enclosure, which contains not only the output stage(s), but the entire amplifier system. Non-metallic enclosures (for example, speaker cabinets) require the output stage to be in a shielded "box" with ventilation to allow heat to escape. The simplest type of enclosure for a Class-D amp is an aluminum (or other metal) box with evenly spaced vent holes.

Typical hole spacing is ½", and typical diameter is ⅜". The enclosure needs vent holes to provide some convection airflow while containing emissions. Good PCB layout is critical, not only to minimize transmitted RF, but also to ensure well-controlled power output and low output impedance.

Analog audio is still prominent. In some digital audio systems, there is no access to the digital data, and "legacy" audio equipment outputs only analog audio. Existing analog equipment can drive a digital Class-D amp if an analog-to-digital converter (ADC) is used to derive digital data from the input.

BRIDGED OUTPUT

By "bridging" the output of an amplifier, you can obtain nearly twice the voltage across the load, resulting in roughly four times the output power. A dual supply (positive and negative voltages) is not necessary to run in this configuration. Typical Class-AB amps are half-bridge, dual supply type.

Another consideration with Class-D amps that makes bridging popular is that non-bridged output stages suffer from the power-supply "pump" effect. During output state transitions, the output filter recoils from being driven at a high voltage when the drive is released (FET is turned off). After you turn the power supply switch (FET) off, flyback current from the inductor pumps the capacitor of the opposite side through the body-diodes of the FETs. This moves the power supplies up and down, distorting the output.

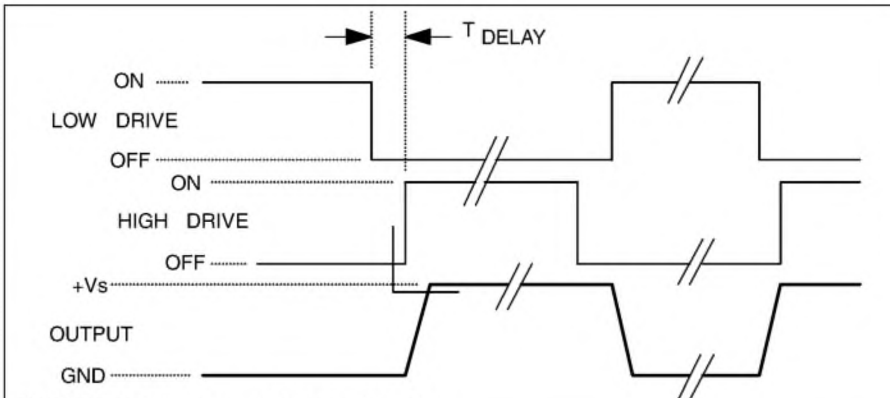


FIGURE 5: Delays to prevent shoot-through.

In a bridged configuration (*Fig. 1*), the current pump from the output filter can be applied to only one supply reservoir (capacitor), so the pump effect doesn't occur. The full-bridge configuration (*Fig. 2*) has two sides, the A-side and the B-side. The A-side switching output inverts with respect to the B-side switching output.

In a Class-D amp, the power-supply voltage determines the gain of the amplifier. It is possible to vary the power-supply voltage to control "volume." This works only if the system is open-loop (no feedback) because feedback corrects most of the gain difference.

SA1 SUBWOOFER AMP

The SA1 amp consists of a modulator board and an output stage board. The separation of these two sections allows future upgrades, including full-range operation or more power, without replacing the entire system. For example, a stereo system would consist of a stereo modulator board and two output stage boards. Separation of the modula-

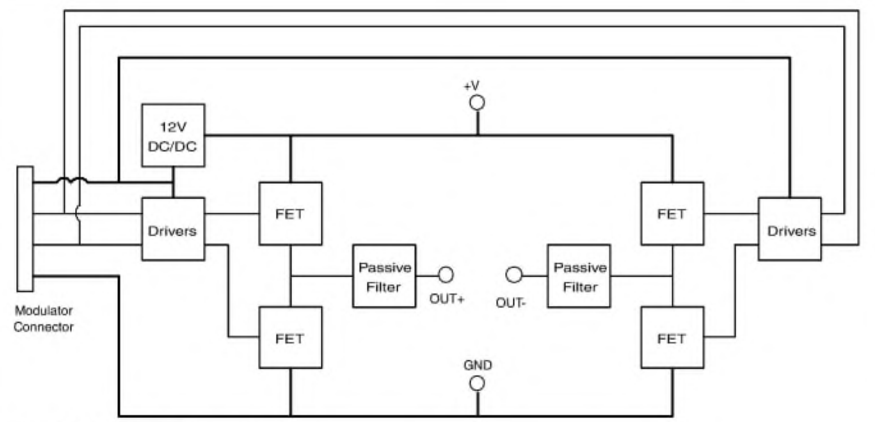


FIGURE 6: Output stage board.

tor and output stage makes it easy to shield the output stage by itself.

The SA1 runs from an analog input, but is a digital Class-D amplifier. An ADC converts the analog input to digital; this configuration is compatible with current surround-sound systems, which require a powered subwoofer and have a line-level analog output for that purpose. The system consists of two boards and a power supply. The ADC and complex programmable logic device (CPLD) are on the modulator board, and the output stage is on its own board (*Fig. 3*).

MODULATOR BOARD

The modulator board (*Photo 1*) converts the analog input to control signals used by the power circuits (*Fig. 4*). The heart of the system is a CPLD, which functions as a signal processor and controller. The controller portion runs the ADC and is used to shut down the output stage in the absence of audio (see next paragraph). The audio input to the CPLD is I^2S type (common interface for ADCs and DACs), and the output of the CPLD is PWM.

AUTO SHUTDOWN

Second, in digital Class-D, the audio level of the input is measured digitally, eliminating the need for a level detector (comparator). Third, the timer used to measure how long audio has been below the threshold is simply a counter in the digital domain. These advantages eliminate circuitry and increase reliability in the case of a digital Class-D amp.

DEAD-TIME DELAYS

PWM, generated internally by the CPLD, drives the output stage with two signals, high-drive and low-drive. This is done to prevent shoot-through, which is the condition where current flows through both FETs in series, effectively shorting out the power supply (*Fig. 5*). Shoot-through is a concern because the FETs and FET drivers have inherent propagation delays and limited slew rate. Delays in the drive allow one FET to turn fully off before another in series is turned on.

OUTPUT BOARD

The output board (*Photo 2*) converts the low-level drive signals from the CPLD to an output powerful enough to drive a speaker. It drives the load through a passive filter used to remove the switching frequency. High-current inductors are used in the filter to reduce series resistance. FETs function as switches between the power supply and the filter (*Fig. 6*).

The circuit accepts 3.3V-level signals and converts them to 12V-level signals using a FET driver. The 12V-level signals drive a "half bridge FET driver," which in turn drives the FET gates through small resistors. The FETs drive the passive output filter in a full-bridge configuration.

Digital Class-D amplifiers can be made full range. In order to reproduce

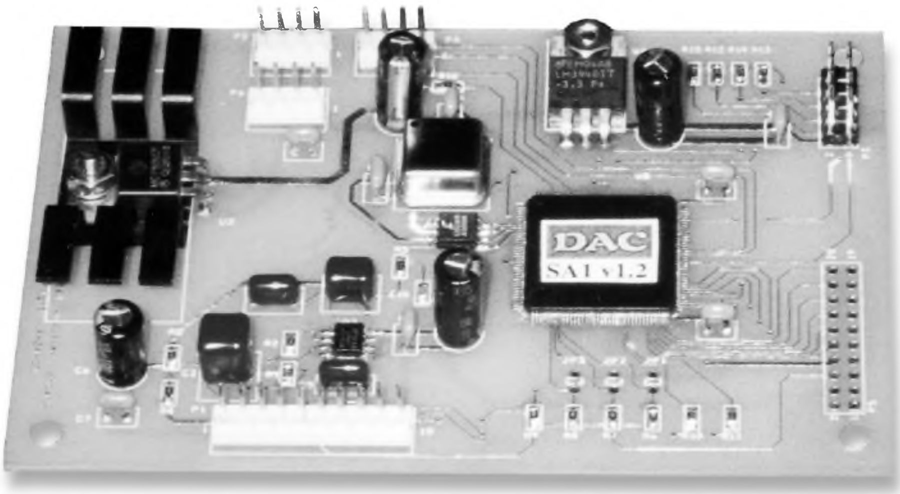


PHOTO 2: Output stage board.

SA1 SPECIFICATIONS

Rated power output: 115W into 4Ω (1% THD+N @ 100Hz)

Instantaneous peak power: 230W into 4Ω

THD+N: <0.1% up to 50W @ 100Hz

Signal-to-noise ratio: 90dB

Dynamic range: 90dB

Intended load: 4Ω

Frequency response: 20Hz to 1kHz ±1dB

Auto standby: 1 minute, 25 seconds after input is <30mV RMS

PWM switching frequency: 384kHz

Master clock: 49.152MHz

REFERENCES

1. Alan Bayko, "Build a 200-watt Digital Power Amplifier," *Electronics Now* 4/97, pp. 33-44.
2. Michael Fremer, "Sharp SM-SX100 Digital Integrated Amplifier," *Stereophile*, 7/00, pp. 73-79.
3. Bascom H. King, "Tact Audio Millennium Digital Amplifier," *Audio*, 6/99, pp. 52-57.

higher frequencies than the subwoofer range (>1kHz) with low distortion, additional digital signal filtering is needed. This additional filtering is not included in the SA1, which can be used full range if the analog input filter is taken out of the circuit, but the amp may sound harsh at high frequencies.

In Part 2 we'll continue with a look at digital amps in general, and the SA1, in particular. ❖

Test Tracks

Which music selections do you use to test your audio system?

Let's hear from you. Simply describe your seven favorite pieces (not to exceed 1000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to:

"Test Tracks"
Audio Amateur, Inc.
Box 876, Peterborough, NH 03458

We will pay a modest stipend to readers whose submissions are chosen for publication.

High-Quality ($180) Control Unit

If you are currently using a modified solid-state Integrated Power Amplifier or a passive control unit to obtain your audio output signals, this control unit will enable you to improve your system and get great performance, for a modest $180 investment. **By Joseph Norwood Still**

Here's a control unit (*Photo 1*) that will add realism, liveliness, and clarity to your audio system, regardless of the source: CD tuner, CD Walkman, satellite audio, phono, and so forth. It is absolutely quiet when operating in the line level, CD Walkman, or phono states. It is outstanding in the phono state and has enabled me to truly enjoy my record player.

The control unit has a two-section, six-position selector switch. Four positions are used for line-level operation, while the last two positions are dedicated exclusively to CD Walkman and phono use. The control unit also has right and left volume controls; to prevent crossover distortion, a balance control is not used.

The illuminated power switch turns your entire sound system on or off using a female AC receptacle that may be connected to a 6- or 8-AC terminal strip (*Photo 2*). This is a useful and very practical feature.

The control unit does not employ negative feedback in any of its amplifier stages or in the RIAA network. The unit (*Fig. 1*) provides excellent high quality performance. It also provides at least 30V (RMS) output in any mode (phono, CD Walkman, line amp) before clipping—at great dynamic range.

ABOUT THE AUTHOR

Joseph Norwood Still comes from the small town of Halifax, Pa. In the early '50s, he became interested in building amplifiers, but his career intervened. He did get one article published in *Electronics World* and in *Hi-Fi Annual*. After he retired, he began writing for *Glass Audio* as an outlet for his old interest in amplifiers.

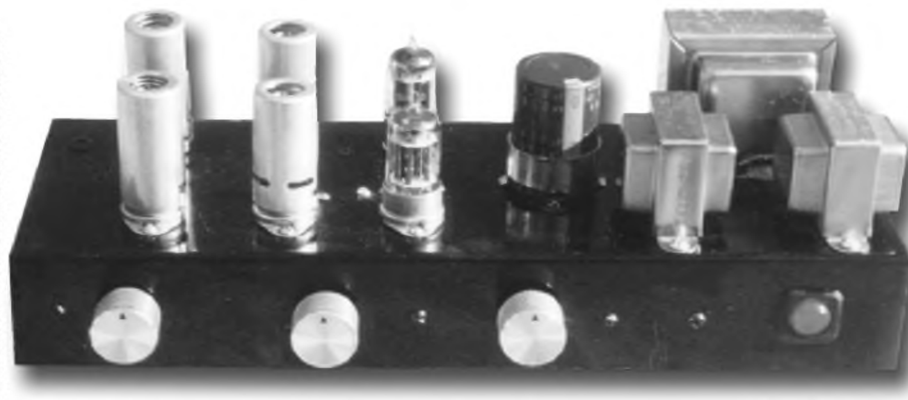


PHOTO 1: Front view of control unit.

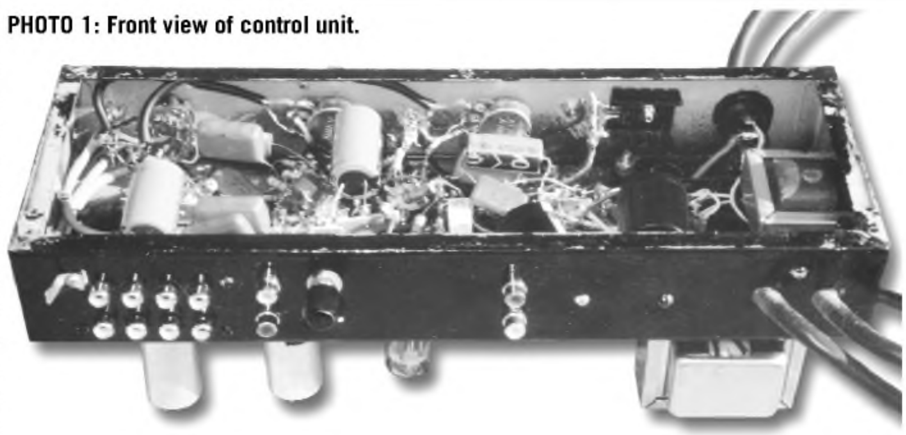


PHOTO 2: External view showing the rear apron and internal wiring.

LINE AMPLIFIER

The line amplifier uses a paralleled 5687 (V.G. of 12) to obtain low output impedance, wide bandwidth, and low noise performance. The heater of the 5687 is operated at a reduced voltage of 10V DC (versus 12.6V DC) to obtain low noise operation from this stage. (The heater power for the 5687 is excessive, operating at the same power as the 6L6. The high heater power was intended for use in a computer application. For the application, as a line-level audio amplifier, 10V/380mA is more than adequate for proper operation of the 5687.)

The frequency response of the 5687 at 6V RMS output is flat from 10Hz to 20kHz, tapering to 5.8V RMS at 30kHz, and is flat from 10Hz to 20kHz, feeding an amplifier via 6′ audio cable (Radio Shack 15-1505). The noise of the line-level amplifier stage is 0.4mV with volume control (R10) full-on and the input open, non-shorted. (The noise measurement is made using an oscilloscope with shielded test leads. All subsequent noise and ripple mea-

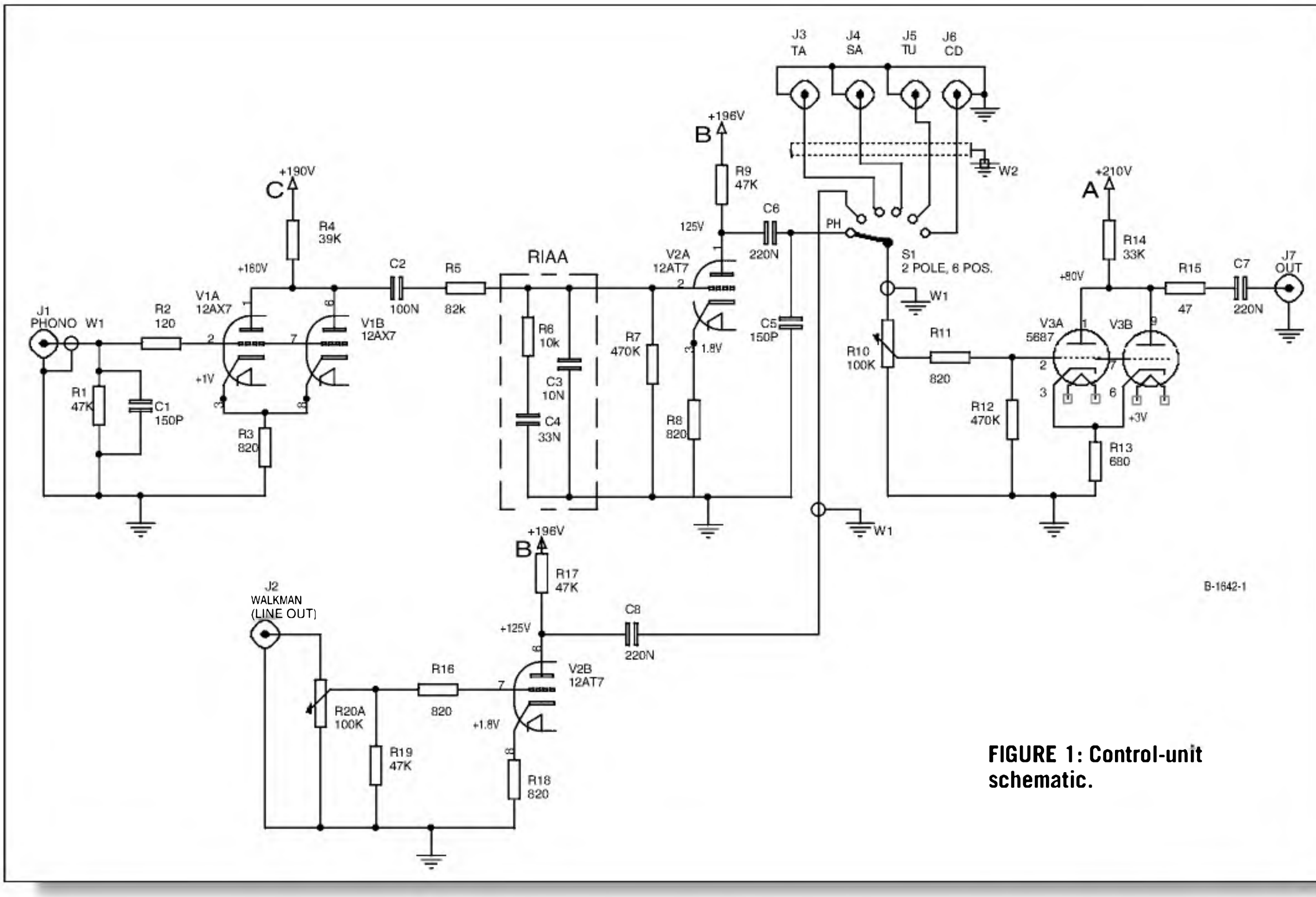


FIGURE 1: Control-unit schematic.

surements throughout this article use an oscilloscope.) The distortion from 20Hz to 20kHz of the line-level stage is less than 0.36% at 5V RMS output, while at 3V RMS output it is less than 0.24%. In my opinion, paralleling more than one 5687 is unnecessary, since the desired bandwidth is obtained and the sound is solid and well-defined. The 470kΩ (R12), 820Ω (R11), and 47Ω (R15) are added to the 5687 to ensure stability.

TABLE 1
NOISE COMPARISON TO SOLID-STATE PHONO PREAMPS

MY VACUUM TUBE PHONO PREAMP (MEASUREMENT AT OUTPUT OF LINE AMPLIFIER, 5687)	
Noise	1.3mV, R-channel, 1.5mV, L-channel, R10 set to ¾-on position
Input sensitivity at 1kHz	.003V for 3.0V output
KENWOOD STEREO INTEGRATED AMPLIFIER, MODEL KA-3500 (PHONO SECTION)	
Noise	17.2mV, R-channel, 13mV, L-channel*
Noise*	1.5mV, R-channel, 1.5mV, L-channel
Input sensitivity at 1kHz	.003V for 1.5V output (at output of volume control)
PYRAMID STUDIO PRO, MODEL PR6002 (PHONO SECTION)	
Noise	3.2mV, R-channel, 5.7mV, L-channel*
Noise*	1.8mV, R-channel, 2.5mV, L-channel
Input sensitivity at 1kHz	.003V for 1.0V output
REALISTIC (RADIO SHACK), MODEL 42-2109 (PHONO PREAMP)	
Noise	12.8mV, R-channel, 14mV, L-channel*
Noise*	6.5mV, R-channel, 6.2mV, L-channel
Input sensitivity at 1kHz	.003V for 0.25V output
ESOTERIC SOUND PROFESSIONAL MOVING-MAGNET PREAMP	
Noise	3.5mV, R-channel, 3.6mV, L-channel*
Noise*	1.0mV, R-channel, 1.0mV, L-channel
Input sensitivity at 1kHz	.003V for 0.28V output

*Ground receptacle of the phono preamplifier (or UUT) returned to the ground receptacle of AC wall plug. The ground wire of the record player has the same effect.

IMPORTANT: I made these measurements with the input of the devices open, not connected to a phono cable. The transistor does have noise performance advantages when connecting a low impedance phono cartridge by way of an appreciable length of shielded cable to a low input impedance of a transistor (or low noise of an FET); versus the high input impedance of a vacuum tube. This must be factored in when reviewing the noise measurements in *Table 1*.

TEST EQUIPMENT USED

H-P 331A distortion analyzer

Heathkit sine-wave, square-wave audio generator, IC-5218

DMM, Radio Shack 22-168A oscilloscope, Proteck G502

DRIVER AMPLIFIER

The driver amplifier uses a 12AT7. One section of the 12AT7 is used to provide additional amplification (V.G. of 16) for the low-level line output of a CD Walkman (typically 0.1V to 0.2V RMS). The other section of the 12AT7 (V.G. of 16) is used to amplify the phono signal. (For further details of this section of the 12AT7, see the section designated "Phono Preamplifier.")

Technical measurements for the CD Walkman stage (12AT7) are such that the frequency response of the driver amplifier is flat from 20Hz to 20kHz when measured at the output of the line amplifier, 5687. The frequency response is flat from 20Hz to 15kHz when connected to an amplifier through a 6′ cable (Radio Shack 15-505). The noise of the driver stage with the input open or with CD Walkman connected by way of a 3′ audio cable and all potentiometers set three-quarters-on is 1.0mV RMS. I measured the above at the output of the 5687 line amplifier.

The maximum distortion of the driver stage (12AT7) from 20Hz to 20kHz (measured at the output of the line amplifier, 5687) is 0.2% at 5V RMS and 0.12% at 3V RMS. I used the 47kΩ resistor (R19) to provide a proper match to the CD Walkman's line level output. The 820Ω resistor (R16) ensures stability of the 12AT7. The 100kΩ poten-

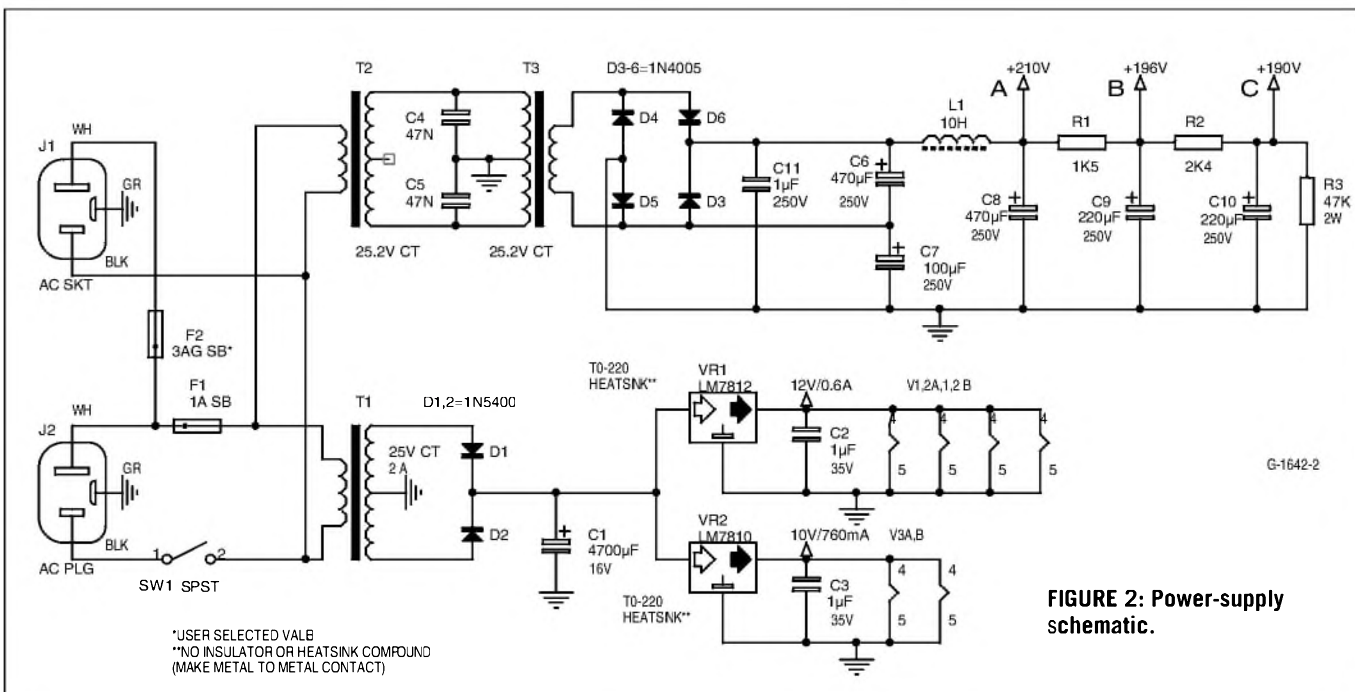


FIGURE 2: Power-supply schematic.

PHOTO 3: Stereo console.

tiometer (R20A) is required when the line output of the CD Walkman is constant; if the line output level of the CD Walkman is variable, you may omit the potentiometer.

Note: If the phono stage is not used, I recommend a 5814 or 12AU7 be used, since its voltage gain of 11 is more compatible with a CD Walkman or an external phono preamp. The 12AU7/5814 may be used without circuit changes and provides the same distortion as the 12AT7.

Optional change: The six 0.22µF coupling capacitors may be paralleled by 0.022µF capacitors (A.E.S., C-PD022-400) for high frequency enhancement.

PHONO PREAMPLIFIER

The phono preamplifier is very quiet; it has only 1.3mV right-channel and 1.5mV left-channel noise at the output of the 5687 line amplifier. I obtained this noise measurement with the volume control (R10) set to its three-quarters-on position and the input of the phono section open (ungrounded). There is no "frying egg" sound, hum, microphonics, or acoustical rumble. The 12AX7 and 12AT7 shielded tube cans must be in place for lowest noise performance. Also, the ground wire from the record player motor must be secured to the ground "lug" on the preamplifier.

The preamplifier is well designed. Paralleling the Sovtek 12AX7LPS reduced the noise and increased the gain by 27%. To reduce noise and distortion I employed no cathode bypass capacitors in the control unit.

The gain of the cascaded 12AX7 and 12AT7 typically provides a "swinging" 1.0 to 2.5V RMS output from the line amplifier (5687) when playing records. I'm not fond of SRPP circuits, and paralleling the 12AX7 enabled me to develop a phono-stage that is simple and alleviated the need for using the typical circuit arrangement of two cascaded 12AX7 triodes. This setup provided the low-noise, low microphonic outcome. Also, paralleling the 12AX7 appears to give it the sonic characteristics of a medium mu-triode. Obtaining the proper results from the RIAA circuit required several attempts. I finally obtained an RIAA curve, which gave excellent sound throughout the audible spectrum. I got this result with a flat response from 20Hz to 12kHz, tapering to 0.5dB down at 18kHz; a sharp drop-off occurs from 20Hz to 10Hz.

Every component from the plate of the paralleled 12AX7 to the grid of the 12AT7 can be considered part of the RIAA network. The 82kΩ resistor (R5) provides impedance decoupling and loading to the RIAA network (R6,C4,C3). The 470kΩ resistor (R7) also loads the RIAA circuit.

Capacitor C1 (150pF) tailors the high-frequency output of the magnetic pickup and also acts as a stopper for any noise or radio frequency pick up, which I did not experience. Capacitors C1 and C5 and resistors R2, R11, R15, and R16 are added to ensure stability. Listening tests confirmed that 150pF is the maximum value that should be used with cartridges specified at a 220pF load. The 1.0V RMS output of the paralleled 12AX7 phono stage causes distortion of less than 0.25% from 20Hz to 20kHz (measured at the plates of the 12AX7).

Typical output of the phono preamplifier, measured at the output of the line amplifier (5687) when playing a record, is a “swinging” 1 to 2.5V RMS. The noise of the phono preamplifier, measured at the output of the 5687 when connected to the record player with a 9′ audio cable (6′ Radio Shack 15-505 audio cable plus 3′ record player cable), is 4mV RMS. With the 9′ cable disconnected, the noise of the phono preamplifier, measured at the output of the 5687, is 1.3mV right channel and 1.5mV left channel with R10 set to three-quarters-on position. This preamplifier is intended for moving-magnet phono cartridges with typical outputs of 3–7mV.

The noise measurement of 1.3 to 1.5mV of my phono preamplifier was better than four solid-state phono preamplifiers I tested (*Table 1*). Does this brief analysis indicate that a well-designed tube phono-preamplifier has as good or better noise performance than the typical commercial solid-state phono preamp?

Note: Prior to making the noise measurements, the volume control R10 is set to three-quarters-on. This setting corresponds to a 1.0V RMS output at the 5687 with a 3mV, 1kHz signal at the input of the 12AX7.

It is important to note that the inclusion of the phono preamp circuit creates complex wiring problems and crowds the components. It also requires good soldering techniques, good vision, and placement of the parts for minimal electrical interaction. If you are comfortable with this environment, then building the phono preamp circuit

AMPLIFIER PARTS LIST ($88)

DOUBLE-ORDER PARTS PRECEDED BY*

C1, C5	150pF	AES	*CSM 150
C2	0.1μF, 400V	AES	*CPD 1-400
C3	0.01μF, 400V	AES	*CPD 01-400
C4	0.033μF, 400V	AES	*CPD033-400
C6-C8	0.22μF, 400V	AES	*CPD 22-400
R1, R9, R17, R19	47k, 1W	AES	*RE 47k
R2	120, 0.5W	AES	*RI 120
R3, R8, R11, R16, R18	820, 1W	AES	*R-B 820
R4	39k, 1W	AES	*R-B 39k
R5	82k, 1W	AES	*R-B 82k
R6	10k, 0.5W	AES	*R-1 10k
R7, R12	470k, 0.5W	AES	*R-1 470k
R10	100k, single pot, audio taper	RS	*RS271-1722
R13	680, 1W	AES	*R-B680
R14	33k, 2W	AES	*RF33k
R15	47, 1W	AES	*R-B47
R20	100k, dual pot, audio taper	RS	271-1732
S1	2 pole, six positions	RS	275-1386
Knobs	two packs	RS	274-424
Aluminum cans, shield		AES	*P SS9-162
V1	Sovtek 12AX7LP	AES	*12AX7LPS
V2	12AT7	AES	*12AT7
V3	5687	AES	*5687
VX1-VX3	T9, 1⅛″ mount center, ¾″ hole	AES	*P-ST9-213
J1, J7	RCA jacks	RS	*274-852
W1	Single-conductor shielded cable (3′)	RS	271-1722
W2	Four-conductor shielded cable (3′)	RS	910-1604
	Order from RadioShack.com, 1-800-442-7221		
Chassis	Steel 13.5 × 5 × 2	AES	P-H1441-18
Bottom Plate	Steel 13.5 × 5	AES	P-H1413-18
Rubber Feet		RS	64-2342

POWER SUPPLY PARTS LIST ($88)

T1	25V, C.T., 2A	RS	273-1512
T2, T3	25.2V, C.T., 450mA	RS	273-1366
C1	4700μF, 16V	RS	U119355095
C2, C3	1μF, 35V	RS	272-1434
C4, C5	0.047μF, 50V	RS	272-1068
C6-C8	470μF, 250V	ALEL	EC-4725
C9, C10	220μF, 250V	PE	020-1440
C11	1μF, 250V	RS	272-1055
VR1	7812 (12V, 1A)	RS	276-1771
VR2	7810 (10V, 1A)	PE	NTE-1932
2 HEATSINKS	TO-220	RS	276-1363
CR1-CR2	IN5400	RS	276-1141
CR3-CR6	IN4005	RS	276-1104
L1	10H, 270Ω, 90mA	AES	PTC-7X
R1	1.5k, 5W	AES	RQ 1.5K
R2	2.4k, 2W	AES	RF 2.4k
R3	47k, 2W	AES	RF 47k
FX	Fuse holder, chassis mount	RS	270-739
F1	3AG, 1A, slo-blo	RS	270-1021
S1	SPST, lighted, rocker, 10A	RS	275-692
F2	3AG, slo-blo, user determined	RS	———
Extension cord, three-prong male, female connector rated for 15A		—	———
seven solder-type terminal strips (five-lug tie)		RS	274-688
1-roll, rosin core 60/40 solder		RS	64-006

RS = Radio Shack
AES = Antique Electronic Supply 480-820-5411
PE = Parts Express, 1-800-338-0531
ALEL = All Electronics 1-800-826-5432

described in my article should create no problems; if, however, you prefer "uncluttered" projects and simplicity, then I recommend that you omit my phono preamplifier and add a remote preamp.

An excellent preamp that provides broadcast-quality performance is the Esosteric Professional Phono Preamp, which is available for $80. Eliminating the phono preamp in my article saves about $35. This would bring the cost of the control unit, plus the esosteric phono preamp, to about $225.

POWER SUPPLY

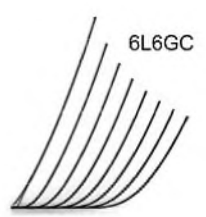
The power supply has a DC heater supply and a DC plate supply. The plate supply has two transformers, T2 and T3 (120V AC, 25V AC, 450mA), hooked "back to back." Transformer T3 is tied to a full-wave rectifier bridge (CR3-CR6), with a voltage-doubler circuit formed by capacitors C6 and C7.

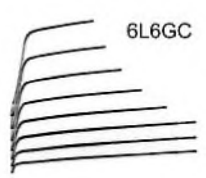
The output of C6 is fed to a 10H choke, L1. Capacitors C8, C9, and C10 and resistors R1 and R2 provide additional filtering and decoupling to the driver and phono stages of the control unit. The plate-supply output is terminated by bleeder resistor R3. The output of the plate supply is 210V DC, 25mA, and the AC ripple is 20mV RMS. Noise from the 120V AC line is filtered at the center-tapped secondary of transformer T3 by capacitors C4 and C5.

The heater supply has a 120V AC primary to 25V AC (2A) secondary transformer T1, which is connected to a full-wave rectifier (CR1, CR2) and filtered by capacitor C1. The DC voltage at the output of the heater supply is 16V DC, 1.3A, and the ripple voltage is 0.4V AC RMS. The output is fed to two voltage regulators VR-1 and VR-2, which are cooled by two heatsinks (Radio Shack TO-220, 276-1363).

The output of VR-1 is 12V DC and the ripple voltage is 15mV RMS. The output of VR-2 is 10V DC with a ripple voltage of 12mV RMS. Capacitors C2 and C3 prevent VR1 and VR2 from oscillating.

The power supply has an illuminated switch, S1, and two fuses, F1 and F2. Fuse F1 is a 3AG type, rated at 1A. You should determine the type and rating of





F2 based on your user requirements; for my system I use a 3AG, 8A fuse. The fuse-holders are Radio Shack, chassis mount type (270-739).

The power cord for connecting the control unit to the 115V AC line is a three-conductor extension cord, rated for 15A service. It is cut to the desired length and the male-end is connected to the nearest AC wall receptacle. The female end of the cut extension cord is used as a receptacle for a six- or eight-connector power strip (*Fig. 2*).

Important: Be sure to maintain the proper order of connection when wiring the ends of the extension cord wires to the fuses, switch, and transformers of the control unit. The green wire is ground, the black is "hot," and the white is neutral. Warning: Lethal voltages are present; exercise appropriate caution when constructing the control unit, and never leave the control unit upside down if children are present.

ACOUSTICAL INTERACTION

The control unit and its various amplifier stages are impervious to microphonics and acoustical rumble. It is mounted in a cabinet (*Photo 3*), which is 66″ × 30″ × 19″ and houses a 12″ subwoofer, two 8″ 20 oz magnet mid-woofers, two 5″ 15 oz magnet cone-type midranges with sealed backs, and four soft-dome tweeters. It also houses my 40W triode/60W ultralinear amplifier (*GA* 2/98).

The remaining equipment is located at the center of this box, which is also the location for the control unit and exposes the control unit to severe vibrations when the power amplifier is operating at full output levels. Despite the vibrations of the box, the control unit's phono and other amplifier stages do not exhibit any signs of acoustic rumble or microphonics. The record player sits only 4′ from the cabinet and exhibits no acoustic interaction.

The vibration and the environment are so severe that I used a piece of foam to prevent the pickup of the CD player from jumping. I also applied stick-on rubber feet to the bottom of the CD Walkman to prevent its pickup from jumping. If it handles this environment well, you will have no problems with an installation with a centrally located equipment rack and remotely located speakers.

OPTIONS

You may omit the two 12AX7 phono stages. I definitely recommend keeping the CD Walkman stage, since it is a real convenience to have a single-disc CD player (must have a line output jack). The other section of the 12AT7 should be used, since it's a real convenience to have an extra stage that can amplify a 0.1V RMS signal; for this stage I recommend adding a 100k dual potentiometer. If you do have a record player, I definitely recommend building the phono stage, since it's truly an outstanding performer.

CONSTRUCTION TIPS

I used point-to-point wiring throughout the control unit, which also includes four five-lug terminal strips for the power supply components and three five-lug terminal strips for the amplifier—two terminal strips are located between the 12AT7s, and two are located between the 12AX7s. The volume control (R20) is located near the rear of the chassis, while the choke (L1) is on the far side of the power supply (*Photo 4*). The two heatsinks (TO-220) and voltage regulators VR1 and VR2 are opposite each other in the power-supply compartment.

The fasteners to secure tube sockets are 2-56, while those for the six five-lug terminals are 4-40. I used 6-32 fasteners to secure heatsinks and voltage regulators VR1 and VR2. I secured transformers with 10-32 fasteners.

Fourteen RCA phono jacks are required. Locate two at the output of the 5687s, two at the input of the 12AT7, and ten near the 12AX7s (all are located on the back-panel of the chassis). Four aluminum tube covers, which are used on the 12AX7s and 12AT7s, are required.

The plate resistors of the two 12AX7s are returned to a common tie-point of the five-lug terminal strips and bypassed to ground and filtered by capacitor C9. The plate resistors of the two 5687s are returned to a common tie-point of the power-supply five-lug terminal strip and bypassed to ground and filtered by capacitor C8.

Use shielded microphone cable for the inputs and outputs of 12AX7, 12AT7, and 5687 as required. Use four-conductor shielded cable to connect J3-J6 to rotary switch S1. (Ground all cables at both ends). The three 470µF capacitors are mounted beneath the chassis in the vicinity of the transformers. Do not locate the one capacitor on top of the chassis as shown in the photo.

You should place a ground lug (right- and left-channel) between 5687 and 12AT7. The grounds of RIAA circuit and 12AT7 grid and cathode resistors should be soldered to this ground lug. A ground lug should be between the right- and left-channel volume controls. The ground-taps of the potentiometers should be soldered to the lug.

All ground-lugs throughout the control unit must have metal-to-metal contact. This requires scraping the paint in the vicinity of the lug from the chassis until the bare metal is exposed. Be sure to use shielded (microphone) cables for all long signal leads in the control unit. ❖

Speaker City's MTM-18 Loudspeaker Kit

Reviewed by Mark Florian and Joseph D'Appolito

PHOTO 1: **The MTM-18 loudspeaker.**

PHOTO 2: **Scan-Speak 18W/8545K woofer and D2905/9900 Revelator tweeter.**

Tweeter: Scan-Speak D2905-9900 Revelator; woofers: Scan-Speak 18W/8545K; advertised frequency response: 40Hz–25kHz; sensitivity 91dB (1W/1m); power rating: 150W (RMS); cabinet dimensions: 24″H × 8¾″W × 12⅛″D; weight: 41 lbs; price: $1,611.74 shipping not included. Speaker City USA, 115 S. Victory Blvd., Burbank, CA 91502, (818) 846-9921, FAX (818) 846-1009, www.speakercity.com.

Most readers are long familiar with the reputation and quality of Scan-Speak drivers. Headquartered in Vidabaek (Veed ah bek) Denmark, Scan-Speak supplies drivers to several high-end loudspeaker manufacturers all over the world. What you probably don't know about Scan-Speak is that it was originally formed by Dynaco (Denmark Audio Company). After changing hands several times, Vifa purchased it in the early '90s. Then a few years later the factory was burned to the ground by a disgruntled employee.

Torben Sondergaard took over as managing director of Scan-Speak shortly after the fire and brought Lars Goller over from Vifa. Lars designed the SD (symmetric drive) motor systems; the NRC (non-resonance chamber) used in their tweeters which lowers the resonance frequency; and the NRCS (non-resonant cone structure). Basically, Lars designed and developed the present-day Scan-Speak line. Today he is chief engineer of the whole company, Vifa/Scan-Speak.

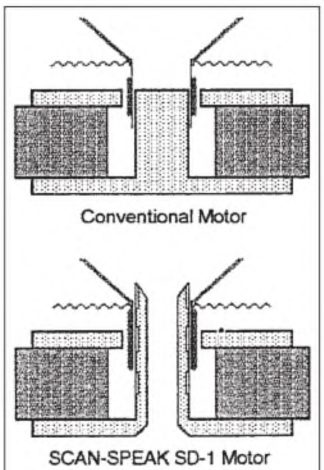


FIGURE A: **Comparison of Scan-Speak SD-1 motor system versus conventional.**

FEATURES

The MTM-18 (*Photo 1*) was designed by Murray Zelligman at the request of Scan-Speak USA to showcase their unique drivers. The design uses the 18W/8545K 7″ woofer (*Photo 2*), which features a cone constructed of carbon fiber loaded paper which is then coated with a damping compound, resulting in a very stiff cone that is well damped. Other features include a cast magnesium basket, rubber surround, linear spider, and a 1.67″ (42.5mm) 100W voice coil. F_S is listed as 28Hz, V_{AS} is 49 ltr.

I measured the cone diameter from the center of the surround at 5.43″ (138mm), thus giving an effective cone area of 23.25 in^2 (150cm^2). Scan-Speak's patented magnetic motor design, SD-1, allows for a large p-p *linear* excursion of 13mm and a *maximum* excursion of 20mm. The voice coil is over three times longer than the height of the air gap! By extending the pole piece well beyond the top plate, the voice coil always surrounds an equal amount of the pole, thereby producing a symmetric magnetic field around the air gap (*Fig. A*).

The benefits of this design are greatly reduced low-frequency second-harmonic distortion and intermodulation distortion. Copper shorting rings are also bonded to the pole piece to prevent large swings in voice-coil inductance during large excursions. Finally, the top of the vented pole piece is concave and shaped like an inverted cone, which eliminates reflections between it and the dust cap while also cooling the motor. This is a unique and sophisticated motor system.

The tweeter is the Scan-Speak D2905/9900 Revelator 1″ soft-dome (*Photo 2*). Unique to this tweeter is a machined aluminum faceplate 130mm in diameter, which helps to control directivity down to 2kHz, thus providing a smooth off-axis response to better

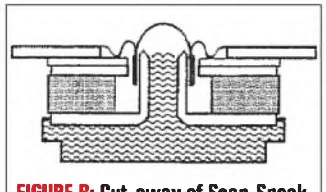


FIGURE B: **Cut-away of Scan-Speak D2905/9900 motor.**

PHOTO 3: Contents of the MTM-18 kit, shown with new crossovers.

match the woofer. Other features include a long-coil, short-gap geometry (*Fig. B*) much like the woofer, the SD motor system, a 1.1″ (28mm) 225W voice coil, no ferrofluid, and an inverted cone-shaped pole piece, beveled on the outside, vented into a large rear chamber to eliminate reflections between the pole and the dome and lower resonance. F_S is listed as 500Hz, linear excursion is 1mm p-p, and maximum excursion is 3mm p-p.

KIT CONTENTS

The contents of the MTM-18 kit are shown in *Photo 3*. Initially the kit was shipped with a different crossover, which was assembled very poorly (more about this later). The crossovers shown in *Photo 3* were sent later and assembled much better.

All resistors are Axon 1% non-inductive wirewound units with a power dissipation rating of 12W RMS. All capacitors are 5% metallized-polypropylene Axon True Caps. The Solo air-core inductors are made by CFAC and employ a 14 AWG equivalent copper foil claimed to be 99.99% pure. The layers of foil are separated by thin sheets of clear polypropylene. Claimed benefits of this design are: negligible skin-effect resistance, lower DC resistance, and less inductor reactance error.

The crossover schematic is shown in *Fig. C*. The woofer circuit is second-order with a Zobel network to flatten out the impedance rise at higher frequencies. The tweeter circuit is third-order with a small resistor in the "T" leg to adjust the damping. An RLC trap is used to tame the resonance peak, since no Ferrofluid® is used. Finally, a resistor voltage-divider is employed to bring down the tweeter level to match that of the woofer.

CROSSED-OVER CROSSOVERS

I noted several problems with the first set of crossover networks that were shipped with the kit (*Photo 4*):

- The components were attached to ¼″ hardboard with silicone adhesive. The component leads and connecting wires were simply twisted together and then soldered, without anchoring them to the board in any way. Speaker City chose Axon 8 speaker cable between the binding posts and the boards, between the separate woofer and tweeter boards, and to connect the woofers. This cable is .4″ in diameter and consists of "8-20 AWG solid-core conductors, PVC jacket, on a non-interleaved spiral. Four conductors with black insulation, four conductors with gray insulation." The cable is very stiff and rather unwieldy in short lengths.
- Several of the thin, fragile leads of the film inductors broke (*Photo 5*).
- The tweeter was connected with a smaller Axon cable consisting of four 22 AWG solid-core copper wires encased in a clear jacket. Closer inspection revealed a very thin, clear, plastic wrap between the wire and jacket. This wire was also stiff and inappropriate for the thin, fragile connectors on the tweeter.
- The other end of the cable wasn't terminated properly at all. I tugged firmly on all the crimped connectors and most came off in my hand! Closer inspection revealed that the wrong tool was used on all the solderless connectors. Crimp dies for an *uninsulated* crimp connector and *insulated* crimp connectors are different; they don't even look the same.
- I also found numerous instances in which the solid-core wire had been nicked when stripped of its insulation (*Photo 6*), which will cause the wire to break at this point when stressed.

Many months later I received a new set of crossovers (*Photo 7*). This time the construction was much better. All parts were soldered to brass through-holes that had been anchored to a plastic base plate. Where external wires needed to be connected, male ¼″ spade connectors were anchored to the board at the connection point.

I still have a concern regarding the film inductors and how their leads are attached. It seems a hole was punched through the foil lead, so that the through-hole could be inserted. The leads themselves are less than 0.17″ wide,

PHOTO 4: Original crossovers.

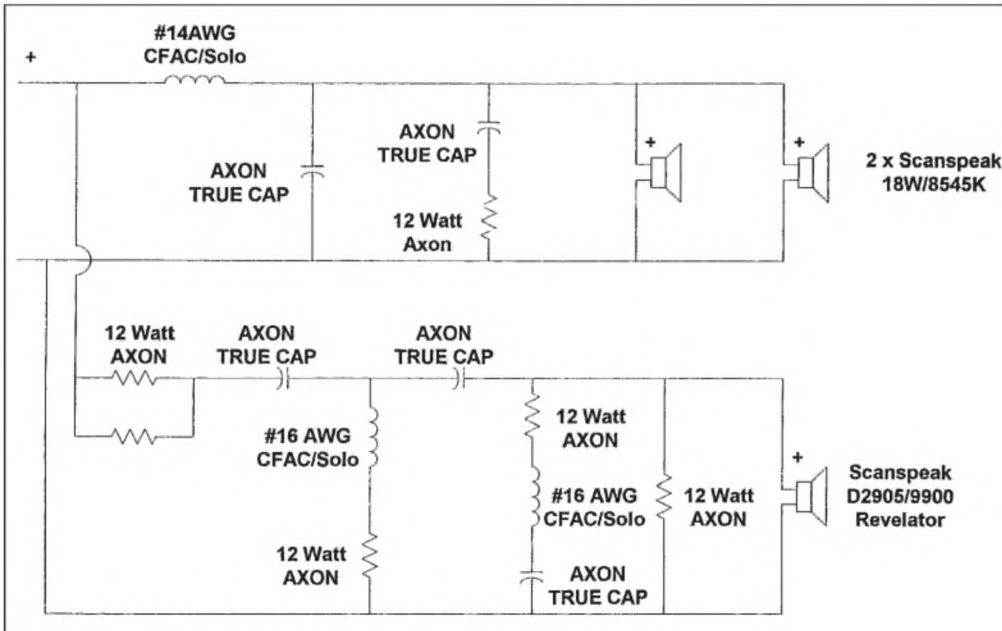


FIGURE C: MTM-18 crossover schematic.

PHOTO 5: **Note shiny copper foil just outside the coil bobbin where inductor lead used to be attached. The strain of the large Axon cable attached to the other end caused it to fatigue and break.**

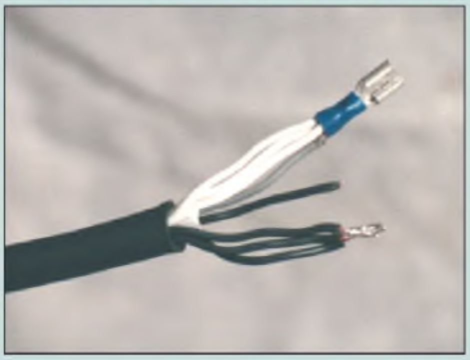


PHOTO 6: **Short wires on woofer connector broke off, longer wires pulled free of the crimp. Note lack of solder on longer wires.**

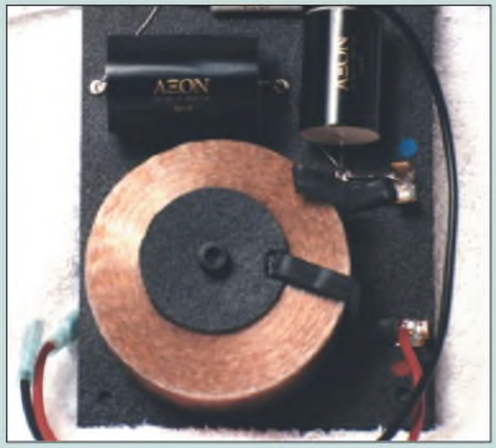


PHOTO 7: **New woofer crossover.**

and the through-holes seem to be just as wide if not wider, thus they might actually sever the lead. I was able to solder most of these, but on some the heat-shrink tubing ran right up next to the joint, so there wasn't any copper showing to solder.

I recommend that Speaker City use a short piece of 18 AWG solid copper with the end bent back on itself, forming a U on its side. The foil could then be looped through this in an S-shape. Then the U could be crimped shut with a pair of pliers and the joint soldered. This would give a strong mechanical connection, and the solder would fuse the joint and protect it from oxidation. The other end of the wire could be inserted into the through-hole and soldered in place.

The new crossovers also had color-coded dots on the boards that made identification of the terminals easy, since the new instructions referred to connectors by their color. A nice touch. I placed small dots on the connection diagram using corresponding colored markers.

Included with the new crossovers was several feet of Axon 8- and 4-conductor cable, which I did not use. It's actually nice wire and there is nothing wrong with it, but it's meant for connecting a speaker to an amplifier, not for inside wiring. Also included were only seven insulated female solderless connectors, none of which fit the small connectors on the tweeter.

Fortunately, I had on hand several sizes of *uninsulated* crimp connectors, the appropriate crimping die, and several feet of Belden 8719 cable, out of which I pulled the inner conductors. This is an 18 AWG *stranded*, tinned wire made of electrolytic tough pitch (ETP) copper (99.95%) insulated with polyethylene. It's very flexible and solders easily because it's already tinned.

Also included in the kit was a large piece of Axon 1 5/8″-thick five-layer Black Hole pad. No information was included as to the construction of this material, and I was unable to find any on Speaker City's or Orca's websites or by calling Orca.

Then one day, I happened to find an info sheet in an old Focal file. The five layers consist of "a high-loss vibration damping material, a polyester urethane open cell foam, a thin, flexible vinyl barrier septum, another layer of polyester urethane foam" and finally, facing the air in the cabinet, "a thin diamond pattern embossing, densified with polyurethane film surface." That last description is more than a little cryptic.

CABINETS

The cabinets are nicely finished in oak veneer with a deep red-brown color and solid wood to protect the veneer on the edges. The front and back are painted gloss black. This cabinet is slot-loaded at both the top and bottom of the front, as opposed to employing cylindrical vent tubes. These slots (1/2″ × 7 1/4″) are of such an extreme height-to-width ratio, though, that surely they cause port nonlinearities and excessive noise at higher volume levels. Because of their location, they also cause the enclosure to be taller than it would be if round ports were installed on the rear.

Missing was the usual hole in the back for a mounting cup. Instead, long Axon binding posts are fitted directly into the back of the cabinet via pre-drilled holes. I actually prefer these posts, as they don't require cutting out a section of the cabinet. In addition, they are of higher quality than binding posts commonly encountered in cups.

Contrary to the supplied information, the cabinets do not contain "H" braces. The two "braces" used each consist of an oval piece of 3/4″-thick MDF that has had most of the inner material removed, to resemble a square "O," leaving a perimeter that is approximately 0.8″ thick. So, there is no "bar" connecting the sides of the brace or the sides of the cabinet. This doesn't make for much of a brace.

One of these braces is used above the tweeter and another below it; there is no other bracing in the cabinet. The front is made of 1″ MDF, the sides, top, and bottom, 3/4″. I was unable to measure the back, but the supplied information says it is 1″ MDF. Finally, the woofer holes are chamfered on the backside to ease the abrupt edge.

PUTTING IT TOGETHER

Assembling the kit was not particularly difficult, but the instructions could have been clearer. For instance, the first instruction says to mount the crossovers inside, but it doesn't say where. Ordinarily, this is not much of an issue, but this cabinet is different due to the two slot vents in both the top and bottom, neither of which can be obstructed with crossover or foam. Since the holes were predrilled for the posts towards one end of the cabinet, I initially assumed they do not mount there.

Another consideration is the bracing installed rather close to the middle of the cabinet, thus preventing both boards from being installed side-by-side on the back. I called the company seeking advice, but they weren't of any help, replying "mount them wherever."

So I figured the best way given the above constraints was to plug the predrilled holes with hot glue and redrill them in the center of the cabinet, right behind the tweeter (*Photo 8*). This way the boards could be separated from each other at the maximum possible distance from the woofer magnets without obstructing the slot vents.

With the posts sticking out from the back of the cabinet, you can no longer lay it down on the floor. So I used the open end of an egg crate as a stand. The instructions didn't include what lengths to cut

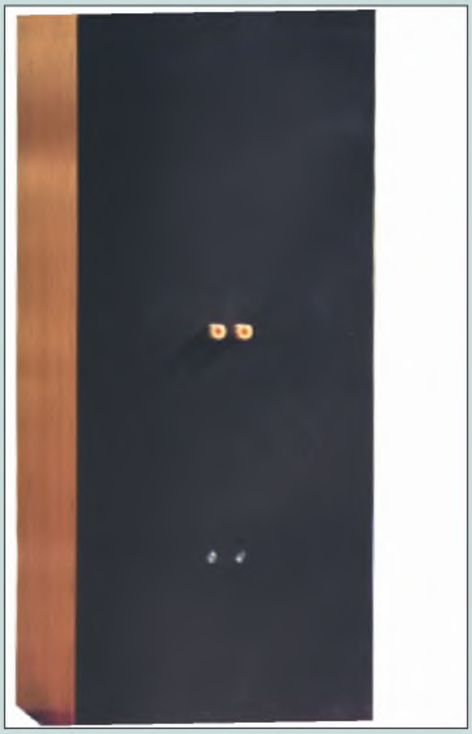


PHOTO 8: Binding posts have been moved to the center of the cabinet. Note pre-drilled holes when cabinets were built have been plugged with hot glue.

the supplied wire, so I decided to measure this first, install the female connectors on one end of the wire, then connect this to the crossover *before* gluing the crossovers in place.

I cut the wires to length, crimped on the uninsulated connectors with the *correct* die and then soldered the joint. The back of the crossover boards was covered with a thin foam material held in place with adhesive. Possibly to minimize vibration of the crossover boards? I don't know. I decided to use hot glue, which hardens quickly and holds well, to attach the boards to the back.

The next step is to mark and drill the mounting holes for the drivers. At this point, I noticed that the surround was completely separated from the frame in several places on one of the woofers. Closer inspection revealed that the surround was attached at only a few points and that there was no adhesive on the frame. I checked the other three, which all were firmly attached around their perimeters. When I received the woofers, they were in the factory-sealed boxes, so clearly this was a manufacturing defect that had passed unnoticed by Scan-Speak's QC. I took it to a friend in town who specializes in loudspeaker repair and he fixed it up.

I used a combination square to line up the driver mounting holes so that they were horizontal and not tilted. I find this easier than measuring as the instructions suggest. To mark the holes, I used a sharpened pick, very similar to an ice pick, only shorter. This makes a noticeable mark on the black baffle and gives a small indentation for the drill bit to prevent it from skidding across the surface.

When drilling the mounting holes, I highly recommend using a brad-point bit. These are true wood bits as opposed to the blunt-nosed metal bits commonly encountered. The sharp point prevents the bit from skidding while starting the hole. After drilling all holes, I vacuumed out the shavings.

Installation of the Black Hole pad comes next. The instructions didn't specify how large to cut the pieces or how to install them, and the guy I talked to at Speaker City didn't know either. A little math showed that if I covered both side walls, the opening for the woofer magnet would be reduced to $3\frac{7}{8}$″. The diameter of the magnet is 4.82″, so clearly it wouldn't fit.

I would need to move the pad back from the front about 3.25″ since this is the height of the woofer. This would leave a bare wall right next to the driver. In addition the width of the pad would have to be further reduced to account for the crossover boards, which stand about $2\frac{1}{2}$″ tall and take up nearly the entire width of the cabinet. Remember, these pads are $1\frac{5}{16}$″ thick.

By now it was becoming obvious that this wasn't going to work. Black Hole 5 is simply too thick to be used in a cabinet as small as this. I'm sure in big subwoofer cabinets, it works fine. Furthermore, I was also reminded of what happened in the review of the Parts Express dual 5″ Vifa kit (*SB* 1/98, p. 45), in which too much foam that was too thick was stuffed into a vented enclosure and killed the low-frequency response. I didn't wish for that to happen here. More is not always better.

Fortunately, I had on hand some gray open-cell foam that is about 0.82″ thick. I cut pieces to line the enclosure sides and shelves of the two slot vents, and a small piece to cover the rear between the braces where the binding posts are. Finally, I covered the crossovers, making sure to keep it away from the entrance to the slot vents.

Next, I applied the gasketing material to the rear of the drivers. The gasket supplied is very small, $\frac{1}{16}$″ thick × $\frac{3}{8}$″ wide. However, it easily fit between the holes and frame on the woofer so it wasn't necessary to pre-punch the gasket from the holes before inserting the screws. To make insertion of the connectors to the drivers easier, I applied a little Cramolin. This is particularly important with the tweeter because the terminals are thin and very fragile.

It also appeared that the terminal length on the tweeter had been shortened by clipping it off, though I don't know why. The black screws supplied featured a fine-thread like a drywall screw, so I substituted some I hand on hand from Lee Valley Tools that feature a much coarser thread more appropriate for MDF. Note that a #8 screw will not go through the holes on the 8545K drivers. I used a #6 × $\frac{3}{4}$″ screw (#01Z65.06).

With the loudspeakers assembled, I played a 25Hz sine wave at enough volume to exercise the suspension for about 12 hours, followed by the break-in noise track from *Stereophile*'s Test CD #3 for an additional 24 hours. Facing the speakers toward each other and reversing the connections to one will cancel most of the noise and help maintain the peace.

Using a LofTech TS-2 sine wave generator, I adjusted the frequency until I observed minimum cone motion by watching a cone dot and then turned up the volume to a medium level. This would be close to the box tuning frequency where cone motion is minimum and vent output is maximum. I measured 42Hz. At this point, there was a significant amount of vent noise present, confirming my earlier concerns about the width/height ratio.

SPEAKER SETUP

Equipment used in evaluating the MTM-18s included a California Audio Labs Icon Mark II CD player and a POOGE'd 5.5 Philips DAC960 D/A converter, using the variable outputs to feed one of Erno Borbely's Servo-100 amplifiers. The dimensions of the room are $18\frac{1}{2}$′ long by $15\frac{1}{2}$′ wide, with an opening to the kitchen $8\frac{1}{2}$′ wide by 8′ tall on the right side, when facing the speakers. The floor is carpeted and the furniture consists of a large entertainment center behind the speakers, a piano, large table, and two couches. The ceiling is vaulted with the center ridge 11′ high. All listening was done with the grille cloths off.

Initially, I used some Target metal speaker stands with spikes that stand 16″ tall. This put the tweeter at 28″ above the floor. I adjusted the spikes so that each stand was level on the concrete floor, punching through the carpet and pad. Then I sat down for some listening.

LISTENING TESTS

When evaluating speakers, I like to use material that contains good recordings of female vocals and acoustic instruments, such as guitar and piano. Right away I noticed a pronounced bass response; it sounded excessive and out of balance with the rest of the spectrum. This effect still persisted after listening to several recordings, so I decided to make a change.

(to page 51)

■ TESTING SPEAKER CITY'S MTM-18 LOUDSPEAKER KIT

REVIEWED BY JOSEPH D'APPOLITO

I ran a series of impedance, frequency response, and distortion tests on the Speaker City MTM-18 loudspeaker kit constructed by Mark Florian. *Figure 1* is a plot of system impedance magnitude. At low frequencies the plot displays the double-peaked curve of a vented system. The impedance minimum of 3.5Ω at 38Hz indicates the vented-box resonant frequency. There is a second local minimum impedance of 3.0Ω at 136Hz.

In addition to the two impedance peaks at low frequencies, a third peak occurs at 1.1kHz, where the woofer and tweeter crossover networks interact to form a parallel resonance. Impedance phase lies between +50° and −56° over the full audio range. This system can reasonably be rated at 4Ω.

FREQUENCY RESPONSE

Figure 2 shows the MTM-18's far-field frequency response with the microphone placed along the tweeter's axial centerline at a distance of 1.35m. This is a quasi-anechoic response[1], which is valid above 200Hz. The plotted response has been normalized to 1m to obtain system sensitivity.

With vented systems I usually obtain low-frequency response by combining near-field port and woofer responses. The MTM-18 ports exit on the front baffle. It is very difficult to get independent port and woofer near-field data when they are this close together. For this report I used a ground-plane measurement (see chapter 4 of reference 1) to get the low-end response (*Fig. 3*).

The system low-frequency response (*Fig. 3*) was spliced to the quasi-anechoic response (*Fig. 2*) at 200Hz to get the full-range system response (*Fig. 4*). Sensitivity averages 88.3dB SPL/2.83V/1m over the two octaves centered on 1kHz. Relative to this level the −3dB low-frequency point is 47Hz.

Between 400Hz and 1300Hz there is a response trough that is 2dB below the average response above 2kHz. This response depression in the critical midrange may give the MTM-18 a somewhat dark or recessed perspective. Below 400Hz response rises 5dB, peaking at 86Hz relative to the 400Hz level. Brief listening tests in my studio confirmed the somewhat exaggerated bass response. The effect, however, was not unpleasant.

CUMULATIVE SPECTRAL DECAY

The MTM-18 cumulative spectral decay (CSD) response (*Fig. 5*) shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.05ms increment of time. The total vertical scale covers a dynamic 33dB range.

Ideally the response should decay to zero instantaneously. Inertia and stored energy that take a finite amount of time to die away, however, characterize real loudspeakers. A prominent ridge parallel to the time axis would indicate the presence of strong system resonance.

The first time slice in *Fig. 5* (0.00ms) represents the system frequency response. There are no strong ridges in this CSD. Tweeter decay is rapid and well controlled. The woofer and its crossover network control decay response below 3kHz. The overall decay performance is quite good.

SYSTEM STEP RESPONSE

Figure 6 is a plot of system step response. It is obtained by a numerical integration of the system impulse response. The ideal step response should be a single rapid rise followed by a smooth decay through the 0.00 level.

Figure 6 shows two separate arrivals of acoustic energy. The initial, sharper, positive spike is the tweeter arrival. It is followed by the woofer arrival, peaking about 0.4ms later. Both drivers are connected with positive polarity, but the system is not time-coherent.

A better view of this behavior is seen in *Fig. 7*, which is a plot of excess group delay versus frequency referenced to the tweeter's acoustic phase center. This is a plot of delay in milliseconds versus frequency. (For a detailed description of excess group delay, see reference 1.) In a time-coherent system this plot would be a flat line.

Above 10kHz excess group delay is essentially zero because it is referenced to the tweeter arrival time in this frequency range. The curve rises gradually below 10kHz, reaching a plateau below 1kHz of 318µs, or 0.318ms. This plot shows that over its operating frequency range, the woofer is 318µs behind the tweeter.

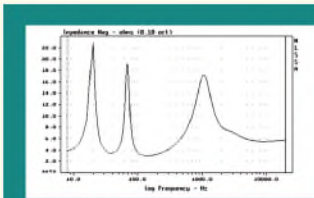


FIGURE 1: MTM-18 impedance plot.

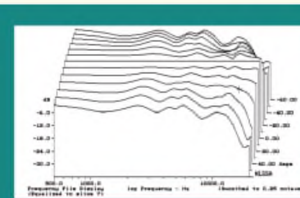


FIGURE 8: Horizontal polar response.

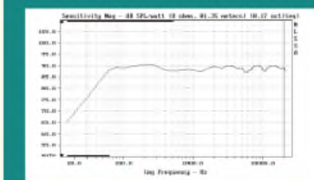


FIGURE 2: Far-field response on-axis.

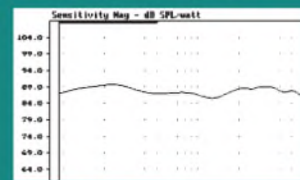


FIGURE 9: Horizontal response over a 60° window.

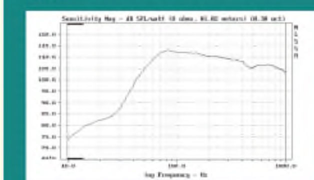


FIGURE 3: Low-frequency ground-plane response.

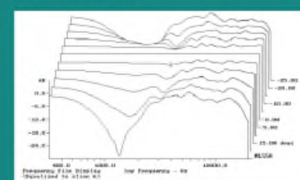


FIGURE 10: Vertical polar response.

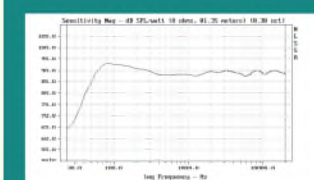


FIGURE 4: Full-range response.

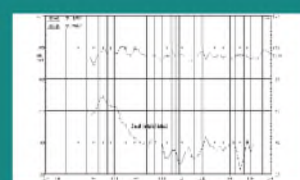


FIGURE 11: Second harmonic distortion.

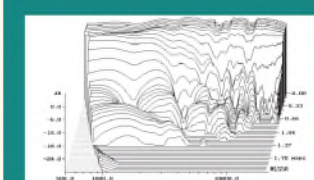


FIGURE 5: Cumulative spectral decay.

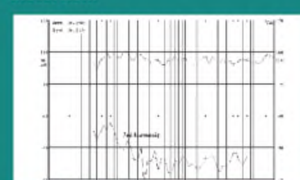


FIGURE 12: Third harmonic distortion.

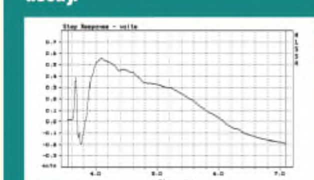


FIGURE 6: Step response.

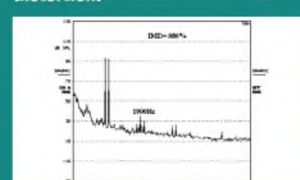


FIGURE 13: Woofer intermodulation distortion.

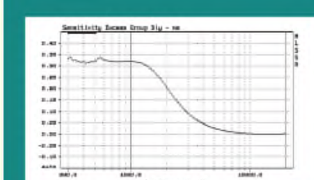


FIGURE 7: Excess group delay.

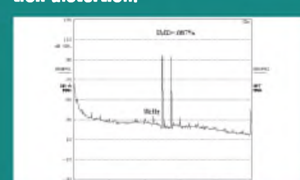


FIGURE 14: Tweeter intermodulation distortion.

ABOUT THE AUTHOR

Joseph D'Appolito, *aX* regular contributor and author of many papers on loudspeaker system design, holds four degrees in electrical and systems engineering, including a Ph.D. Previously, he developed acoustic propagation models and advanced sonar signal processing techniques at an analytical services company. He now runs his own consulting firm specializing in audio, acoustics, and loudspeaker system design. A long time audio enthusiast, he now designs loudspeaker systems for several small companies in the US and Europe.

(from page 47)

I spread the speakers 60° apart and placed them on top of the Swan IV bass cabinets. This put the middle of the tweeter dome 45″ above the floor. I also moved the speakers back further so that they were now 9′ away from my sitting position. The bass response was smoother and more even. The soundstage was wide and deep and very detailed. Bass response was authoritative, given the twin woofer MTM design.

The stiffness of the carbon fiber cone, superior magnet structure, and long-throw makes the Scan-Speak drivers more than capable of producing tight, clean, and "quick" bass transients without distortion or blurring of the detail. In addition to music, the MTM-18s performed well with several movies, even without a subwoofer.

After many hours of listening to all types of music, a couple of patterns began to emerge. First, the Revelator tweeter is extremely

TESTING SPEAKER CITY'S MTM-18 LOUDSPEAKER KIT

POLAR RESPONSE

Polar response is examined in *Figs. 8–10*. *Figure 8* is a waterfall plot of horizontal polar response in 10° increments from 60° left (−60°) to 60° right (+60°) when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00°. For good stereo imaging the off-axis curves should be smooth replicas of the on-axis response with the possible exception of some tweeter rolloff at higher frequencies and larger off-axis angles.

You can see the expected rolloff of tweeter response at higher frequencies and larger off-axis angles. This performance is fairly typical of 28mm dome tweeters. *Table 1* lists the left and right off-axis responses at 15kHz and all angles up to ±60°. Below 8kHz horizontal coverage is the broadest and smoothest of any speakers I have tested in this series.

The average response over a 60° horizontal angle (±30°) in the forward direction is shown in *Fig. 9*. The response trough between 400Hz to 1.3kHz persists at all angles over the 60° range. However, there is almost no change in frequency response relative to on-axis levels. This indicates good direct field coverage in the primary listening area with almost no change in spectral balance with position. Image stability should be very good.

Figure 10 is a waterfall plot of vertical polar response. Responses are shown in 5° increments from 25° below (−25°) the tweeter axis to 25° above it. Worst-case departure from on-axis response at ±5° is −1.3dB. At ±10° the worst-case departure is −4dB. In both cases this deviation occurs near the crossover frequency.

At larger angles we see broad dips forming through much of the midrange. This is actually an advantage of the MTM geometry, since it eliminates strong floor and ceiling reflections through the critical midrange.

HARMONIC DISTORTION

I ran harmonic distortion tests at an average SPL of 90dB at 1m. Ideally, harmonic distortion tests should be run in an anechoic environment. In practice, it is important to minimize reflections at the microphone during these tests. Out-of-phase reflections can produce false readings by reducing the level of the fundamental while boosting the amplitude of the harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5m from the loudspeaker.

Figures 11 and *12* show second and third harmonic distortion levels in dB SPL versus frequency plotted in 1/6-octave steps. Distortion is plotted in dB SPL level. Unsmoothed system frequency response is also plotted on these figures. The second and third harmonic distortion levels at 50Hz are 2.8% and 0.68%, respectively.

All system harmonic distortion is well below 1% above 120Hz. Tweeter harmonic distortion is below 0.15% at almost all frequencies. This is excellent performance. The low-frequency harmonic distortion levels are among the lowest I have reported.

INTERMODULATION DISTORTION

I next measured intermodulation distortion. In this test two nearby frequencies are input to the speaker. Intermodulation distortion produces output frequencies that are not harmonically related to the input and are much more audible and annoying than harmonic distortion.

Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

I first examined woofer intermods by inputting 900Hz and 1kHz signals at equal levels. These frequencies should appear predominantly in the woofer output. Total SPL with the two signals was adjusted to 90dB at 1m. The MTM-18 system output spectrum for this test is shown in *Fig. 13*.

The two largest spectral lines represent the input signals. The largest distortion product is second order at 1900Hz and is 61.3dB below the main output, which is equivalent to 0.086% distortion. This is better than many solid-state amps, most tube amps, and any other speaker I have reported on! (The lines each side of the 1900Hz line at 1800 and 2000Hz are harmonic distortion components.)

I measured tweeter intermods with a 10 and 11kHz input pair also adjusted to produce 90dB SPL at 1m (*Fig. 14*). The largest intermods are at 9 and 12kHz. Total IM distortion is 0.087%. Again, the best level of any tweeter measured so far.

The last IM test examines cross-intermodulation distortion between the woofer and tweeter using frequencies of 900Hz and 10kHz. (A 1kHz signal would produce intermods that fall on harmonic distortion lines, confusing the results.) Ideally, the crossover should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. The spectrum resulting from this test is shown in *Fig. 15*. The largest IM product at 10.9kHz is 64.9dB below the main output. Total distortion is only 0.06%. An excellent result!

ADDITIONAL TESTS

I conducted all of the above tests with the grille off. *Figure 16* shows the MTM-18 system response with the grille on, but referenced to the response with grille off; that is, it plots the difference in response under the two conditions. Below 2kHz the grille has little effect. Above 2kHz, however, the grille causes ragged response deviations of 3.5dB peak-to-peak. As usual the grille has only cosmetic value.

Two samples of the MTM-18 system were available for testing. The two units were arbitrarily labeled "A" and "B." All of the tests described so far were conducted with the "A" sample.

One question of interest is how well the two samples match? Frequency-response matching of the "A" and "B" pair is shown in *Fig. 17*. This is a plot of the response difference between the "A" and "B" samples. The two systems match within ±0.5dB up to 7kHz. Above 10kHz differences of ±2.5dB are seen.

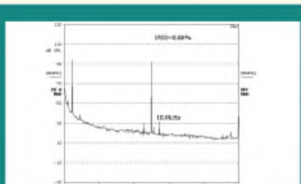


FIGURE 15: Woofer/tweeter cross intermods.

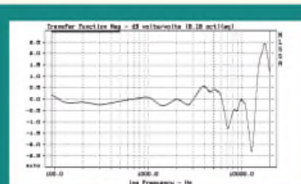


FIGURE 17: MTM-18 comparison: system A versus system B.

FIGURE 16: Effect of grille on MTM-18 frequency response.

REFERENCE

1. J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Corporation, Peterborough, NH, 1998.

TABLE 1

LEFT/RIGHT POLAR RESPONSE AT 15KHZ

ANGLE	LEFT (DB)	RIGHT (DB)
10	−0.6	−0.7
20	−2.0	−2.3
30	−4.5	−4.1
40	−7.0	−6.5
50	−11.1	−9.3
60	−15.2	−12.6

A note on testing: The Speaker City MTM-18 kit was tested in the laboratories of Audio and Acoustics, Ltd. Measurements were made with the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems using an ACO 7012 ½″ laboratory-grade condenser microphone and a custom-designed wideband, low-noise preamp. A computer-controlled OUTLINE turntable on loan from the Old Colony Division of Audio Amateur Corporation was used to perform the polar response tests.

open and detailed and not kind to recordings with even the slightest bit of brightness, for they will be revealed without any masks whatsoever. Smooth recordings sounded that way. Good recordings of cymbals, gongs, and bells were remarkably transparent and crystal clear; the Revelator is easily capable of reproducing their sharp transients. Remember this tweeter uses no Ferrofluid. Recordings that, from experience, sounded overly bright, were reproduced as such. This would be an excellent tweeter for testing various crossover components, as it is accurate enough to reveal the finest details.

Second, the midrange seemed a bit recessed between the bass and treble. It sounded fine with just vocals and a piano, for instance. But as soon as other instruments appeared with the vocals, they seemed to upstage it. With recordings that feature a good deal of bass material, the midrange will be upstaged even more.

COMMENTS ON MEASUREMENT RESULTS

The tall narrow peaks on the impedance plot (*Fig. 1*) show that the vented box cabinet exhibits a high enclosure Q, indicating low box losses. If I had lined the walls with the Black Hole 5 pad as provided, they would have been much shorter, possibly killing the box Q. According to Joe D'Appolito, Black Hole 5 is not as efficient in increasing the effective box volume as regular foam, due to its high-density layers. It can actually decrease it if used excessively.

The frequency response plot of *Fig. 4* confirms my findings in the listening tests; namely, the exaggerated bass response and the recessed midrange. Since Joe spliced *Figs. 2* and *3* together at 200Hz, *Fig. 4* shows that the difference in level between 80 and 200Hz is −2dB. Between 200 and 400Hz, the level drops another −3dB.

Though the +2dB peak at 80Hz is not unpleasant by itself, as Joe noted, it stands out even more because of the falling response from 80Hz through 400Hz. Furthermore, the critical midrange region from 400 to 1500Hz is down 5dB compared to this peak. This is a bit much for such an expensive kit. I don't expect it to be ruler flat; however, the critical midrange shouldn't be relegated to such a wide, deep trough either.

I decided to run a simulation in TopBox because my suspicion is that the enclosure is too small. Cutting through all the measurements, it appears that the empty box volume of the supplied cabinets is 26.64 ltr. When drivers, crossover, and bracing are subtracted, it leaves just 22.96 ltr.

TopBox's suggested volume is 29 ltr using published T/S numbers on Scan-Speak's website, but 51.5 ltr using numbers I measured with Clio. This is quite a discrepancy. In comparison, the North Creek Rhythm using the 8545 woofers in a QB3 alignment has a box volume of 42 ltr. The gap closes when you add the 3.68 ltr I measured for crossover, bracing, and drivers to the 29 ltr TopBox number, giving 32.68 ltr.

Using Clio, I measured the four 8545K drivers, but averaged only three of them together: f_S = 30.71Hz, Q_{MS} = 6.866, Q_{ES} = 0.37, Q_{TS} = 0.335, V_{AS} = 36.68 ltr, Re = 5.6Ω, Mms = 22.283g, C_{MS} = 1.21, BL = 8.057, L(1K) = 0.74mH, L(10K) = 0.26mH. Plugging these numbers into TopBox results in an optimum box volume of 51.5 ltr and a box tuning frequency of 32.8Hz.

It would seem that these drivers are stiffer than usual, with f_S, Q_{MS}, and Mms being high and V_{AS} being on the low side, thus causing the larger box size. For whatever reason, it appears the box is too small for a pair of these woofers, thus leading to the large peak at 86Hz. Perhaps this small box was used for aesthetic reasons or to accentuate the bass response. It would be an interesting exercise to build a larger cabinet, install the drivers and new crossover, and re-evaluate.

While looking over the results from Clio, I noticed that one set of the driver parameters was quite a bit different than the others. Upon closer examination of the suspect driver, I noticed that the spider had "sunk" toward the magnet, thus flattening out the normally curved surround. It's as if you press on the cone and hold it there. This is sometimes referred to as "sunken spider syndrome," which is a sure sign of a defective driver. I checked the other three and they appeared normal.

This is the first time I have witnessed this defect in a kit. Since the drivers were sealed in factory boxes, I contacted Scan-Speak USA and a replacement was sent. Upon its arrival, I broke it in and then checked the T/S parameters with Clio. They were within range of the other three. It turns out that the defective driver was the same one with the loose surround described previously.

I wish I had been able to catch this problem before the speakers were shipped to Joe for testing. However, at the time I had no reason to suspect the defective driver had anything more than a loose surround. I've been told that sometimes a spider will sink after it's been broken in, thus appearing normal out of the box. Other times its defect will be immediately apparent.

The effect of the unique faceplate on the Revelator tweeter is shown in the horizontal polar response waterfall plot of *Fig. 8*. Broad, smooth, even coverage. The vertical polar response waterfall plot of *Fig. 10* shows the characteristic broad dips throughout the midrange at higher angles that the MTM design is known for.

The distortion curves are extremely low, no doubt due to Scan-Speak's excellent motor design. *Figure 15* shows that the crossover is performing an excellent job of keeping cross-intermodulation distortion to an absolute minimum. Putting a grille on the MTM-18 is a big mistake, as shown by the ragged response in *Fig. 16*. Finally, *Fig. 17* shows that the two speakers are pretty evenly matched, save for the response above 10kHz, where large differences in the tweeter response are encountered. Matching the tweeters could probably improve this significantly.

CONCLUSION

The MTM-18 is a somewhat mixed bag. The drivers are certainly extraordinary and capable of incredible detail, transparency, and depth, *when implemented with a proper crossover and proper enclosure*. Possibly one or both of these is at fault for the tipped-up bass response and the recessed midrange. It all depends on your taste and what type of music you enjoy listening to.

The large, stiff Axon cables are very difficult to work with and terminate properly, much more so in a small enclosure. The first set of crossovers was very poorly assembled, which is disappointing given the quality of the components used. The second crossovers were assembled much better, but the same Axon wire was included. The holes for the binding posts were located in the wrong place. I don't see how the crossovers could have been mounted in the cabinet without putting them on the shelf that makes up the slot vents.

The cabinet itself really needs additional bracing, at least a true "H" brace as advertised. Better yet, a vertical brace would stiffen the long sides. The narrow slots should be removed in favor of tubes inserted into the rear baffle to free up additional room within the enclosure. The instructions need to be re-written with far more detail, such as where to put the crossovers, how long to cut the wires, and what size to cut the foam pieces.

Finally, my overall impression is that this kit is composed of a number of parts without much thought given to integrating them into a complete system. Thus the system performance doesn't live up to its potential. The sound is not well balanced between bass, midrange and treble, as it should be.

I believe if the enclosure volume was increased and the overall frequency response better balanced, then this would be an outstanding kit worthy of the price. In its present incarnation, however, it needs more work. ❖

SOURCES

Lee Valley Tools
USA: 12 East River St., Ogdensburg, NY 13669
Canada: 1090 Morrison Dr., Ottawa, ON K2H 1C2
(800) 871-8158, FAX 1-800-513-7885, e-mail: customerservice@leevalley.com, website: www.leevalley.com

Speaker City USA's response:

First, we would like to thank audioXpress *for giving us the opportunity to have one of our kits reviewed in the magazine.*

This was our first review with audioXpress, *and we provided them the best parts available to us at the time. We suggested improving the standard version of the MTM-18 by adding a Black Hole 5 dampening pad and using Axon 4 and Axon 8 internal hookup wire (even though the standard version of this kit comes with .75" open cell foam for dampening insulation and #14 soft-braided copper for internal hookup wire). These added improvements are not necessarily easy to apply, but make a substantial difference to the overall sound and performance of the kit if you are inclined to work through the minor challenge of installation.*

THE CROSSOVER

*The hand-built preliminary crossover network (*Photo 4*) that was first included with the MTM-18 kit was prematurely sent before we had the chance to have our production facility build the correct unit. We normally have all of our crossover networks built by Vertek/XTC located in Grass Valley, Calif. But at the time we were approached to review this kit, we had not received the final production model of this crossover. As soon as the new models arrived (*Photo 3*), we sent them immediately to Mr. Florian. We should have waited for the production models to be completed before shipping, but were eager to have the review started and the product delivered as soon as possible.*

All of our networks we supply are precision-made to the highest tolerances and quality control. We have never experienced failure of any of the crossover units we have produced and have the fullest of confidence that our customers will be completely satisfied with their performance. Slide-on terminals are supplied to make the proper connections to the crossover and speaker terminals, or the customer can solder all connections.

THE CABINET

After weighing all possibilities, we opted to incorporate slot vents in MTM-18 because we believed this would produce better bass response and also add more internal cabinet structural support. Optimally, in order to eliminate port noise, the port area should be equal to that of the woofer's cone area, but this can only truly be realized in tuned passive radiator systems. In a tuned port system in order for the port to tune correctly in the given enclosure volume, the area of the port is typically smaller than the speaker's cone area. This is a common dilemma posed by bass reflex systems, and we believed that simply placing a port in the rear, as suggested by Mr. Florian, would deprive the cabinet of the additional structural bracing and also reduce the system's low-frequency dynamics.

Earlier versions of the MTM-18 had only a single open brace. They were originally designed with an H-brace, but we decided to keep the brace open after doing more study of the effects of internal reflections caused by brace surfaces. All of the cabinets we offer have a common philosophy in mind: aesthetics, sturdiness, and cost value (of course, there are virtually no limits that you can go to when building a cabinet and making a cabinet with as low resonance as possible, but at what financial costs?). For those who are interested, we offer construction plans so you can basically go hog-wild with bracing and multi-layer composite panels (which we support!).

We've come to the conclusion that a line needs to be drawn in order to preserve the cabinet's integrity as well as offer an affordable product to our customers. Also, we suggest the use of a Black Hole 5 dampening pad to further eliminate any internal cabinet panel vibrations. And lastly, we have made production changes to the enclosure to position the terminals in the center of the box to assist in mounting the crossovers.

THE RESPONSE

To address the issue of the low-frequency rise that is mentioned by Mr. Florian and documented by Mr. D'Appolito, installing Black Hole 5 in key positions of the interior of the cabinet as well as stuffing a small amount of Dacron fiber into the port exit can control this. This "damped port" concept controls the rise that the smaller cabinet causes and increases the dampening characteristics of the woofers. This "tweak" is subjective to an individual's preference of bass and the type of audio information that is used, as well as the speaker's relative position to the perimeter room. We have considered the idea of a larger enclosure but at the time prior to the review, we decided that a bookshelf/stand type would be our first choice for production.

*Of the hundreds of these kits sold, only one customer ever brought up the rise in the low frequency range, and at that time we gave him these suggestions and resolved this problem. As an alternate solution we offer a version of the MTM-18 kit that utilizes the 18W/8546 Kevlar woofers (*The Kevelator*). Using the Kevlar cone in this cabinet works to help control the rise in the lower-frequency response. If you have any further questions regarding this subject or need data or support, you can call Frank Guerrero in technical assistance at 818-846-9921 or e-mail staff@speakercity.com.*

QUALITY CONTROL

The one woofer mentioned in the review that experienced damage was purely a transportation issue that was caused by the woofer shifting inside the factory box during transit, not by Scan-Speak's lack of quality control. We have sold hundreds of thousands of Vifa/Scan-Speak components over the past 24 years and never experienced any type of quality-control problems. This supports a level of confidence agreed upon by our customers, as well as manufacturers who choose to use them. The damaged driver was immediately replaced by Scan-Speak without hesitation.

SUMMARY

For 24 years Speaker City USA has had a long tradition of striving to be of utmost help to our customers and will continue to do so for many years to come. We would like to note that the system in review has been sold to hundreds of audio enthusiasts here and abroad, with not one of them having been returned. Here are just a few responses from some of our satisfied customers:

"I love my MTM-18s! What a sound! I am completely satisfied with their performance and my tube amps sound great on them. Many thanks, Speaker City."
Michael Quach—System Specialist & Audio Enthusiast, Honeywell Engines & Systems, Sun Valley, Calif.

"It was the combination of your knowledge combined with well-crafted products that made shopping with you a pleasant experience."
Phil Orlandini—Department of the Interior & Audio Enthusiast, Denver, Colo.

"You guys are great! Thanks for all your expert service and product support! You have a customer for life."
Eric Samsell—Audio Enthusiast, Valley Village, Calif.

"The Scan Speaks I purchased from you are amazing! My mixes have never sounded better."
Jeffery Thomas—Studio Engineer, Los Angeles, Calif.

Product Review

Marantz PMD340 Pro CD

Reviewed by Gary A. Galo

Marantz PMD340 Compact Disc Player. Superscope Technologies, Inc., 2640 White Oak Circle, Aurora, IL 60504, (630) 820-4800, FAX (630) 820-8103, www.superscope-marantzpro.com. Price: $789.

PHOTO 1: Front view of the Marantz PMD340. This solid, professional CD player has excellent operation feel and a variety of user-modifiable features.

The PMD340 (*Photo 1*) is the top model of Marantz's current line of professional compact disc players. There are three models in the line, nearly identical in appearance. The PMD330 has unbalanced RCA analog outputs and an RCA S/PDIF digital output. The PMD331 and PMD340 add balanced XLR analog outputs, along with S/PDIF or XLR and Toslink optical digital outputs (*Photo 2*).

All models have a high-performance CD mechanism, but the PMD340 has a heavy-duty laser mechanism with diecast metal parts to increase reliability under the most demanding professional conditions (*Photo 3*). All models include a pitch control with a ±12% range of adjustment in 0.1% increments (12% is a whole tone). Pitch is adjusted with a rocker switch on the PMD330; a rotating "pitch wheel" is used on the 331 and 340, both of which also incorporate a ±8% pitch bend, a feature sometimes used by DJs. A rack-mount front panel is standard on all three models.

All players feature built-in infrared remote-control receivers, along with RC5 remote connections on the rear panel. The previous-generation Marantz PMD321 and 320 required the purchase of an external infrared receiver in order to operate with I/R remotes. You can still buy the outboard I/R receiver, which plugs into the RC5 jack, for installations in which the remote receiver needs to be placed in a location different from the CD player. The RC5 system allows several Marantz products to be daisy-chained with RCA cables and controlled by a single remote.

The Marantz RC1020DC hand-held I/R remote control is not supplied with any of their CD players—the remote control must be purchased separately. The PMD340 also has a 25-pin "D" connector on the rear panel—this is a GPI I/O control port, which allows fader start.

Other professional features shared by these new Marantz players include Auto Cue, adjustable Fade-In/Out, adjustable end-of-track monitor, and adjustable end-of-track warning. It is also possible to set up these players for automatic continuous play on power-up. Program Play, Random Play, and Repeat Play are also featured.

These are also the first CD players, professional or consumer, which will play CD-RW discs (as long as the discs have been properly finalized). All three models include variable-output headphone jacks. The displays also support the CD Text function. If the model number, PMD340, sounds familiar, it's because Marantz has recycled it—around 20 years ago the PMD340 was a portable cassette deck.

CIRCUIT DETAILS

Like its predecessors, the PMD320/321

PHOTO 2: Rear view of the PMD340. This top-of-the line player in Marantz's current professional series has three digital outputs, along with both balanced and unbalanced analog outputs.

ABOUT THE AUTHOR

Gary Galo is Audio engineer at the Crane School of Music, SUNY Potsdam, where he also teaches courses in music literature. A contributor to AAC since 1982, he has authored over 150 articles and reviews on audio technology, music, and recordings. He is the Recording Review Editor of the *ARSC Journal* (Association for Recorded Sound Collections), and is co-chair of the ARSC Technical Committee (www.arsc-audio.org). He also reviews books for *Notes–Quarterly Journal of the Music Library Association*, has written for the Newsletter of the Wilhelm Furtwängler Society of America (www.wfsa.org), and is a contributor to the *Encyclopedia of Recorded Sound in the United States.*

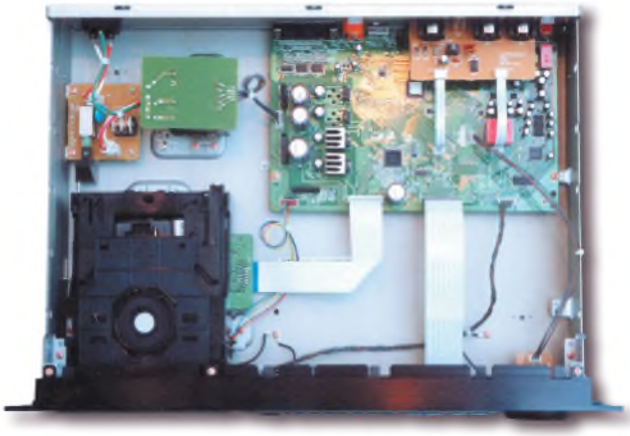


PHOTO 3: Inside the PMD340. The high-performance mechanism incorporates a heavy-duty laser pickup, designed for the demands of the professional user. All ICs on the main PC board are surface-mount devices.

series (the PMD320 was reviewed in *AE* 4/98), Marantz has simplified the design by using a combination digital filter and D/A conversion chip. For this purpose, the new player features a Burr-Brown PCM1710 chip, which is one of the best combination chips currently available, incorporating an 8× oversampling digital filter, a delta-sigma DAC, and an analog voltage output (the I/V converter is built-in). The 1710 can accept either 16-bit or 20-bit word lengths.

The left and right voltage outputs of the PCM1710 feed a JRC NJM4560 dual op amp. A simple third-order analog filter is incorporated in the feedback loop of each channel of the 4560. The 4560 op amps are also used to provide the balanced analog outputs in the PMD340.

An NJM4556 dual op amp is used as the headphone amplifier. Electrolytic coupling capacitors are used throughout the analog signal paths. The ±15V supply rails for the analog circuitry are regulated by NJM78M15 and NJM79M15 three-terminal, 500mA IC regulators. The +5 and +6V digital supplies employ NJM7805 and NJM7806 IC regulators.

A Sony CXD2585Q decoder chip, which feeds the PCM1710, also provides the digital output signal. A TX176 Toslink transmitter and the pulse transformer for the RCA S/PDIF output are tied directly to the digital output line, without any buffering.

In the PMD340, the digital output line also feeds an SN75158 dual differential line driver (compatible with the EIA/TIA-422-B and ITU V.11 interface standards). The low-impedance differential outputs of the SN75158 feed a pulse transformer for the XLR digital output. The PMD340's digital outputs remain active when the pitch control is engaged.

Many of the new 96kHz/24-bit outboard D/A converters, such as The Parts Connection's DAC 2.6 and 3.0, will operate with variable-pitch CD transports without an external sampling rate converter. Some D/A converters incorporating an additional low-jitter phase-locked loop may lose-lock when used with variable-pitch transports. I am uncertain whether The Parts Connection's DAC 2.7 and 3.1 upgrades will function in this mode without an external sampling rate converter.

FEATURES

Marantz has made it possible for users to adapt the operation of the PMD340 to a wide variety of needs through a Preset menu that controls 21 different operational parameters. As shipped, the PMD340 seems to be set up with radio-station users and DJs in mind. Parameter No. 13 selects the operation that is performed after you load a disc into the tray—PAUSE is the factory default, allowing the immediate start of sound when you depress the PLAY button. But you can easily change this to STOP (probably most useful to consumers) or PLAY.

Another default keyed to radio-station and DJ use is the time display mode when the unit is powered up—the default is Trk REM, or time remaining on the track being played. You can change this to track elapsed time, disc remaining, disc elapsed, or disc total. Most consumers will prefer the track elapsed mode, but pressing the DISPLAY button on the front panel, or the remote, allows you to manually select any time mode, regardless of the Preset menu setting.

One feature I find particularly useful is No. 19, which selects whether or not to display a warning when a skip occurs on a defective CD. If the ALERT function is set to ON, any data loss resulting from the inability to track results in the word INTERRUPTION repeatedly flashing on the screen. The time and track information also freezes where the error occurred, so you can go back to the same point and try again (after cleaning the disc, for example). The player will attempt to continue playing the disc, even though the display remains locked in the ALERT mode. Pressing STOP restores normal operation of the player and display.

You can disable the digital outputs in the Preset menu. When the digital output is active, the pitch bend feature will not work. There are numerous additional Preset menu choices, affecting the auto-cue function, key lock, fade-in and fade-out time, end monitoring, and display language, among others. No. 21 allows you to return all presets to the factory defaults.

REMOTE CONTROL

The PMD340 is a pleasure to operate. The soft pushbuttons have a solid, firm feel and should withstand the day-in, day-out demands of professional use. My only operational quibble is with the optional remote control—the optical transmitter is too directional. You must point the remote control nearly straight at the player in order for it to function.

If you purchase the PMD340, I recommend against buying the Marantz remote. Instead, get yourself a Radio Shack 15-1994 6-in-1 Smart Remote (see sidebar). This "universal" remote has three programming codes for Marantz CD players—0157 works great with the PMD340, duplicating all essential functions of the Marantz remote.

In addition, it has four learning buttons, which you can program for other functions on the Marantz remote. I chose to duplicate the DISPLAY time and the DOOR open/close functions. But if you prefer to program the learning buttons on the Radio Shack remote, you will need to buy the Marantz remote anyway.

The Radio Shack remote is so non-directional that I can point it nearly anywhere in my listening room and still op-

(to page 61)

RADIO SHACK 15-1994 REMOTE

Reviewed by Gary A. Galo

Radio Shack 15-1994 6-in-1 Smart A/V Remote. Available at all Radio Shack stores, www.radioshack.com. $39.95 (four AAA batteries not included).

Those in search of a basic "universal" remote may not need to look any farther than Radio Shack's 15-1994 6-in-1 Smart A/V Remote. This remote will operate up to six different audio and video components, including VCR, television, CD player, and cable box or satellite receiver. The AUX 1 and AUX 2 buttons support other A/V accessories. A PLUG 'n POWER button supports several brands of home-automation equipment. The device doesn't support any DVD players.

Programming the remote is quite simple. Manufacturers' codes for a variety of equipment are provided in the manual, and updated code charts are available on Radio Shack's website. In many cases, several codes are included for each manufacturer. You will need to try each one until you find the one that allows the remote to operate with the model you own. I especially like Radio Shack's generous return policy: If the remote doesn't work with your equipment you can return it for a full refund, as long as you have saved the original box, packing, and manual.

The 15-1994 also has four "learning" buttons, called Smart Keys, which you can program to duplicate functions not supported by the normal array of buttons. In order to "teach" it new codes, point your original remote at the Radio Shack remote, "nose-to-nose." You can program all four buttons for each of the components you use with the remote. In other words, you can program the four buttons for one set of VCR functions, and an entirely different set of CD player functions.

If you depress the backlight button, the keypad lights up in a soft blue tone for five seconds. Thereafter, each time you press any button on the remote, the keypad lights up, and stays lit for about five seconds. This consumes battery power, however. If you remove the batteries, the remote will remember all stored settings until you install new batteries.

The 15-1994 also supports a number of special operational features, including picture-in-picture and favorite channel scan. A sleep function and punch-through feature are also included. Punch-through allows you to control the volume of your television, for example, while using the remote with your VCR, without requiring you to alternately press TV and VCR device keys.

The 15-1994 remote is solidly built, and, with its suede-like finish and soft buttons, has an excellent operational feel (*Photo A*). One nice feature is the lack of directionality. You can point it nearly anywhere in the room, and it will operate your equipment. It makes a nice replacement for any remotes you have that are very directional.

There are many universal remotes on the market far more elaborate (and expensive) than the 15-1994. But for most essential functions on audio and video equipment, it will do quite nicely. I highly recommend this product.

PHOTO A: The Radio Shack 15-1994 6-in-1 Smart A/V Remote. Compatibility with a wide variety of equipment, lack of directionality, and four learning buttons make this inexpensive remote a winner.

Testing Enigma Acoustique's Oremus Loudspeaker

Reviewed by Joseph D'Appolito and Rita and Dennis Colin

I ran a series of impedance, frequency response, and distortion tests on the Oremus loudspeaker from Enigma Acoustique. *Figure 1* is a plot of system impedance magnitude. At low frequencies, the plot displays the double-peaked curve of a vented system.

The impedance minimum of 6.91Ω at 50.5Hz is a good measure of the vented-box resonant frequency. At 212Hz there is a second local minimum impedance of 6.75Ω. The very small difference between these two minima indicates efficient reflex action.

In addition to the two impedance peaks at low frequencies, a third peak occurs at 1.6kHz, where the woofer and tweeter crossover networks interact to form a parallel resonance. Impedance drops to a low of 4.2Ω at 20kHz, and impedance phase lies between +32° and −55° over the full audio range. The Oremus is correctly rated as an 8Ω loudspeaker.

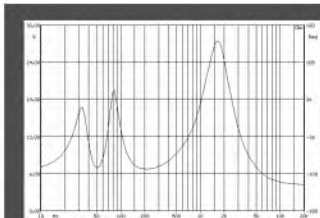


FIGURE 1: Oremus loudspeaker impedance.

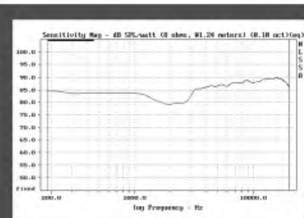


FIGURE 2: Oremus far-field response.

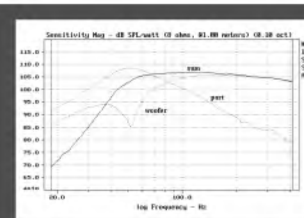


FIGURE 3: Near-field port and woofer responses and their sum.

FREQUENCY RESPONSE

Figure 2 shows the Oremus's far-field frequency response with the microphone placed along the front baffle axial centerline, on a level with the bottom of the tweeter flange and at a distance of 1.24m. This is a quasi-anechoic response.[1] It is valid above 200Hz. The plotted response has been normalized to 1m to obtain system sensitivity.

Figure 3 illustrates near-field woofer and port responses. These responses are summed by the MLSSA system, giving proper weighting to the difference between the woofer and port areas, to obtain the complete low-frequency near-field system response.[1] The dip in woofer response at 50.8Hz is another indication of vented-box resonant frequency and agrees well with the impedance data.

The system near-field response (*Fig. 3*) is spliced to the quasi-anechoic response (*Fig. 2*) at 200Hz to get the full-range response without the use of an anechoic chamber (*Fig. 4*). There are several important features on this plot. First, there is a broad response dip of 5dB between 1 and 3kHz. Above 3kHz, response rises to a peak of 5dB at 15.5kHz. The dip in the critical midrange usually leads to a recessed image; that is, the instruments or soloists appear to be slightly behind the speakers.

The high frequency rise may make the speakers appear overly detailed. Finally, notice the very rapid drop-off of low-frequency response, which falls 26dB in the first octave, to below 50Hz.

The shape of the response curve makes an estimate of sensitivity somewhat problematic. My best estimate is 86dB/3.83V/1m. Relative to this level, the low-frequency −3dB point is 52Hz.

Figure 5 is a plot of system and individual driver responses on an expanded frequency scale. Woofer and tweeter responses overlap by one octave in the crossover region (2–4kHz). Crossover overlap can be an advantage in some designs, but here you see that the phase angle between the drivers is greater than 90° so that their responses actually subtract and cause the response to dip.

SYSTEM STEP RESPONSE

Figure 6 is a plot of system step response, obtained by a numerical integration of the system impulse response. The ideal step response should be a single rapid rise followed by a smooth decay through the 0.00 level.

Figure 6 shows two separate arrivals of acoustic energy. The initial sharper positive spike is the tweeter arrival, followed by the woofer arrival, peaking about 0.25ms later. The drivers are both connected with positive polarity, but the system is not time-coherent.

ABOUT THE AUTHOR

Joseph D'Appolito, regular contributor to *aX* and author of many papers on loudspeaker system design, holds four degrees in electrical and systems engineering, including a Ph.D. Previously, he developed acoustic propagation models and advanced sonar signal processing techniques at an analytical services company. He now runs his own consulting firm specializing in audio, acoustics, and loudspeaker system design. A long time audio enthusiast, he now designs loudspeaker systems for several small companies in the US and Europe.

A better view of this behavior is shown in *Fig. 7*, which is a plot of excess group delay versus frequency referenced to the tweeter's acoustic phase center. This is a plot of delay in milliseconds versus frequency (see reference 1 for a detailed description of excess group delay). In a time-coherent system this plot would be a flat line.

Above 10kHz excess group delay is essentially zero since it is referenced to the tweeter in this frequency range. The curve rises gradually below 10kHz, reaching a peak of 0.3ms at 2kHz and then falling back to about 0.15ms below 700Hz. This plot shows that over its operating frequency range, the woofer is between 0.15 and 0.3ms behind the tweeter depending on frequency.

CUMULATIVE SPECTRAL DECAY

The Oremus cumulative spectral decay (CSD) response is presented in *Fig. 8*. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.06ms increment of time.

The total vertical scale covers a dynamic range of 34dB. Ideally the response should decay to zero instantaneously. Inertia and stored energy that take a finite amount of time to die away, however, characterize real loudspeakers. A prominent ridge parallel to the time axis indicates the presence of strong system resonance.

The first time slice in *Fig. 8* (0.00ms) represents the system frequency response. There is a strong response ridge at 3224Hz coming from the woofer. The ridge extends out to about 1.9ms. There is also a great deal of "hash" in the tweeter decay above 6kHz starting at 0.4ms and lasting out to 1.3ms. The tweeter hash can lend a sense of false "air" to the sound. See the accompanying critique for more about this.

POLAR RESPONSE

Polar response is examined in *Figs. 9–12*. *Figure 9* is a waterfall plot of horizontal polar response in 10° increments from 60° left to 60° right when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00°. For good stereo imaging the off-axis curves should be smooth replicas of the on-axis response with the possible exception of some tweeter rolloff at higher frequencies and larger off-axis angles.

You can see the expected rolloff of tweeter response at higher frequencies and larger off-axis angles. This performance is fairly typical of 19mm tweeters with a recessed dome. *Table 1* lists the off-axis responses at 15kHz and all angles up to ±60°.

Horizontal polar response is broad and uniform. There is some mild reduction in off-axis response in the midrange at larger angles. This is fairly typical of two-way systems. On the whole, however, imaging should be good.

The average response over a 60° horizontal angle (±30°) in the forward direction is shown in *Fig. 10*. In this plot you see that the broad dip between 1 and 3kHz persists and actually worsens somewhat. I suspect this will make the system sound even more recessed than the on-axis response would indicate.

On the other hand, the rise in tweeter response is reduced due to the tweeter's falling off-axis response. This may make the audible effect of the peaking less apparent since the human ear integrates direct and reflected sound when

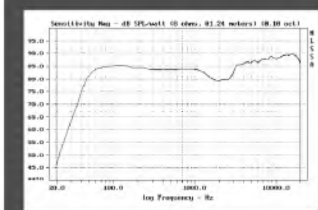


FIGURE 4: Full-range free-standing frequency response.

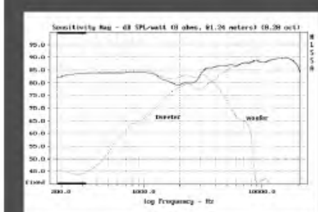


FIGURE 5: System and driver responses.

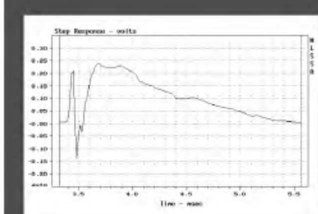


FIGURE 6: Step response.

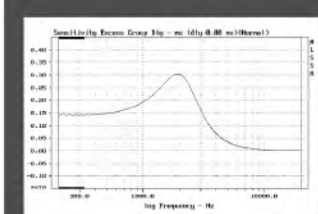


FIGURE 7: Excess group delay.

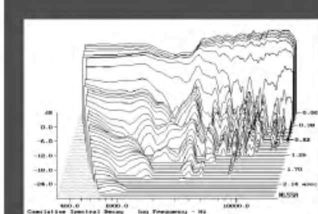


FIGURE 8: Cumulative spectral decay.

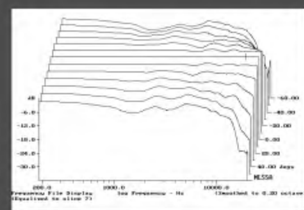


FIGURE 9: Horizontal polar response.

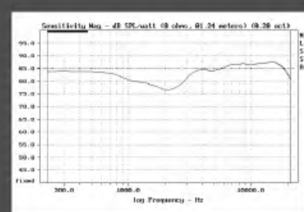


FIGURE 10: Average horizontal response over ±30°.

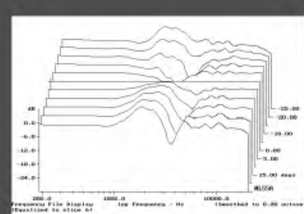


FIGURE 11: Vertical polar response.

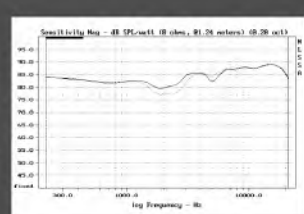


FIGURE 12: Response: on-axis (dotted) and 5° up (solid).

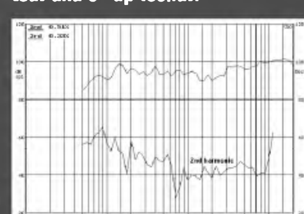


FIGURE 13: Second harmonic distortion.

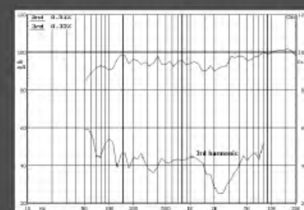


FIGURE 14: Third harmonic distortion.

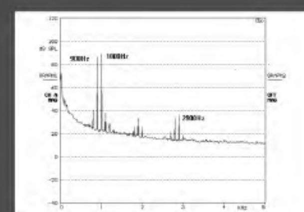


FIGURE 15: Woofer IM distortion.

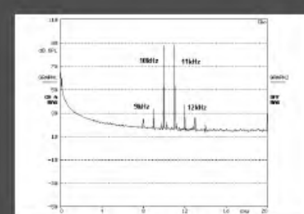


FIGURE 16: Tweeter IM distortion.

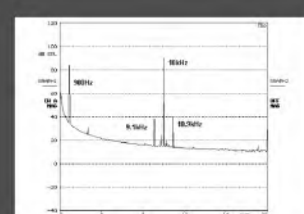


FIGURE 17: Cross IM distortion.

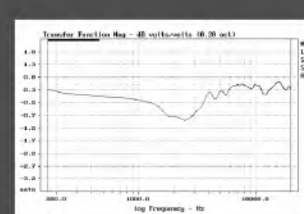


FIGURE 18: Oremus pair match.

Reviewed by Rita and Dennis Colin

EQUIPMENT AND SET-UP

We used the same Nakamichi AV-1 receiver (100W/channel) and Yamaha CDC 755 CD changer (plus turntable and cassette player) that we've become very familiar with after three years, and on which we've heard many speakers, some very good and some not.

THE LISTENING ROOM

Approximately 20′ × 18′ × 8½′ (3000ft^3), the room is moderately damped with stuffed chairs, carpet, and drapes; it is well dispersed by numerous openings and stepped walls. Room response is smooth (for a room) to below 16Hz. Many other speakers sound excellent in this room, including the Swans M1 (review, *SB* 3/99, p. 36).

We placed the Enigmas on stands with tweeters at seated ear height (≈36″), 3′ from the front wall and 4½′ from side walls (11′ apart); the distance to listeners was approximately 12′.

SOURCE MATERIAL

We used the *Hi-Fi News and Record Review* Test Disc III CD (tracks 2, 4, 5, 6, 7, 10, 14), and also played a variety of other material. The Enigmas were broken in with about two hours of moderately loud music.

THE SOUND

I first heard the Enigmas with Joe D'Appolito at his house, when he was trying them out in a new music/home-theater system. With no thought of reviewing these units, Joe and I simply listened a few minutes to Capriccio Italien and Jacintha, a nice-sounding female jazz singer. We were both immediately impressed with a clean sound and particularly clear and extended treble—an unexpected triangle-type percussion instrument surprised us with "in your face" presence, and Jacintha's voice with saxophone accompaniment had captivating "breathiness."

Several weeks later, Ed Dell asked me to review these Enigmas, which Joe then brought to my house. Joe, my wife Rita, and I sat down for an extended evaluation. My first impression, with Jacintha, was almost the same as at Joe's house (my room is more damped than his; I noticed less midrange "presence," but the speaker sounded close enough that I could "tune out" the room effects).

But after a while, something bothered me—the musical and voice tones were clear enough and smoothly extended, bass and treble, but something was "thin" about the sound. Rita didn't like the sound right from the start—"too thin, weak presence, where's the midrange?"

DETAILED SONIC IMPRESSIONS

1. High-frequency detail, such as bell and violin overtones, was very cleanly resolved. It's interesting that the treble sounded over-present (relative to midrange) after extended listening (to me, immediately to Rita), but not bright in the sense of "screechy" (as did the Kirksaeter SL-60s we reviewed in *GA* 5/00).
2. Listening to the *Hi-Fi News* tracks, we noted the sound lacked the normal "fullness of midrange" necessary to present the sensation of "solidity of tone." On "Peter and the Wolf," for example, there wasn't the "lush feeling of a forest" that this composition normally presents on other speakers. On this and almost all the review source material, the individual midrange tones were not distorted, resonant, or uneven in response; however, the general midrange area sounded suppressed.
3. Bass sounded smooth, detailed, and extended (I would estimate) to about 50Hz. String bass instruments were very naturally reproduced.
4. Dynamic range was good for this size speaker.
5. Overall, the sound was frustrating—individually, the bass, midrange up to approximately 2kHz, and treble seemed to be of high quality, but listening full-spectrum, I would guess that the midrange was depressed (especially around 2–3kHz) and the highest frequencies were upward-sloping.
6. About the latter, I find that with a very good tweeter, such as I believe that in the Enigma to be, sloping up the highest frequencies does not cause "coloration," but simply an exaggerated sense of "air" and "detail." In fact, a good test of tweeter smoothness is to try a preamp with tone controls and turn the treble up full. Unnatural, yes, but with a very smooth-response tweeter there will be just emphasis, not coloration, whereas an inferior tweeter will sound like what it is, only much more so.
7. The depressed midrange effect was most noticeable with voices; they sounded too distant and thin. Also lacking "solidness" or "body" were horns and midrange strings.
8. Rita did not like the sound at all; some of her favorite singers—Linda Ronstadt, LeAnn Rimes, Aaron Neville, Julio Iglesias, and Luciano Pavarotti—simply sounded unacceptably lacking in presence and naturalness. On the other hand (or should I say spouse), I could enjoy some of the individual aspects of the speakers, such as the extended treble and bass, and freedom from audible resonances.

But viva la difference; it makes for spice among spouses!

WHAT'S IN A NAME?

Not much, according to Shakespeare. But these Enigmas are appropriately named! They apparently have very good drivers (Joe tells me the tweeter is by Scan-Speak) but a significantly non-flat integrated output. Another "enigma" is the fact that, while I didn't listen extensively to them at Joe's house, I still think they sounded better there. My guess is that the warmer acoustics there filled in some of the midrange, and/or the favorable first impressions and sparkling-clear highs "rang our bells" for a while.

SONIC CHARACTERISTICS RATINGS

Table 1 gives sonic characteristics ratings. To convey some perspective to my choice of ratings within the 0-10 scale, I submit the following attempt at descriptive characterization examples:

9-the best I've ever heard from a speaker regarding the particular sonic characteristic
7-typical for a good-quality speaker
5-mediocre, good for a boom-box
1-don't even think about it!

On this scale, 10 would be the best I believe possible from two-channel stereo; the elusive "perfect enough" speaker.

COMMENTS ON MEASUREMENTS

Seeing the axial response affects my eyes just the way the sound affected my ears. Smooth within large regions, but unbalanced.

Does three-of-a-kind constitute a trend? *The Absolute Sound*, Issue 125, p. 97, contains a review of the Mirage MRM-1, which appears to have a frequency balance similar to the Enigmas and Kirksaeters. Paul Seydor says, "Despite the thought and care that have gone into the design of this speaker, I found the MRM-1 to have one of the weirdest tonal balances of any genuinely high-fidelity speaker I have encountered. There is a broad, relatively deep midrange trough that starts around 200–300Hz and extends to between 5–6kHz, where it starts a gradual rise."

The specifics of the Enigma's response are different, but the general shaping appears similar. And another quote from Mr. Seydor (just before the above) may help explain the "enigma" of more favorable impressions initially than long-term: "At the end of

SONIC CHARACTERISTICS RATINGS FOR THE OREMUS ENIGMA

		1	2	3	4	5	6	7	8	9	10
Presence (overall realism)	RC					X					
	DC						X				
Freedom from Distortion	RC			X							
	DC								X		
Frequency Response Smoothness	RC			X							
	DC							X			
Low-Mid-High Balance	RC			X							
	DC			X							
Treble Quality	RC		X								
	DC								X		
Midrange Quality	RC			X							
	DC				X						
Bass Quality	RC					X					
	DC								X		
Bass Extension	RC					X					
	DC							X			
Immediacy and Transient Response	RC					X					
	DC								X		
Image Focus	RC			X							
	DC							X			
Stereo Soundstage Realism	RC		X								
	DC				X						
Ambience	RC		X								
	DC							X			

the day, the overall tonal balance determines the sound of the speaker."

Maybe for me, but Rita noticed it immediately. I think I (and dare I generalize to males?) may at first listen too analytically, thereby being overly impressionable by specific "bells and whistles" (bells here being a good example; I liked their sound on the Enigmas). But after a while, this focused analysis-mode becomes tiring; then you stop listening for details and hear, as Mr. Seydor said, the overall tonal balance.

One question: How can a speaker have "one of the weirdest tonal balances" and yet be a "genuinely hi-fi speaker?" Does this mean it's still better than any "non-hi-fi" speaker; the "weirdness" is "acceptably mild?" Sorry, but I find this concept mildly weird!

The Enigmas, however, didn't sound "weird" to me; they sounded like two very good drivers with a crossover that produced a suppressed midrange and upward-sloping treble. Therefore, I suggest these speakers would be a good project for those who enjoy crossover design and tweaking. On the other hand, you might like the stock speaker, particularly in rooms with strong midrange reinforcement and/or high-frequency absorption.

judging the overall spectral balance of a loudspeaker.

Notwithstanding these problems, the horizontal coverage is quite good. There will be only small changes in spectral balance with horizontal position. Image stability should be good.

Figure 11 is the waterfall plot of vertical polar response. Responses are shown in 5° increments from 25° below (–25°) the tweeter axis to 25° above it. In all cases response changes significantly with vertical off-axis displacement. I suspect this is a direct consequence of the driver crossover overlap and inter-driver phase angle. Interestingly, response at 5° up is smoother than the on-axis response (*Fig. 12*).

HARMONIC DISTORTION

I conducted harmonic distortion tests at an average SPL of 90dB at 1m. Ideally, harmonic distortion tests should be run in an anechoic environment. In practice, it is important to minimize reflections at the microphone during these tests. Out-of-phase reflections can produce false readings by reducing the level of the fundamental while boosting the amplitude of the harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5m from the loudspeaker.

Figures 13 and *14* show second and third harmonic distortion levels in dB SPL versus frequency plotted in 1/6-octave steps. System frequency response is also plotted on these figures. Worst-case low-frequency second harmonic distortion is 4.5% at 90Hz. The third harmonic hits 3.8% at 56Hz. All system harmonic distortion is well below 1% above 150Hz. Tweeter harmonic distortion is below 0.4% at all frequencies. This is good performance for a small monitor speaker.

A note on testing: The Enigma Acoustique's Oremus loudspeaker was tested in the laboratories of Audio and Acoustics, Ltd. Measurements were made with the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems using an ACO 7012 ½" laboratory-grade condenser microphone with a custom-designed wide-band, low-noise preamp. The polar response tests were performed with a computer-controlled OUTLINE turntable on loan from the Old Colony Division of Audio Amateur Corporation.

INTERMODULATION DISTORTION

Next I measured intermodulation distortion. In this test two nearby frequencies are input to the speaker. Intermodulation distortion produces output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion. Properly interpreted, intermodulation distortion can reveal a great deal about speaker performance.

Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

TABLE 1
HORIZONTAL POLAR RESPONSE AT 15KHZ

ANGLE	RESPONSE(DB)
10	–0.6
20	–2.0
30	–4.3
40	–7.3
50	–11.5
60	–14.2

I first examined woofer intermods by inputting 900Hz and 1kHz signals at equal levels. These frequencies should appear predominantly in the woofer output. Total SPL with the two signals was adjusted to 85dB at 1m. The Oremus system output spectrum for this test is shown in *Fig. 15*. The two largest spectral lines represent the input signals.

IM products appear at 800, 1100, 1900, 2800, and 2900Hz. The largest distortion product is second order at 800Hz. It is 51.4dB below the main output, which is equivalent to 0.27% distortion. Total woofer IM is 0.4%. This is average for small monitor speakers I have tested.

I measured tweeter intermods with a 10 and 11kHz input pair also adjusted to produce 85dB SPL at 1m (*Fig. 16*). IM products can be seen at 8, 9, 12, and 13kHz. The largest intermods are at 9 and 12kHz. Total tweeter IM distortion is 0.23%. This is better than average for a small monitor speaker.

The last IM test examines cross intermodulation between the woofer and tweeter using frequencies of 900Hz and 10kHz. (A 1kHz signal would produce intermods that fall on harmonic distortion lines, confusing the results.) Ideally, the crossover should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. This spectrum resulting from this test is shown in *Fig. 17*.

The principal IM products fall at 9.1 and 10.9kHz. Total distortion is 0.34%. This is somewhat on the high side compared to other systems I have tested. It may be due to the choice of crossover slopes.

PAIR MATCHING

Two samples of the Oremus loudspeaker were available for testing. The two units were arbitrarily labeled #1 and #2. All of the tests described so far were conducted with #1. One question of interest is how well the two samples match?

Frequency response matching is shown in *Fig. 18*, which is a plot of the response difference between the #1 and #2 samples. The two systems match within +0.62 and –0.89dB. This level of match should guarantee good image stability over the full frequency range. ❖

REFERENCE

1. J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Corporation, Peterborough, NH, 1998.

Marantz CD
from page 56

erate the Marantz player. Although the manual lists Marantz RC-5 remote control codes for operating the variable-pitch feature, the Marantz remote does not support this.

OPERATION

The new Marantz players have an intelligent keypad algorithm that eliminates the need for those clumsy "greater than 10" buttons found on most CD players. On a CD containing less than ten tracks, if you press "1" the player will immediately search for and play track 1. If the disc has more than ten tracks, when you press "1" the player will hold position for two seconds and wait for additional input—a "2" if track 12 is desired, for example. If there is no additional input, the player will begin playing track 1 after two seconds.

The players also have a programming function that allows you to program up to 20 tracks on a disc, in any order you choose. A cue function allows you to begin playback at any spot you desire in a given track.

Sonically, I found the PMD340 to be an improvement over the PMD320. The new player yields a cleaner, more transparent, and smoother sound than its predecessor. I suspect that this is due, in part, to the Burr-Brown PMC1710U filter/DAC chip. The improved sound may also be attributable to lower jitter.

Though a bit below audiophile standards for stand-alone players in this price range, the professional user is likely to find the PMD340 a sonically satisfying performer. Where audiophiles will find the PMD340 useful is as a transport. Used via its S/PDIF output, with a high-quality outboard D/A converter, the PMD340 can serve as the front end of a fine CD playback system.

I compared the performance of the PMD340 with both S/PDIF and AES/EBU digital interconnects, using The Parts Connection DAC-3 and DH Labs interconnect cables. Strangely, I found the sonic performance of the AES/EBU connection inferior to the S/PDIF—the detail, resolution, and soundstaging are better with the S/PDIF coax connection. I recommend the S/PDIF connection if you use this player as a transport in a setup involving critical listening.

Among currently available CD players with variable pitch, the PMD340 is now a first choice for this application, either when used as a transport or a stand-alone player. The PMD340 is a well-built, solid performer, which should easily satisfy the demands of the professional user, and is highly recommended.

I should also mention that at The Crane School of Music/SUNY Potsdam, we purchased the PMD330 (the "bottom" model in this new series) for use as a transport with an NAD118 digital preamplifier. This is part of a new stereo system I assembled for our newly renovated lecture and recital hall. We are extremely pleased with the performance of the PMD330 in this application, and I expect it to hold up well for years to come.

Depending on the features you desire, and the price you wish to pay, the two lower models in the new Marantz line also deserve consideration. ❖

Speaker-Cable Impedance Concerns—Real or Imaginary?

Wire is drawn—so are conclusions about the audibility of cable imperfections. **By Dennis Colin**

I measured extensively, from DC to 50MHz, a longer-than-normal extent (32.5′) of very inexpensive speaker cable (Radio Shack 2 × 16 gauge). I chose this extra-long and ordinary wire to set a conservative upper limit on parasitic effects. Measured were R, L, and C; 20kHz loss, phase shift, and delay; RF response to 50MHz; characteristic, or pulse impedance Z_o, transit time; and propagation velocity. Also, for entertainment, I measured some time-domain reflectometry (TDR).

Questionable claims, such as the audibility of a few copper oxide molecules or dielectric distortion, I address with calculations and measurement reports. Finally, I realize that our hearing, with a range greater than 120dB and the ability to resolve more than 4000 frequency components in about one cycle of each, is the ultimate judge. So I anticipate seeing some careful, extensive, and unbiased blind-test comparisons.

FREQUENCY AND PULSE RESPONSE

Using the setup of *Fig. 1*, *Photos 1–6* show the cable's square-wave responses with a 4.4Ω load resistor. The 200kHz response was made to calculate rolloff and delay dispersion, while I included the 1MHz response for curiosity's sake. The 20kHz and 200kHz responses (note different amplitude scales) are mostly an exponential rise (first-order HF rolloff) except for a small initial step, that is due to the small (≈ 50nH) inductance in the load resistor (which compensates some of the cable rolloff).

With the 20kHz square wave, of course, all its harmonic components are beyond audibility except (for those with exceptional hearing) the fundamental. The cable is 3dB down at about

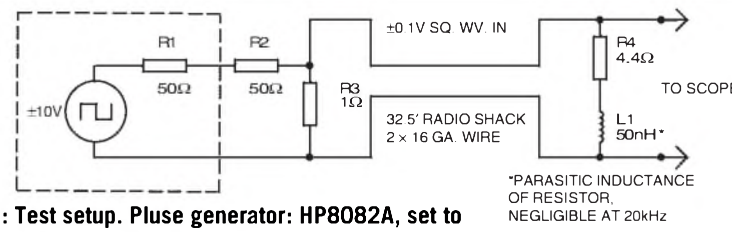
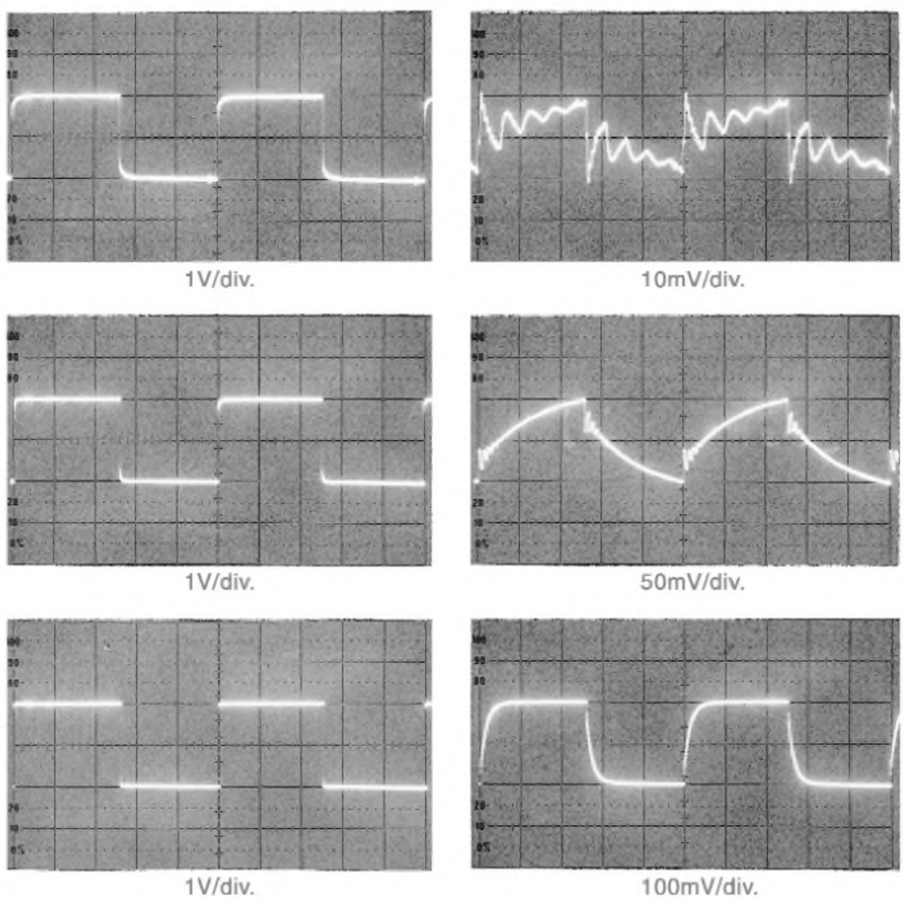


FIGURE 1: Test setup. Pluse generator: HP8082A, set to square wave, 5ns rise and fall time; scope: HP1725A, 1MΩ Z_{in}, differential input with Radio Shack cable.



PHOTOS 1–6: Square-wave responses of 32.5′ 2 × 16 gauge wire with 4.4Ω load. 1MHz, 200ns/division (top); 200kHz, 1μs/division (middle); 20kHz 10μs/division (bottom).

ABOUT THE AUTHOR

Dennis P. Colin graduated with a BSEE from the University of Lowell (MA), and is currently an Analog Circuit Design Consultant for microwave radios. Previously a band keyboardist and recording engineer, he has been published in the *Journal of the Audio Engineering Society*. Colin has demonstrated the audibility of phase distortion at Boston Audio Society and designed the "Omni-Focus" speaker bi-polar coincidental with phase-linear first-order crossover, ARP 2600 analog music synthesizer, 1kW bi-amp and PWM supply at A/D/S, and Class D amps.

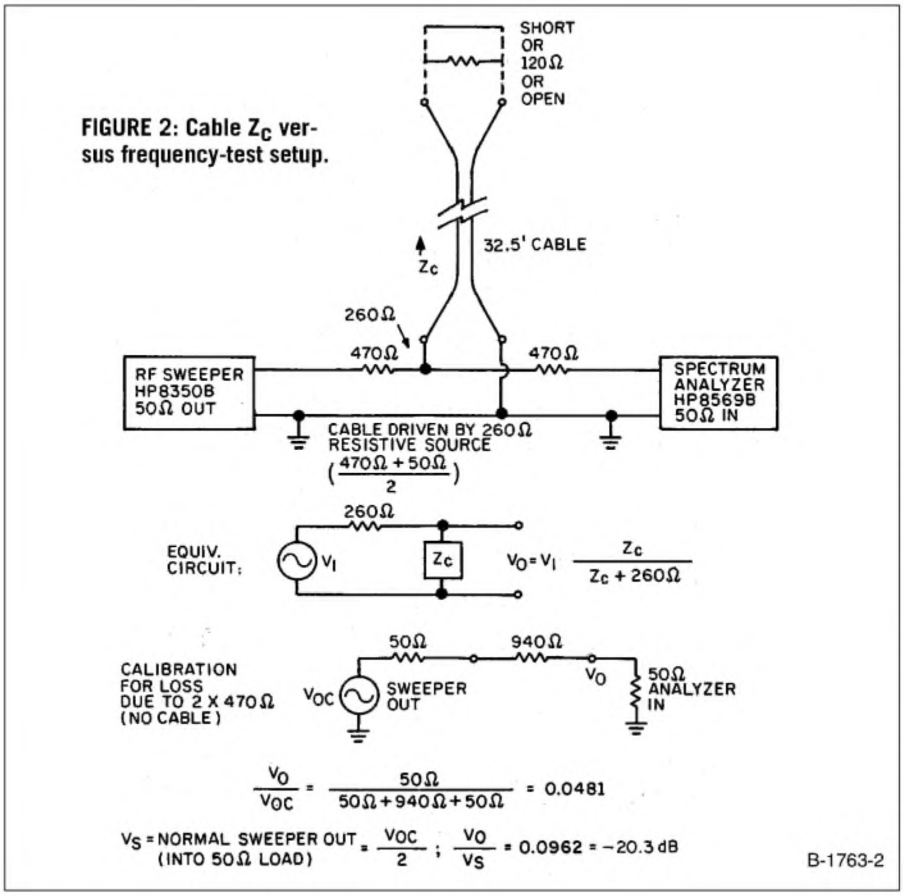


FIGURE 2: Cable Z_C versus frequency-test setup.

100kHz, with nearly first-order rolloff, corresponding to the observed 1.6µs rise-time constant. From this, the amplitude and phase errors at 20kHz are about −0.2dB and −12°, and the maximum time smear, DC −20kHz, is 1.6µs. Can you hear this? Very unlikely; moving your head ⅛″ can cause greater deviations.

DISCRETE PARASITICS

A 20kHz wavelength, while only 0.68″ in a sound wave, is 9.31 miles as an electromagnetic wave in space. In a typical cable, it's about seven miles. So obviously wave-propagation disturbances are negligible in an 8′, 32′, or even 100′ piece of speaker wire.

You can therefore characterize the cable in terms of series resistance (R), shunt-leakage conductance (G), series inductance (L), and shunt capacitance (C). Any nondefective cable has next-to-nonexistent leakage G (1000s of megohms). But R, L, and C can be significant; R causes signal loss, and possibly audible response changes due to speaker-load variations.

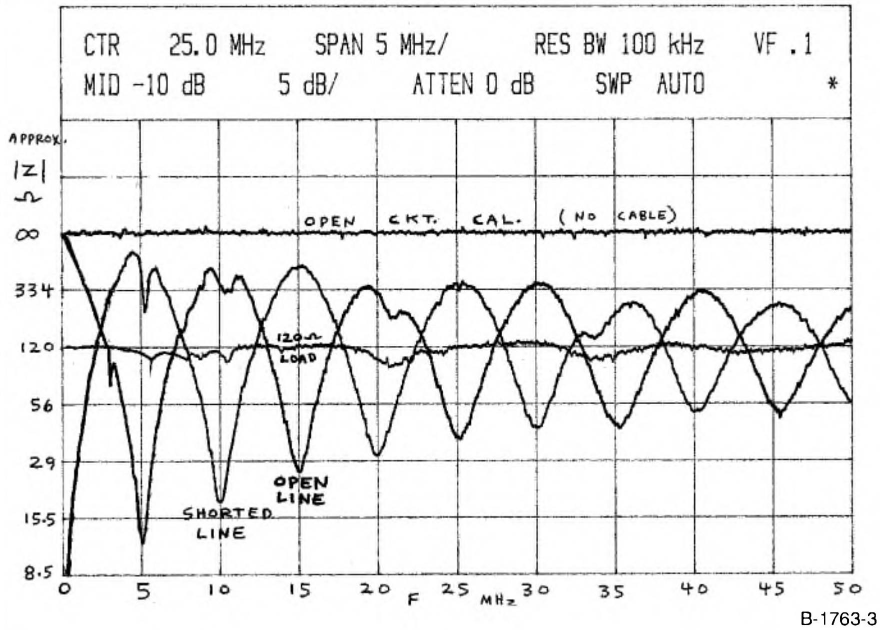
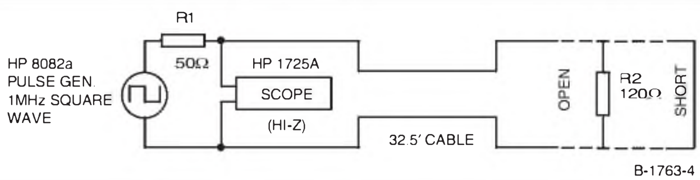


FIGURE 3: 32.5′ Radio Shack 2 × 16 gauge speaker cable impedance versus frequency.

FIGURE 4: Time-domain reflectometry.



L and C, of course, form a low-pass filter, except when a cable is terminated in its Z_o, which is not the case here. The Radio Shack wire's Z_o is about 120Ω, typical for parallel twin-lead. A short (<<λ/4) transmission line loaded by less than its Z_o (the case here) has a frequency rolloff mostly determined by L; here the values correspond to the observed f_3 of 100kHz. (I should mention that Joe D'Appolito says that sufficient L and C could contribute to the instability of some amplifiers.) The measured parasitics are: R = 0.159Ω, L = 6.51μH, and C = 455pF. These have the following effects with a 4Ω load resistance:

1. R causes a flat-spectrum loss of 0.34dB.
2. L causes a 3dB rolloff frequency (re DC) of 98kHz; at 20kHz, response is down 0.18dB and phase shift is −11.6°. Maximum time smear is 1.6μs (audibility threshold reported to be 100μs at best).
3. A shunt capacitance of 455pF has an impedance at 20kHz of 17.5kΩ. In paral-

lel with a (worst-case) speaker load of 16Ω, this causes a rolloff (at 20kHz) of 0.0000004dB, and a phase shift of −0.05°. And even if this capacitance were 100% nonlinear, the maximum distortion it could cause ("dielectric distortion") would be 0.00004%.

And remember, this is 32.5′ of cheap 16-gauge wire!

PHOTO 7: Pulse generator, no load.

PHOTO 8: Cable, 120Ω at end.

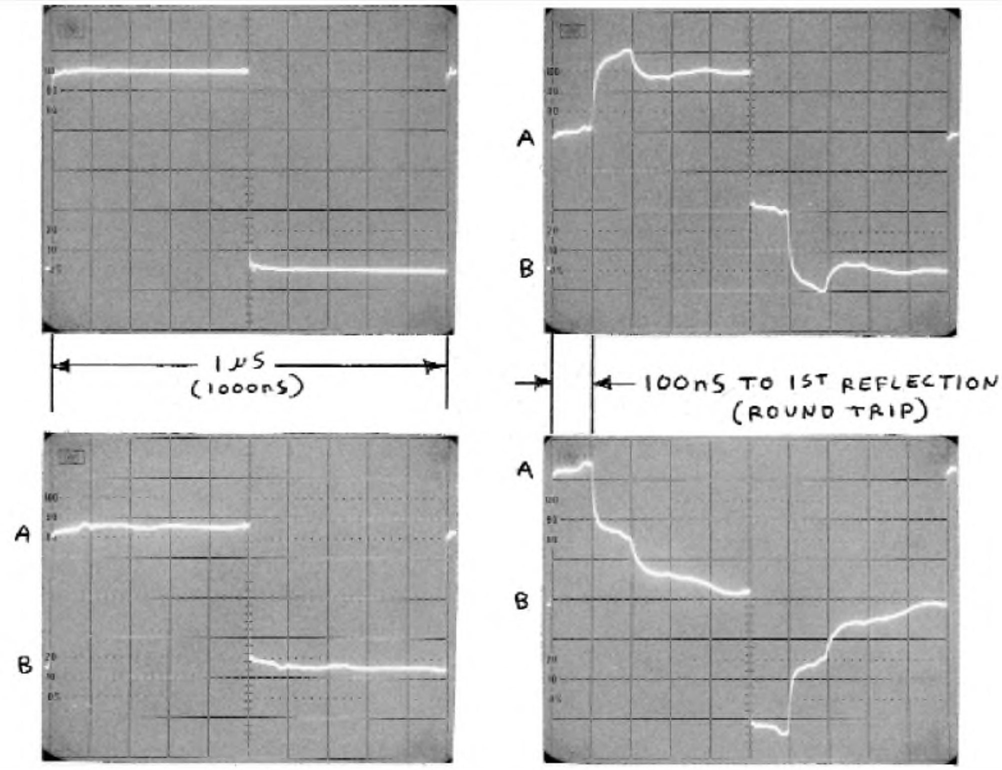


PHOTO 9: Cable, open end.

PHOTO 10: Cable, shorted end.

RF RESPONSE AND STANDING-WAVE RESONANCES

I used the setup of *Fig. 2* to measure cable input impedance from <1MHz to 50MHz, with the cable terminated in a short, an open, and a 120Ω resistor. 120Ω is the calculated value of Z_o, from $Z_o = (L/C)^{1/2}$, where L and C are per-unit length, provided that this length is much shorter than a wavelength, which is true of 32.5′ at and below 20kHz.

Figure 3 shows the results. Note the fairly constant 120Ω input impedance when the cable load is 120Ω; this confirms that value of Z_o. With the far end of the cable shorted or opened, you see the classic resonances at multiples of the quarter-wave frequency, about 5MHz.

Of significance to audio is the absence of resonance below the minor one around 3MHz, even with open or shorted termination.

TIME-DOMAIN REFLECTOMETRY (TDR)

Figure 4 shows the cable input voltage when driven by a near-perfect 1MHz square-wave source with resistive 50Ω Z_o (*Photo 7*, with cable disconnected). *Photo 8* is with the cable, which has a 120Ω resistor load. This is the previously determined Z_o value. Note the good square-wave response, which is due to each input-voltage step sending itself down the cable (as with a guitar string); 50n later, it reaches the far end 32.5′ away (flying at 66% the speed of light). Only with a load equal to Z_o will there be no reflection back; all the energy in those zooming E and M fields is ideally absorbed.

But any other load, and especially an open or a short, reflects some or all of it back; that's what causes the "comb filter" resonances in *Fig. 3*. *Photos 9* and *10* show this return pulse. But note how they both have a 100n initial plateau (A), the same voltage jump from (B) as with the 120Ω load. This means that the impedance seen by the pulse generator is the same with any cable load for the first 100ns following a signal change.

100ns is the round-trip time, twice the 50ns cable-transit time. This return pulse is in phase with the input signal when reflected off an open end, and is out of phase with a short-circuited end, as shown by the opposite-step polarities after 100ns in *Photos 9* and *10*. Some of this also re-reflects off the 50Ω (not Z_o) driving source, causing a decaying pulse train.

Also note that, prior to the next signal step (500ns from photo start), the voltages in *Photos 9* and *10* have just about settled (I chose the 1MHz frequency to give that enough time). In *Photo 9* (open), it settles to the no-load pulse-generator output—so it takes about 500ns after the driving input stops changing for the cable input to appear like its load, an open circuit. The same with *Photo 10*, except it decays to 0V as the input eventually resembles the shorted output.

TDR finds its application in locating shorts, opens, or other defects within a cable. Here, of course, it's not applicable in determining audio-cable quality, unless you can hear a 5MHz resonance! But I explored this both out of curiosity (can I use cheap speaker cable for RF antennas?) and to see the relation between time and frequency responses of transmission lines, which of course include vibrating strings and air columns.

It's interesting to note that the reflections seen in *Photo 4* (cable output with 1MHz square wave input) are less damped (more cycles seen) than in *Photos 9* and *10*. That's because *both* cable ends are highly mismatched (1Ω source, 4.4Ω load, versus 120Ω Z_o). For those frequencies high enough to resonate, the term VSWR (voltage standing-wave ratio) is used to express impedance mismatch, or reflectivity thereof. But at 20kHz, with a seven-mile wavelength, there are no waves on a 32.5′ cable.

Finally, I would like to address three concerns sometimes claimed about cables:

Nonlinear Distortion?

Some propose the existence of audible distortion due to (1) imperfect atomic structure in conductors and (2) dielectric nonlinearities.

In answer to that, consider that even at 1GHz in cheap RF cables, distortion is between 140–160dB down; –140dB is a whopping 0.00001%! So at 20kHz? Probably lower than thermal noise.

Time Smear?

32′ of Radio Shack wire has about a 1.6μs time smear, DC –20kHz. Reported values of the audibility threshold are about 100–1000μs.

Dynamic Range Compression?

Yes; maybe 0.1dB at 10,000W.

A VERY INTERESTING TEST

Several decades ago, *Audio* magazine conducted a listening test comparing three speaker cables:

1. intentionally too-thin wire, 22 gauge or so;
2. ordinary cable, 14 or 16 gauge, I think; and
3. the then-newly introduced Monster Cable™, which was basically just very heavy-gauge wire for that time.

Two sets of tests were "conducted": First, with the listeners knowing which cable they were listening through, and then a blind test. Between the too-thin 22 gauge and the ordinary 14 or 16 gauge, there was a significant statistical preference for the heavier wire, even in the blind test.

Comparing the "ordinary" heavier wire with the Monster Cable, there was a preference for the latter, when the audience knew which cable was used. But with the blind testing, it was statistically even.

CONCLUSIONS

I'm not saying you can't hear the improvement of a high-quality dedicated speaker cable over Radio Shack wire. But if you can, verified in a real test, the reasons are likely to be:

1. Very small signal loss; 0.3dB with this extra-long cheap cable.
2. Very small but audible response deviations due to cable inductance or capacitance interacting with speaker impedance or amplifier stability. Bear in mind that (1) you can, under ideal controlled conditions, hear perhaps 0.1dB changes; however, moving your head 1″ makes a significant change; and (2) the slightest audible frequency-response change can sound like changes in distortion and dynamic range.
3. Interference pickup; if you're in an electromagnetically noisy area, an expensively shielded cable may be needed. This is unlikely, however, with speaker cables; perhaps it's so for line levels. ❖

Confessions of a Former Driver Designer

Here's the inside scoop on a day in the life of a driver designer for a high-volume manufacturer. **By Perry Sink**

The audio construction hobby is driven by the desire to delve into the nitty-gritty of audio systems to control those subtle, underlying details. If you build amplifiers, you probably appreciate the opportunity to design the whole thing from scratch. Most folks who go to the trouble of building an amp aren't going to buy just a high-power IC and slap it on a heatsink. You select every component in order to maximize performance at every level.

Building speakers is a bit different, though. Most of us don't have the luxury of building drivers from the ground up. The closest we get is simply being able to choose from many, many drivers, and hope to pick one that matches our priorities very closely. However, then we have complete control over every other detail—enclosures, crossovers, and so forth.

As a driver designer at Jensen in Chicago from 1993-1995, I had the opportunity to experience a different perspective, which I'm going to share with you. But first, a little background...

EARLY AGE

I was 12 years old when I acquired what would prove to be a "permanent" urge for a better stereo system. My financial resources, based on allowances, odd jobs, and occasional stints as a substitute paperboy, were extremely thin, and after visiting all the stereo shops in Lincoln, Neb., my prospects looked a little bleak for anything beyond the most basic system.

But Radio Shack offered a book on building speakers, which I bought. And then I bumped into a guy at one of these shops, Steve Peters, who was apparently willing to give up a sale (my eye was on his Boston Acoustics A60s) in order to teach me the ropes of building my own system. Eighty bucks later, I had a Peerless woofer and Tonegen tweeter from the *McGee Radio Catalog*, some crossover parts, and the laborious job of cutting particleboard with a jigsaw.

Those speakers actually sounded fairly decent, and I was hooked. I built another pair (which I sold) and continued through high school and college picking up extra bucks building speakers for friends. When I was a senior in high school, I secured an arrangement with a local dealer, selling my brand in his store. A year later I was studying Electrical Engineering at the University of Nebraska, and in '92 I left with my degree, a motional feedback subwoofer as my senior project, and a focus on control systems and communications.

Jensen, then in Schiller Park, IL, offered me a job, so I moved to Chicago and started working in the OEM division, which services automotives. During my time at Jensen, I was involved in designs for the '94 Ford Probe, '95 Jeep Cherokee, '95 Acura Vigor and Integra, '96 Honda Civic, and '97 Chrysler Cirrus and Stratus. I also worked on a "skunkworks" project to develop a low-cost motional feedback subwoofer. (I left before that project saw the light of day, but it was fairly successful, and if someone wishes to pursue it, it's still lurking in my files!)

ABOUT THE AUTHOR
Perry Sink is the National Sales & Marketing Manager for Synergetic, an industrial networking hardware/software development firm, and also does a limited amount of outside consulting in direct-response advertising and marketing development. His e-mail address is ebiz@futurezone.com.

OEM VS. AFTERMARKET

There were two divisions of Jensen's engineering department: OEM, which built speakers for new cars; and Aftermarket, which was responsible for products sold at car stereo dealers, Wal-Mart, and elsewhere.

During the interview, they asked me which department I was interested in because they had one opening in each. I really didn't know, but somehow I wound up in OEM, and in retrospect, I'm glad I did.

Here's why: In aftermarket, some product manager, whose previous job was in the marketing department at Whirlpool or some such place, gave you a pair of speakers from Sony or Pioneer and said, "Make our next model sound just like these, because they're outselling ours. And they must look like this drawing right here. And you need to do it with these parts."

Basically, in aftermarket, I would have had both hands tied behind my back. Guys who worked in aftermarket usually had voodoo dolls of marketing people stuck full of pins, and they tended to drink a lot. Dilbert clippings on their bulletin boards carried the sentiment, "We encourage a healthy tension between engineering and marketing." They enjoyed going to car stereo shows, though, so they could sit in a conversion van with 14 subwoofers and 3000W of power, watching girls in skimpy bikinis and feeling their pantlegs flap in the bass.

HIDDEN OPPORTUNITY

Ironically, OEM was almost the opposite. You'd think that Honda or Chrysler would have bound us in a merciless strait jacket of specifications (actually, Ford did), but mainly the design just needed to be cheap. By cheap I

TABLE 1		
PART	PER PIECE COST	TOOLING COST
Cone	0.25–0.70	5k–15k
Surround	0.10–0.25	10k–20k
Spider	0.20–0.30	2k–5k
Back plate	0.15–0.20	50k–100k
Front plate	0.10–0.20	25k–50k
Whizzer	0.10–0.25	20k–30k
Magnet	0.25–0.75	50k
Voice coil	0.25–0.70	1k–5k
Basket	0.40–0.80	50k–150k

mean shaving pennies to the tune of $3.75 per speaker, and 0.01 cents precision on pricing. While it needed to fit in a particular mounting hole, and have a minimum sensitivity (88.5dB, or something like that), essentially I had a free hand, especially on the "basic" systems (as opposed to the premium systems).

My mission: To make four $3.75 speakers sound good in a Honda Civic, using the car's Fujitsu Ten radio and its pre-defined mounting holes and locations. To a die-hard audiophile purist, this seems a pretty dreary proposition. But to a guy like me who loves a challenge, looks at things in terms of "doing the most with what you've got to work with," and is proud if the results come out sounding "surprisingly good," then it's a lot of fun.

The way I figure, any idiot should be able to develop a nice-sounding system if he's got 5000 bucks and Krell amplifiers. But the mark of a skilled engineer is finding combinations of materials, budgets, and design concepts that deliver excellent results with scarce resources—i.e., five bucks instead of 5000.

Most of these designs were based on full-range speakers, not two-ways or three-ways. I was intrigued by that, too. When I was 17, I read E.J. Jordan's manual on driver design, which explains the rationale behind his 50mm module, a driver that covers the range of 100Hz to 20kHz. This manual introduces clear thinking and rationale for full-range driver design, and provided great background for me.

When I showed up for work on February 1, 1993, with my many theoretical ideas about how drivers work—or should work—I was prepared to put all my theories to the test of real measurements and practical economics. To the hobbyist, driver design is a bit mystical, but, after building hundreds of drivers from the back plate on up, measuring them, and listening to them, there's not much that's really "mystical" to me.

Not that I know everything, by any means. But the reality is that real drivers are made of real materials, which have known, measurable properties, and there's a reason for everything you hear, whether real or imagined.

PARTS IS PARTS

At Jensen we had a sample parts room—pegged walls and shelves of back plates, front plates, pole pieces, magnets, baskets, cones, spiders, voice coils, dustcaps, whizzer cones, terminals, lead wire, dozens of versions, and variations of each different part. In the lab we had a dozen different kinds of adhesive and a staking machine for securing front plates to housings.

You could walk into the parts room, grab a handful of parts, and have a working speaker in 30 minutes. Just down the hall was an anechoic chamber and a chart recorder to plot frequency response, impedance, and distortion. Another room had an automated Thiele/Small parameter measurement setup.

Then, of course, you could attach your new speaker to one of our handy-dandy, pre-cut templates, which you could mount in the wall of the listening room, and listen to your new speaker. You could match levels with an SPL meter and compare it to other speakers—all behind an acoustically transparent curtain, if you so desired. You can learn a lot about speaker design, making a dozen variations of a single design and measuring them all—and I did.

One of my most basic observations was that even though you might develop a "mystical" perception about a certain name brand, design preference, or material, when you get right down to it, it's all physics. Voice coils have very specific mechanical and electrical parameters, and quality-control characteristics. Cones come in a variety of profiles, material combinations, and mating surround materials and geometries. You can wax eloquent about the musical qualities and everything else, but it all comes from a very tangible mechanical device, which responds to electrical input in accordance to well-known physical laws.

INSIDE THE OEM SPEAKER BUSINESS

Inside the speaker biz, speaker components come in two categories: software (cones, spiders, voice coils) and hardware (magnets, baskets, back plates). Most of my work was with mass-produced car speakers (100,000 to 500,000 units per year was typical), especially 4–6″ round and 6 × 9s. Cost was always an issue, as was tooling. As a designer, I was strongly discouraged from "tooling" a new part—too expensive, time consuming, and prone to failure.

Table 1 shows the various parts of a typical small, large-volume production speaker and a rough estimate of parts cost, as well as tooling costs.

The total cost for a cheap, high-volume speaker like this is $2–3. Most parts could be used "off the shelf" without additional tooling, so add in the labor to put the speaker together (at Jensen it was 5¢ per five-second operation), and you can sell them for $4 to $7. When I was working there, Jensen made 50,000–70,000 speakers per day, mostly with unskilled workers and relatively little automation.

How good were these speakers? Some were not too bad, in my humble opinion. We made full-range 6 × 9s with whizzer cones that were ±3dB from 55Hz to 17kHz on-axis. They sold for about six bucks each. If you take the stock speakers out of your Ford Escort and measure them, you might be surprised at what you see.

WHIZZER CONES

You can replace the dustcap of a 5″ speaker featuring a smooth response from 80Hz to 6kHz with a whizzer cone to increase the response out to 15 or 20kHz. It's almost as good as adding a tweeter, but not quite. A whizzer attaches directly to the voice coil, and because of its smaller size and (usually) higher density, it responds to frequencies that the woofer can't cover.

Whizzers are highly resonant, with most of the damping, I believe, coming from the air itself. Whereas tweeters can often have a smooth response curve, whizzers always have some (often a lot of) jaggedness, and tend to sound "phasey." You just don't get the clarity you do with a tweeter. Also, the polar distribution looks like a bunch of bananas. But it's much better than no whizzer at all, and the cost increase is perhaps only 10 or 15¢.

At Jensen, we had sophisticated tools like Finite Element Analysis at our disposal, and for some designs—especially those requiring that we tool a new part—they were used quite extensively. To my knowledge, nobody has successfully "modeled" the behavior of a whizzer cone. The tendency is to just fiddle with the various sizes and shapes of a design until something seems to work.

SILK PURSE

I always thought that someone should place a pair of those six-dollar 6 × 9s, which were really more accurate than most people might guess, in a set of beautiful, heavy, rosewood, wool-filled transmission line cabinets with elegant cloth grilles, and market them as minimalist, exotic high-end speakers for $3,500 a pair. No crossovers or any of their associated evils! Imagine the harmonic integrity. Match them up with a single-ended triode amp and some really expensive cables, and you would be in audio purist heaven. (Well, there's a business idea for somebody! I promise you it has an excellent chance of succeeding, too.)

A DAY IN THE LIFE

The job certainly entailed fun stuff, such as brainstorming concepts and listening to test systems in vehicles. But the majority of my time was spent building prototypes and batches of samples, and doing lots of testing and statistical analysis.

One of the benefits of working at a high-volume OEM is that other companies are happy to send you samples. They pay attention when you call them and explain that your company makes 70,000 speakers a day.

They know who you are—especially the cone, spider and voice-coil manufacturers—so it's perfectly normal to call up their samples guy every other week to rattle off some stuff you need. Within a week or so you receive some custom-made parts, possibly with a whole range of variables in the sample lot. (One of the mechanical engineers was fired when someone figured out that the samples he was requesting from a machine shop were actually engine parts for his motorcycle.)

HIGH-END AUDIO MAGAZINE BASHING

Most of the guys I worked with had the same geeky attraction to speakers that I did, so we spent plenty of time discussing audio theories, lore, and strange inventions, and, of course, bashing *Stereophile* and *Absolute Sound*.

As I said before, there's something about designing drivers that teaches you "harsh reality." In the end, there is no mysticism. There's only physics and the attempt to relate it to the subjective experience of listening to music. Some factors make a lot of difference, while others make only a little difference.

Here's my quick-and-dirty list of things you can change in your system to make it sound different, and hopefully better, in relative order of how much they will actually influence the sound:

1. Listening room
2. Crossover design and cabinetry
3. Drivers
4. How much clean power you have available
5. Inductors (replacing ones that saturate with ones that don't)
6. Signal sources, amps, preamps, D/As, and so forth
7. Speaker stands
8. Capacitors
9. Cables

As I worked on audio systems, it became clear to me that items on the top of the list are difficult to change, require genuine insight into problems that most people haven't studied, and are difficult to "sell" in the sense of swiping somebody's American Express Gold Card and providing immediate gratification (with the notable exception of simply buying new speakers off the showroom floor).

However, the items on the bottom of the list are easy to swap and have the highest profit margins, even though they hardly make any difference at all. So these receive the most attention in the press. Somebody will always be happy to lighten your wallet with something that makes more of an imaginary than real difference, simply because it's profitable.

So when our engineers (many of whom own excellent home audio systems) would attend a show such as CES, they would spend most of their time at the high-end portion, taking notice of the best-sounding rooms and thoroughly appreciating the guys that were making real progress and doing real research. They also didn't miss any opportunity to make light of voodoo and magic tricks!

GLUE

You may have never thought about this before, but nearly every part in a speaker is secured by adhesives. The only typical exception is the attachment of the basket to the front plate. This is typically "staked," but even then, is still damped with a bead of glue between the two parts.

Speakers endure horrendous stresses, and it's pretty impressive that they stay together as well as they do. I can assure you that much time and attention is invested in this vital design ingredient. Jensen had a full-time engineer devoted to this purpose, and in the time I was there, he reduced the total number of adhesives used in Jensen's manufacturing from 27 to only five. Very impressive, considering the complexity and diversity of requirements.

Here is a brief list of constraints that a designer must deal with when choosing an adhesive:

How fast does it cure? Adhesives made with volatile, nasty chemicals cure very fast and form very strong bonds, but the fumes might erase your DNA. Water is safe but cures much more slowly. By the way, US environmental laws limit the use of the nastiest adhesive chemicals and therefore eliminate some otherwise attractive choices at the outset. For that reason, foreign-made speakers may have performance advantages, such as higher power handling.

Is it waterproof? Do ultraviolet rays break it down?
Is it brittle or does it have problems as it ages?
What temperature range can it withstand?
What materials will it stick to (and not stick to)?
Will it damage some materials as it cures?
Are there chemicals that accelerate the curing process?
Does it need to cure in an oven?
Does it leak and drip into tiny spaces?

In general, the following are somewhat typical of the kinds of adhesives, which are used for various surfaces:

Surround to basket, spider to basket: Water or solvent-based glue, somewhat resembling "Elmer's glue" in consistency.

Magnet to metal: Vile smelling, solvent-based adhesives, with two parts: the adhesive and the accelerating agent. The glue only cures when the agent is mixed with it.

Cone to surround: Varies widely according to the types of materials used. Hot melt is not uncommon.

Voice coil to cone: This one's critical. Typically cyanoacrylate ("Superglue" such as Loctite) or else two-part epoxy. Epoxy is great, but harder to use. "Superglue" is easy to use and cures quickly with an accelerator, but has problems at high temperatures.

Plastic cones, especially polypropylene, typically don't bond well with much of anything, unless you use special techniques. It's possible with most polypropylene woofers to peel the cone away from its adhesives without much difficulty. Poly introduces power-handling problems. Jensen developed a special technique that worked really well, but if I told you about it, I'd have to kill you.

By the way, guys with PhDs in chemistry just hate it when you call it glue. They prefer epoxy, adhesive, or hot melt. So, please don't denigrate the profession by referring to it as glue.

A SPECIAL "MEETING"

During the summer of '95, orders were slow, production was down, and rumors of "downsizing" were floating around. One day I received a tap on the shoulder and followed someone into the conference room.

An odd assortment of people from various departments were collected there, and Herman, the division manager, was pacing nervously and breathing heavily. He was sort of wheezing and looked really uptight. Herman managed to communicate to us that the company was experiencing a slowdown and had to turn some people loose. So...best wishes to all of us, and someone would be discussing terms of severance with us, and all that fun stuff.

I had discussed this possibility with my fellow engineer Giles Davis just a day or two before, and I figured that this was as good a chance as any to experience something else besides building speakers.

So I left the conference room. In the meantime, everyone else in the company had somehow found out what had happened. They were all "weirded out," with sort of a somber, deer-in-the-headlights, *"gee-I-still-think-you're-a-good-guy and I hope the unemployment line ain't too long for ya and, oh yeah, what if this were me"* look.

I met with a VP who pensively explained my severance plan, which included seven weeks pay—a great vacation if you don't need to find another job! I said, "You know, on the day I was born, it's not like some other guy was born on the same day with the duty of giving me a job. It's been great working here, and thanks for the opportunity. Sorry for the circumstances, but I'm sure things will work out fine."

And an hour later I had cleaned out my desk, and I was driving home at 11:00 in the morning, ready for my next adventure.

SEX-CHANGE OPERATION

I hit the ground running. And it couldn't have been more than three hours before the recruiters started calling with speaker design jobs in Indiana, Michigan, California, and Florida. The list of places where I could be living and working in just a few short weeks was growing.

But my wife was expecting, and we didn't wish to start over after being in Chicago for 2½ years, so we decided to stick around. I made the rounds with the various audio-related firms in Chicago and came up empty-handed.

My next step was to explore sales opportunities. Yes, despite the Hatfields-and-McCoys style of affection that engineers and marketers naturally share, I always believed I could, and should, live with one foot in each world—a good bit different from working in an engineering department, though. No doubt about that!

So I started exploring that venue, and within two weeks I had three job offers. I accepted a position as a manufacturer's representative, selling industrial automation components. That "controls and communications" specialty from school was just as applicable there as it was to audio systems...just a totally different industry and application of that knowledge.

I still work in the controls industry, and now I'm up to my eyeballs in the nuances of industrial networking. I was once an analog type of guy. Now I guess I must be digital.

I'm also a sales guy, not an engineer. And I can tell you, a guy who sells software and doesn't know how to write a single line of "C" code is a dangerous force in the eyes of any R&D department. They probably won't admit it, but I'm pretty sure my engineers have Perry Sink Voodoo Dolls in their desks.

AUDIO'S HOBBY STATUS

Speaker building is all about doing it for the love, not for money. That's the very essence of the term *audio amateur*. And I must admit: Doing speakers every day for a living may be more interesting, in some respects, than doing something more "utilitarian," but also, doing it every day takes some of the fun out of it.

I also must admit that ignorance can be bliss. While it's very rewarding to learn the subtleties and nuances of the art and science of speaker design, romance can also be displaced by reality. And you can't un-learn a truth.

Actually, I think the intrinsic conflict between romance and reality is at the heart of the "high-end audio" debate that I alluded to earlier, as well as driving the intrinsic conflicts that occur between engineers and the sales people who sell what the engineers design.

I'm a member of PSACS (Prairie State Audio Construction Society) in Illinois, which is a motley collection of manic audio constructor hobbyists like you and me. Once the speaker design job was in my rear-view mirror, those PSACS meetings started to seem much more interesting, and tweaking those speakers in my living room slid back up a few notches in my priority list.

The knowledge I gained at Jensen was not only intellectually satisfying, but it even earned me the distinguished title of "Official PSACS Speaker Design Wizard"—an honor that I hold with great reverence and pride.

POST SCRIPT

About three years after I left Jensen, things were going well there, but as I understand, the company found itself in the precarious position of having lots of great OEM designs in the pipeline, but, at the end of 1998, only one paying customer, who eventually cancelled its current project. Jensen OEM couldn't keep the lights on until the next project materialized, so they closed their doors in the spring of 1999.

To everyone's credit, I must say that nearly everyone who worked there when I did still keeps in remarkably close touch, particularly with e-mail—much better contact, in fact, than I've seen in any other company I've worked for before or since. It's a rowdy bunch of folks, and once in a while there's a Jensen reunion at some bar in Chicago, and I can assure you that a good time is had by all!

JOBS

I'm sure a few of you are wondering whether you could get a job like that—designing speakers, or perhaps some other component, for some audio manufacturer. Here are a few angles on that subject.

First of all, I should say that getting a speaker design job in a small, "high end" company is rather difficult, though not impossible. Very commonly, the owners of the company are responsible for that job, and though they may get a steady trickle of resumes, they are unlikely to give up that job to an outsider.

Jobs like the one at Jensen—pragmatic design jobs for large, consumer-driven companies—are much more plentiful and, in my opinion, offer greater technical and educational resources anyway.

Jensen had an extensive R&D department that, in reality, was doing more ground-breaking work and real research than 95% of other high-end, more "prestigious" companies. Sounds like an outrageous statement, but it's really true. You just need to also understand that this R&D was ultimately focused on economic benefits, not philosophical ideals.

Jensen employed lots of mechanical engineers, who often did not have audio backgrounds, as well as electrical engineers who handled the acoustical aspects of projects and were more likely to be "audio geeks." In fact, I'd say nearly all were audio enthusiasts. An EE degree, or a physics degree, are both taken pretty seriously by those who are hiring for such jobs.

As a final note, the audio world is in the midst of a massive collision with the computer world. This is a hot area with a multitude of possibilities for someone with a background in computer science, computer systems engineering, or digital signal processing. MP3 and all that stuff will not only transform the distribution of recordings, they will also change the nature of our audio gear! So if you desire to work in the audio industry, a diploma is a big help, and be sure and subscribe to the *Journal* of the Audio Engineering Society, as well as to *Voice Coil*, the loudspeaker industry insider publication. ❖

Xpress Mail

CROSSOVER CORRECTIONS

Several errors cropped up in Figs. 4 and 5 of "The Actipass Crossover System" (*SB* 8/00, p. 18). They are reproduced correctly on this page.

REGARDING THE PREMIER ISSUE

Great next step beyond the child publications! I thoroughly enjoyed the magazine and even the ads. I almost missed it all, though. The cover looks enough like a catalog that I almost recycled it without review.

Great start, keep up the good work!

Greg Senko
greg@senko.org

As a previous subscriber to *Speaker Builder* (since '97), I am gravely disappointed in the new magazine. If this is how the magazines are going to be then I'm sorry I subscribed this year. I absolutely loved *Speaker Builder*, and was always looking forward to the next issue.

I hope that other *SB* readers feel the same way I do, so maybe you can re-think your magazine strategy. I wouldn't mind getting a magazine every 3 or 4 months, if that's what it would take. I also wouldn't mind spending double the amount.

I don't feel it's worth my time or money to read about someone whining because an engineer performs THD measurements, especially a writer with such relentless, contemptible language.

Glen Caglarcan
gcaglarc@localisp.com

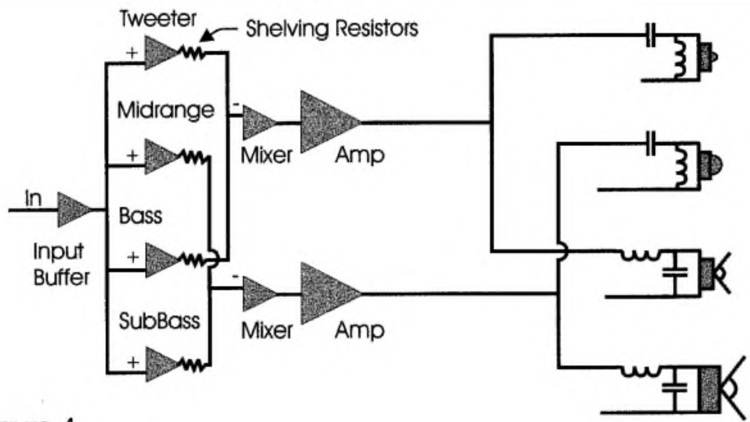


Figure 4

ActiPass 4^{th} order, 4 way
Treble: Active 2^{nd} order High Pass + Passive 2^{nd} order High Pass
Midrange:Active 4^{th} order Low Pass + Active 2^{nd} order & Passive 2^{nd} order High pass
Bass: Active 4^{th} order High Pass +Active 2^{nd} order & Passive 2^{nd} order Low pass
Sub Bass: Active 2^{nd} order Low Pass + Passive 2^{nd} order Low Pass
Alternatively (out of band model)
Midrange:Active 4^{th} order band pass + Passive 2^{nd} order High pass (c.400Hz)
Sub Bass: Active 4^{nd} order Low Pass + Passive 2^{nd} order Low Pass (c.400Hz)
Total of 8 Passive Components

B-1705-4

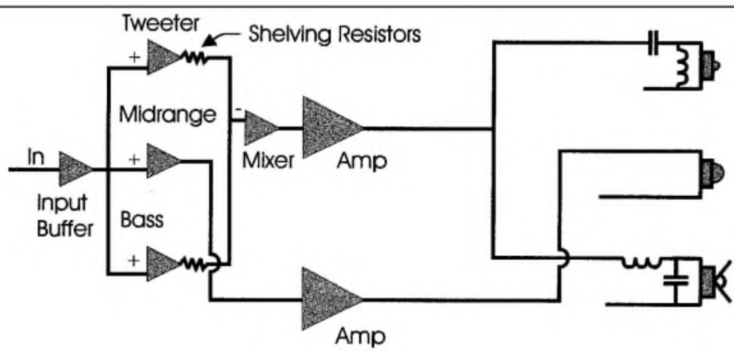


Figure 5

ActiPass 'out of band' 4^{th} order three way
Treble: Active 4^{th} order High Pass + Passive 2^{nd} order High Pass
Midrange:Active 4^{th} order band Pass
Bass: Active 4^{th} order Low Pass + Passive 2^{nd} order Low pass
Total of 4 Passive Components

B-1705-5

I think that the new mag is great—and this is from a 1977 vintage subscriber. (I got all the back issues when I first subscribed.)

The website should be tweaked for speed, however, and the e-commerce stuff should pop right out at you. One idea—an index to all the back issues on the website; I would love to search under Walt Jung, for instance. Another idea—subs to the Brit HiFi mags like *HFN* (I don't call it *HFNRR* anymore—no reviews to mention), *Grammyphone*—perhaps a link to Amazon or CDNow.

Jack Walton
jdwalton@home.com

6L6S: ANTICIPATING DISSIPATION

I really enjoyed Edwin G. Pettis' comments about the 6L6 family of beam pentodes (Letters, *GA* 5/00, p. 52). In his letter, he points out the similarities between the 6L6GC, the 6BG6GA, and the 807.

When I consulted my Sylvania and RCA tube manuals, I was amazed to find that the 6BG6GA has four possible ratings for maximum plate dissipation. In addition to the 19 and 35W ratings cited by Mr. Pettis, the RCAs have a

20W rating, while the Sylvanias have a 22W rating. Since I don't have a GE manual, I can't quote the rating for that brand.

In addition to the 30W rating for the 807, quoted by Mr. Pettis, the Sylvanias are rated at 25W. My RCA manual doesn't list the 807—not even in the obsolete section—so I can't tell what the rating is for this one, nor the GE, as I stated earlier.

The 6L6GC, on the other hand, cuts to the chase with a 30W rating across the board. While the 7581 is a bit huskier with a 35W rating, the 6L6GC is easier to find, and is easily affordable as well. In addition, quality of the foreign brands continues to vastly improve.

All who are interested in the 6L6 should get a copy of *Vacuum Tube Valley*, Issue #13, which features a very interesting history of this tube, along with a brand-by-brand evaluation.

Neal A. Haight
Castro Valley, Calif.

I commend Mr. Pettis for his excellent article. I have found that the 6BG6GA tube was produced in two versions using the large 7581A plate structure. One is the "top" getter, and the other is the clear top with "side" getters. The side-getter tubes have large radiators on the grid and screen-side rods. Some versions also use a "black" plate coating, similar to the RCA 6L6GC classic "black plate" tube.

I have had excellent results with all of the above in my audio projects, but the radiators on the grids make for a most reliable tube, as for guitar amplifiers which can be driven into "distortion" for long periods of time.

Joseph K. Risher
Sounds Great! Enterprises
Stone Mountain, Ga.

TESTING DRIVERS

First, I would like to thank G.R. Koonce for his thoughtful reply to my "dB or Not dB" letter in *SB* 7/00 (p. 38). While I am satisfied with the results of my investigations, he does raise some questions that require clarification.

On the testing of my Doppelganger Snail, the impetus for that design was maximum SPL efficiency. When I first tested it I was disappointed in the SPL/2.83V, expecting it to be 3dB higher, based on the premise that two drivers paralleled would give a 6dB increase over a single driver. Worried that the lower than expected SPL was due to a design flaw, I retested the box with only one driver in circuit. I removed the second driver from the cabinet and sealed its throat opening with a plywood plate. I did so not so much from concerns over throat impedance, but from the realization that if left in place the unwired driver would act as a passive radiator, which would have totally skewed the test result.

As to Koonce's worry over input impedances, he is rightly concerned. In my set of articles on DR horns (*aX* 1/01, p. 52), he will see that I have paid particular attention to impedance in those designs. If two drivers were placed in close proximity to each other on a common horn throat, there would be great potential for impedance-related anomalies, most especially in the effect upon the Fs(h) of the system. However, in the case of the Doppelganger the two drivers vent into separate throats, which merge midway down the horn's length. This minimizes impedance interaction between the two driver units, though undoubtedly this is also responsible for some of the cabinet's response roughness.

Regarding my testing of two horns side by side, this was done in a reasonably anechoic environment: my backyard. The nearest reflective surface, my house, is some 50′ from the test position. This might cause some half-wavelength cancellation around 11Hz, which is not problematic. I elevated the cabinets some 30″ above the ground, atop a test bench (which, in typical Yankee fashion, serves double-duty as a picnic table). Reflectivity off the ground plane, which in this case really is the ground, is minimized by a 4″-thick layer of absorbent material. [OK, I confess—I just don't mow the lawn as often as I should.] I measured at 60″ distance primarily to duplicate Koonce's test conditions, and secondarily to minimize interference from a pesky nest of starlings in the next yard over.

Admittedly, I don't have the test facilities of Joe D'Appolito, but within the requirements of the live sound genre that I specialize in, I'm sure that my results are adequate. As for the esteemed Mr. D'Appolito, he lives on the other side of Lake Winnipesaukee from myself, only some 20 miles distant as the crow flies. I fully expect that when I fire up the prototype of my DR15 horn next summer that he may well be able to conduct SPL readings on it by simply holding his Mitey-Mike out his back door. Joe, I'll let you know when I'm ready to go.

Bill Fitzmaurice
Laconia, N.H.

G.R. Koonce responds:

I thank Mr. Fitzmaurice for the clarification information on his horn testing. I have no fundamental problem with his side-by-side horn tests; after all, they got the right answer! I wanted to point out that horns are large and complex structures making detailed acoustic testing difficult. For these large structures, tests at 60″ do not meet the valid far-field requirement and larger distances make testing more difficult. Recall the rule-of-thumb for a valid far-field test is back three times the total span of what you are measuring, be it a driver, group of drivers, or a complex system.

The fact that putting two drivers on the horn's throat does not produce a 6dB output rise is problematic. Superposition theory clearly shows that when a linear network sums multiple independent, but identical, inputs there is an efficiency rise; 3dB for two inputs. My original article ("Dual-Driver Confusion," SB *5/00, p. 22) showed this happens with dual parallel-driven drivers, and Mr. Fitzmaurice showed it happens for two side-by-side horn systems. Why would it not occur with dual drivers at the horn throat?*

The answer may be in the words "independent" and "summed." As Mr. Fitzmaurice pointed out, two voltage sources connected in parallel do not change the power delivered to a load. Driving a speaker to 2.83V with two amplifiers in parallel puts no more power into the speaker than driving it to the same voltage with one amplifier.

For more power into the speaker you must use the two amplifiers in an "independent" manner that will "sum" their outputs; you bridge the amplifiers so they end up putting 2 × 2.83V across the speaker. When bridged, driving only one amplifier delivers 2.83V to the speaker load for a given amount of signal input power to the amplifier (not power supply power). Driving both amplifiers, thus doubling the signal input power, produces 5.66V across the speaker load and quadruples the output power! Surprise! It is the way linear systems behave with two independent, identical inputs.

So we know when two independent, identical inputs are summed there is a 3dB efficiency increase. We also know when identical voltage sources are simply tied in parallel we get no efficiency change. The question is how independent are the two drivers connected to the throat of a horn? If they are tightly coupled then we should expect no efficiency gain and only a 3dB output rise.

As I understand Mr. Fitzmaurice's Doppelganger Snails, they have independent throats that join midway down the horn. I don't know enough about horns to say whether this "joining" is a tight coupling, a summation of independent inputs, or somewhere in between.

I thus can visualize where a horn could show anywhere between 3dB and 6dB rise in *output for dual drivers in parallel depending on its construction. With the dual drivers at the throat it may be 3dB. With fully independent horns we know it is 6dB. With throats that join along the horn's path, who knows?*

If two drivers at the same horn throat are tightly coupled, this should be reflected by a change of input impedance for one driver when the other is also driven. Said another way, the impedance into the two drivers in parallel should not match one-half the impedance looking into one driver with the other driver shorted (equivalent to connected to an amplifier, but not driven).

As a point of interest it would be good to check the input impedance into one driver with the other both open circuit and shorted. If the two drivers are independent, what is done to the terminals of the not-driven driver should have no affect. Perhaps Mr. Fitzmaurice can examine this for his DR horn articles. I must limit my testing to direct radiators, as I don't have a stock of horns on hand!

TUBE STUDIO

I have subscribed to *Glass Audio* for a number of years. I must compliment you on the article in the 2/00 issue, "Tubes in the Recording Studio" by Lynn Olson. What I have been saying with too many words and for too many years was brilliantly summed up in four pages.

I have operated a recording studio in New York City (Sear Sound) for the past 36 years. We are still primarily a vacuum tube studio, including a tube console built in 1970.

Russ Hamm and I worked on the article published in the *AES Journal* entitled "Tubes vs. Transistors" back in 1973. We relied heavily on the comments of professional musicians in our research.*

Thank you again for presenting such valuable information. It is required reading in my studios.

Walt Sear
New York, N.Y.

**The Hamm article was reprinted in* GA *4/92, pp. 16–19, 23, 26, 42.—Ed.*

Lynn Olson responds:

I am honored to receive a letter from one of the co-authors of the well-known May 1973 JAES article. One of my goals in the 2/00 GA *article, as well as my earlier work in* GA *and* V&T News*, was to extend the conclusions of the Hamm/Sears article to medium and low operating levels. Over the last three decades, an engineering misperception has arisen that any distortion below 0.05% is inaudible, and thus of no consequence.*

It is true that A/B changes in electronics are hard to distinguish when auditioned on speakers with known crossover problems, excess diffraction, cabinet resonances, or resonant drivers, but that is a speaker problem, not a comment on the quality (or lack of it) in the electronics. Modern speakers, especially those with efficiencies 92dB/m or greater (>1% in absolute terms), clearly reveal tube versus transistor differences. The audibility of these colorations increases with increasing speaker efficiency, so I am doubtful that subjective tube/transistor differences have much to do with overload performance, at least in power amplifiers. Since amplifier colorations become progressively more apparent with decreasing power level, the true differences must lie in the mW region.

What appears to block further investigation is the fact that modern transistor elec-

tronics have very low measured distortion in the 100mW power region; in a well-designed transistor amp, THD can be right down in the noise floor. Yet on medium-to-high efficiency speakers, with the amplifier far from overload, amplifiers stubbornly sound quite different from each other, and not just due to trivial differences in damping factor. It takes more than a 2Ω series resistor to make a 200W Class-AB MOSFET amplifier sound like a 3W 2A3 amplifier.

The Rosetta stone to these questions is Norman Crowhurst's article ("The Amplifier Distortion Story, Pt. 1," GA 6/95, pp. 20–30), which should be required reading for all audio-electronics engineers. Using only simple mathematics, Mr. Crowhurst starts with a single idealized tube generating only second harmonic, adds a second (idealized) tube in cascade (resulting in second, third, and fourth), and then makes the circuit push-pull (resulting in third, fifth, seventh, and ninth). To cap it off, he adds feedback, which then produces harmonics out to the 81st order!

All this with pencil, paper, and an ideal model of a triode. As we know, real triodes are worse, and actual pentodes and transistors are ten to 20 times worse, in terms of generating high-order harmonics. Feedback reduces overall distortion, but does nothing to improve the low-to-high distribution of harmonics. In fact, feedback generates new high-order harmonics as a result of sum-and-difference cancellations at the summing node.

The Hamm/Sears JAES article discusses the destructive impact of high-order harmonics on musical timbre. For music with a simple and predictable harmonic structure, this analysis is an excellent starting point. For music with a dense and complex harmonic structure, it is more useful to look at IM distortion. Why?

When there are three or more equal-power dominant tones, IM distortion predominates over harmonic distortion. In fact, the more dominant tones are present in the original source, the faster new IM sum-and-difference products are produced. It is the difference between a straight arithmetic progression (harmonic distortion) versus a geometric progression (IM distortion).

This goes a long way towards explaining why a large chorus (with many equal-power tones at nearly the same pitch) can sound so harsh and distorted on high-feedback solid-state equipment. The steady-state measured THD distortion may be very low, but with real music, there are thousands of sum-and-difference IM terms crowding the noise floor.

The same problem with IM distortion happens to massed violins, or any kind of music in which there are large groups playing together in near-synchronous pitch. The large number of stimulus tones generates extremely large numbers of sum-and-difference IM artifacts that bear no harmonic relationship at all to the original music. Instead of simply changing timbre (like THD), IM distortion is a highly dynamic noiselike "hash" that rides up and down with signal level and harmonic complexity.

I surmise that this dynamic "hash" of dense IM products is what sounds electronic, "canned," and artificial to most listeners. There is no analog in nature to IM distortion, after all, and it is not a major problem in high-quality speaker drivers. This is also very likely the reason that a well-designed SE-DHT amplifier with 5% THD can sound clearer, and more natural, than a solid-state amplifier with 0.005% THD.

It isn't a matter of "euphonic distortion" at all; instead, it is freedom from a forest of high-order IM distortion terms. In an all-triode amplifier, the active devices have intrinsically lower distortion and a simpler harmonic structure, there are not as many devices in cascade, and there is no feedback to multiply the number of harmonics. From this perspective, the 1000:1 difference in full-power THD is completely misleading, and the SE-DHT amplifier is actually more "accurate" in terms of fidelity to (musically significant) low-level signals.

Of course, there are good and bad DHT amplifiers, just as there are good and bad transistor amplifiers. But I wouldn't be at all surprised if most of the sonic differences came down to four basic factors:

1) intrinsic linearity and freedom from high-order distortion at the active-device level (most important of all)
2) linear topology (each active device is in its most-linear operating range)
3) dynamic stability (freedom from Class AB transitions, slewing, and loss of phase margin with reactive loads)
4) low-noise power supplies (freedom from rectifier hash, AC line noise, and a low source impedance over the audio band).

AMP PROJECT

I have thoroughly enjoyed the article series by Joseph Norwood Still

("40W Triode/60W UL Amplifier," *GA* 2/98; "70W Low-Distortion Triode Amplifier," *GA* 5/98, and "100W Triode Amplifier," *GA* 3/00) and appreciate both the direct, thorough style and the content of the amplifiers themselves. I'm beginning my own version of a pair of monoblocks, each using a Sowters 2.2K UL 100W OPT and a Sowters power transformer, which is designed as a replacement for the Dyna ST70, although it will produce 300mA rather than the original's 250mA. I would like to be able to use various output tubes, including PPP EL34, EL37, EL38 (with plate caps), KT88, 6550, KT90, and maybe even 807 (with socket adapters and plate caps).

I will retain the triode/UL switching and would appreciate your comments on this project. The power transformers have a 55V tap for bias, and, to accommodate this flexibility, I will need a bias supply variable from about 25V to about 80V. Suggestions are welcome on a circuit for this–I have some NOS military mA meters and hope to be able to control the bias to each tube separately.

I am a big fan of the Svetlana EF86 and am wondering if you've examined the possibility of using a triode-connected EF86 or possibly even a paralleled pair of triode EF86s in the driver in place of the 5687? You may wish to look on the applications page of Svetlana's website for Eric Barbour's comments on the triode-connected EF86 (http://www.svetlana.com/graphics/TB/no.23fig3.jpg).

Thanks again for writing such a wonderful set of articles. I saw your comment that you and one other reader may be the only ones to own any of these amps. I promise you that won't be true very soon–the biggest problem is deciding which one to build first!

Bill Boyd
Greenville, S.C.

Joseph Norwood Still responds:

Good luck with your Sowters-designed power amplifier. Your choice of Sowters transformers is a good one—and is certainly a worthy consideration for a future article in audioXpress. *I thank you for the complimentary comments, and truly appreciate the analysis of my article.*

Per your questions: the 2.2kΩ UL, 100W Sowters output transformer is an ideal match for PP-P 6550s or KT88s. I prefer the 6550s because at the high power/high voltage level in the triode mode, they seem to behave a little better than the KT88s. I wouldn't even consider the rest of the tubes you mentioned. I would get quad-matched tubes and preferably one extra tube that matches the quad.

Also, when doing the matching, you want tubes that have a medium plate current reading—definitely not on the Ib side—this is very important when operating these tubes as triodes near the design maximum ratings. Also, if you operate the tubes at a resting plate current of 60mA, you can greatly extend their life. However, for superior performance and a reduced life, then 70mA operating current is OK.

I would not use the bias "tap" of the 55V on your Sowters transformer. This tap feeds a half-wave rectifier circuit with its inherently high ripple voltage. Instead, use a Radio Shack transformer rated 120V AC primary, 12.6V, 450mil secondary (part no. 273-1365, cost $3.99).

The transformer is operated reversed and the 12V AC winding is connected across the 6V AC filament windings of the Sowters power transformer. The 54V AC output of this transformer would be connected to full-wave bridge (RS #276-1173), filtered by a 100µF, 200V DC electrolytic capacitor. The output voltage of the supply is 67V DC. The ripple voltage measures 0.18V AC.

This circuit was terminated by an 18kΩ, 2W resistor, which loads the supply more severely than your final bias adjust circuitry and related grid circuits. Be sure to terminate the negative output of the bridge with a 1µF, 250V metal-film capacitor (RS #272-1055) to eliminate switching oscillation from the rectifiers. The bias supply should not be regulated so it can vary in tandem with output of the plate supply when they are subjected to variations in the AC line voltage. Use diodes in the full-wave high-voltage rectifier circuit in place of a vacuum tube rectifier, as this will enhance the regulation of the plate supply. Also, place a 220nF, 600V at the output of the full-wave rectifier to eliminate rectifier switching oscillation.

The EF86, connected as a triode, is a great idea. It is definitely a good tube as a driver or pre-driver, particularly in the triode mode, and features low noise design. I'm sure if Eric Barbour presented data for their use, it can be assumed to be "etched in stone" for selecting the best operating parameters. The only requirement you must follow is that the plate voltage of the EF86 should not exceed 110V, if it is connected to a long-tailed phase inverter.

Actually, for most linear operations it should be operated at 110V DC, which would be measured at the plate. In my opinion, the long-tailed phase inverter is the best inverter because it enables an amplifier to be built using only three stages. It also has only one coupling capacitor in the signal chain at any one cycle of the push-pull process. To parallel or not parallel the EF86 is your decision; you will obtain 30% more gain, lower noise, and lower plate resistance—this is in your hands.

TRIODE AMP UPDATE

Regarding "An Affordable SE Triode Amp" (*GA* 4/00), I have several questions.

There is no mention of power ratings for resistors in the schematics or in the accompanying article text (except for R5 in Fig. 1). Can you enlighten me?

Figure 3 shows +35V at D, which does not appear to connect to any of the other figures included in the article. The text states that the connection shown at "D" is used to bias the heaters off ground, but despite owning several books on tube design, I don't get it. Can you help?

Could I build this amp as a dedicated headphone amp, using only the power supply and power amp sections, and omitting the line and tone control sections (omitting the feedback)?

Robert McLean Lees
RLEES@Upei.CA

John L. Stewart responds:

Thank you for your questions. For whatever reason, I neglected to mention in the article that all resistors, with the exception of R5 in Fig. 1, in the power amp are ½W.

The +35V shown at point D is provided by the cathode bias developed across R5 of Fig. 1, the power amplifier. That would be terminal 3 on the 6EA7/6EM7. The result is that all of the earlier gain stages at lower levels will have their heater-cathode circuits reverse-biased. This usually reduces the heater to cathode leakage; the benefit is a reduction in hum at the amplifier output.

Although 3W sounds like a lot to drive headphones, this circuit would serve well in such an application. My original plan for this project did not include the added goodies of a line amp and tone controls. The power amp and power supply sections will do fine on their own. If the headphones are high impedance, you should parallel with them a 10Ω, 2W resistor so that the power amp is properly terminated.

Feedback is not required in this circuit, so you can leave it out if you prefer. I would also advise you to take great care with your electrical construction, in particular the soldering. In many of the photographs of recently published projects, it appears the authors concentrated on external appearance rather than what was inside the box. That can have a considerable bearing on the successful outcome of your project; badly soldered connections and loose joints guarantee poor sound.

You should use lots of terminal strips, which will facilitate the wiring of the various components. The cost is minimal and greatly enhances the appearance and performance of *the finished product. I prefer the solder lug type available from Antique Electronic Supply. Push the wires through the solder lugholes, but don't wrap them. That way you can easily make wiring corrections or changes.*

If there should be a destructive error in the wiring and your 6EA7/6EM7 expires, it is comforting to realize you could buy ten of these for the cost of a single 2A3 or 45. It is also worth noting that when the 6EA7/6EM7 was being built and used in quantity vacuum-tube production techniques were at their zenith, far better than they were in the 1930s.

While checking the article, I found another error that I missed in the review process. Referring to the Figure 3 Power Supply, you will find the power line capacitors C1 and C2 are labeled 100N. That should be 10N (nanofarads). Three different capacitors from Mallory are available to work at this location in the circuit. The size is not critical, but they should be equal to each other.

These caps remove some of the RFI (radio frequency interference) from the power line that might get into your amp. Suitable part numbers are as follows:

Mallory	*UN103M*	*10N*
	UN472M	*4.7N*
	UN102M	*1N*

For others embarking on this project, it is worth noting that all of the parts could be used later in an improved version of the amplifier. Like all single-ended circuits, the performance at low audio frequencies is compromised by the output transformer. When Hammond originally brought their 125 series to the market more than 40 years ago, they did not have hi-fidelity in mind. In spite of that, it is surprising how well they work in these applications.

A very significant enhancement is possible at reasonable cost by using one of the output transformers now available for single-ended operation.For example, I tried a Hammond 1628E and noted a five-fold improvement in distortion measurements at 100Hz, 1W. If you make this change, then change C4 (the cathode cap in the power amp) to 1000µF, 50V.

Finally, you should be careful while working with voltages commonly found in vacuum-tube circuits. Many with experience in solid-state circuitry are surprised the first time they are whacked by their vacuum-tube project.

HELP WANTED

I just saw a new speaker from Denmark called Avance Century. It had what I was told was a phase plug in front of each driver (with the exception of the tweeter). Can you tell me what a phase plug is and what purpose it serves?

Marty Schwartz
MACEZ101@aol.com

I was wondering where I could find information about the Sapphire Drive Series Speakers. I found them in an ebay auction with little info, and was wondering if there was a site or something that I could read a little more on them? They are 15" speakers. ❖

Jordan Rinke
djjordy@pacbell.net

Readers with information on these topics are encouraged to respond directly to the letter writers at the addresses provided.—Eds.

Glass Shard

Direct-Coupled Circuits Need Regulated DC Heaters

Most readers are aware of the controversy about whether or not regulated DC heaters are necessary. I would like to add some practical information to the subject that I discovered.

I have an old Hewlett-Packard tube oscilloscope whose repair manual states that the heaters of the first two preamplifier tubes in the vertical amplifier sections are operated with regulated DC.[1] It states that, "By supplying this filament voltage from a regulated DC source, vertical trace drift is greatly reduced."

Vertical trace drift is the annoying problem of the scanning electron beam slowly wandering up and down the cathode ray tube's graticule face as the amplifier tubes warm up and operate. If there is excessive drift, then you must constantly use the vertical position control to center the unstable trace you wish to view. Anybody that uses a scope, whether tube or transistor, should be familiar with this phenomenon.

DOMINO EFFECT

These vertical amplifiers are direct-coupled DC circuits that are very susceptible to thermal instability, which is simply caused by the effect of heater-voltage variations. Ultimately, drift is brought on by the fluctuating AC line voltage from the power mains, and this sets in motion a whole domino effect of problems: the fluctuating line voltage causes the heater voltage to vary, which causes, in turn, the tube's heat, bias, and the output voltage to vary...and the direct coupling causes the output voltage of the next tube to vary. This error gets passed on down the line and amplified at each stage, and the end result is distortion.

Incidentally, the input tubes in this particular scope are 12AU7s, but HP implies that any kind of tube in this situation, without regulation, would be sensitive to this problem. But with heater and B+ regulation, that scope, after a few minutes warming up, is very stable.

The *Electronic Designer's Handbook* states, "The effect of heater voltage changes is most pronounced in the input stage of a direct-coupled amplifier. Variations in the heater voltage cause changes in both the grid current and the voltage required between the cathode and the other electrodes to maintain a given value of plate current." It goes on to say that this problem "can be satisfactorily minimized by...applying a regulated DC heater voltage to the input tube."[2]

Even regulation of the power-supply high-voltage B+ for the input tube is recommended for these types of circuits to remove any kind of drift in gain and output voltage. Simply stated, a tube without regulation cannot distinguish the difference between a drift in the operating point or bias and a change in the input signal. Thus, it is essential to keep the bias and the gain as constant as possible.[3]

B+ AND HEATER FACTORS

In the past, DC tube amplifiers were used in analog computers and servo systems. These circuits were plagued with instability due to B+ and heater variations. B+ variations were easily cured with gas-tube voltage regulation, but counteracting and controlling the heater variations required very elaborate measures because no low-voltage regulation techniques were yet available (no zener diodes). This resulted in the advanced vacuum-tube technology of chopper and modulated-carrier amplifiers. Reading about these elaborate techniques, I got the impression that the cure was worse than the disease, but it was all necessary for stable operation.

Today, direct-coupled circuits are becoming increasingly popular, and this means that heater and B+ voltage-regulation considerations are still important and prudent. The reason for their growth is that they do away with capacitors and inductors or both (for output-transformerless OTL amps) in the signal path and their inherent frequency-dependent reactances and phase-shift properties. Some audiophiles consider that these two components degrade the audio signal, and their absence would make the ideal amplifier.

For circuit instances, the split-load phase inverter, cathode-follower, cathode-coupled, cascode, mu-follower, and screen-driven circuits are all configurations that can or do use direct-coupled stages. And the op-amp and differential-input circuits are sometimes DC-coupled to the next stage. These last designs exhibit the least susceptibility to heater variations because of their common-mode rejection characteristics.

REGULATION

For example, look at Dirk Wright's "Grounded Grid Line Stage" (*GA* 5/95, p. 40). This three-tube preamp is completely DC-coupled. However, he did not mention how to power those heaters or B+. Alan Kimmel's "Direct-Coupled Mu Stage" (*GA* 5/96, p. 30) is an innovative directly connected circuit, without an explanation of regulation. And, Bob Danielak's "Direct Coupling the EL509" (*GA* 5/98, p. 48) is another completely DC-coupled three-stage amplifier, but he decided not to use any kind of regulation on the heaters or B+.

According to sources listed in the references and what the engineers at HP have found, the input tubes to these circuits are all susceptible, to some degree, to some kind of bias drift caused by heater voltage and B+ deviations. These sources are unanimous in recommending regulation in vacuum-tube DC circuits.

In direct-coupled audio circuits, of which there are many types, an important aspect is that the bias levels can vary with heater-voltage variations, especially in the input tube. Also, gain and output-voltage levels can vary with the supply-voltage changes. B+ and heater regulation can permit these kinds of circuits to perform more stably in the chosen operating region.

Regardless of how two or more tubes are directly connected together—whether by the cathodes, plate to cathode, or by plate to grid—the bias and output voltage of the first tube sets the bias and output voltage of the next, and any deviation error in the first becomes compounded in the next.[4] An extreme error in an input stage could eventually drive one of the next stages into either saturation or cutoff.[5] So, if you are considering building a direct-coupled preamp or power amplifier, consider the advantages of heater and B+ regulation to make your design perform rock-steady, just like my old oscilloscope. ❖

Michael Kornacker
Palatine, IL

REFERENCES

1. *Dual Trace Oscilloscope, Model 122AR.* Hewlett-Packard Co.
2. Robert W. Landee., *Electronic Designer's Handbook.* McGraw-Hill Book Co. NY, 1957, pp. 3–75.
3. Richard R. Shea, Ed., *Amplifier Handbook.* McGraw-Hill Book Co. NY, 1966, pp. 18-2, 6, 7.
4. F. Langford-Smith, Ed. *Radiotron Designer's Handbook.* RCA, 4th edition, 1953 pp. 529–533.
5. Landee, *op. cit.* pp. 3–73.

Glass Shard

Williamson Preamp Phase-Splitter Applications

I have experimented with an excellent twin triode, which seems perfect for new Williamson-type amplifier projects (*Fig. 1*). The type 6AQ8/ECC85 features rather high transconductance, low microphonics, and an internal shield between the triode sections. Certain brands also have coiled heaters for extra-low hum levels. This type could replace the ECC81/12AT7 for this application.

The response of the circuit is from 15Hz to over 30kHz, with distortion unmeasurable on my equipment and no unbalance or stability problems at the higher frequency range. (I have had no problems with split-load inverters in this regard.) The voltage gain is higher than 6SN7 or 6CG7s (μ = 58 per section). ❖

Joseph Risher
Stone Mountain, Ga.

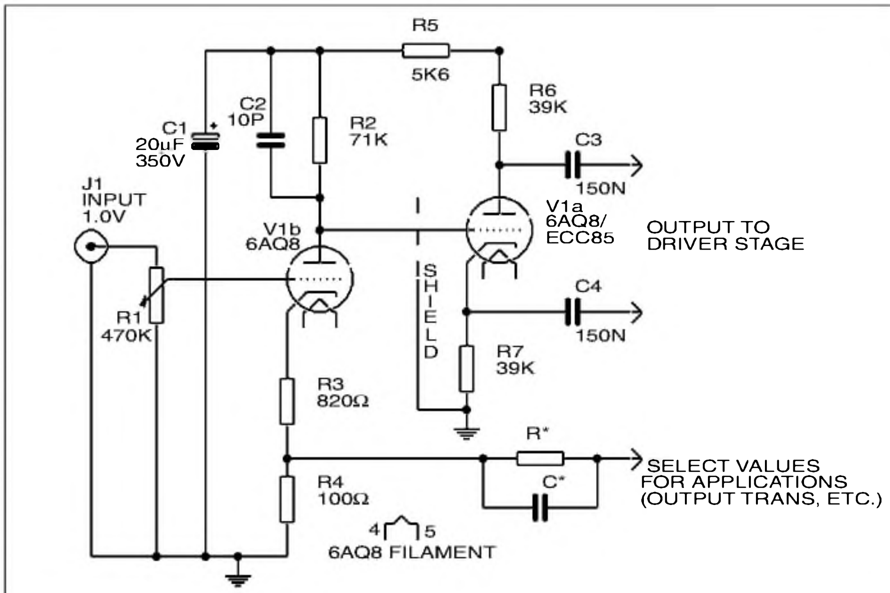


FIGURE 1: 6AQ8/ECC85 twin triode.

Dynaco MkIII *from page 24*

PRE-FLIGHT CHECKOUT

Before powering up the amplifier, use *Table 1* to check, and see that you read the resistances between the pads that correspond to the proper components. For example, between pad L and pad M, your ohmmeter should read 43kΩ. Double-check your wiring and momentarily switch on the amp.

Check that you see a reasonable negative bias voltage at each of the KT-88 grids. Once the amplifier is up and running, measure the circuit voltages at idle and compare them to the values shown on *Fig. 4*. If they are reasonably close, you should be fine. It is critical that the cathode of the 6CG7/6FQ7 is 5 to 10V higher than the voltage on the input grid (pin #7). This sets the bias of the direct coupling relationship.

You may prefer to do a power output test across a dummy load. My amplifier hit 55W at clipping, and went to 35kHz before serious gain reduction set in. I have not done detailed distortion measurements because the modification sounded so much better than the original that it seemed irrelevant. As my dad would say, I would be looking a gift horse in the mouth.

THE SOUND

Now that I went through all the trouble to modify this amplifier, what does it sound like? I found that the original MkIII had a hard quality, which was my single biggest complaint. The modification banished it forever.

However, there seem to be further benefits. The modification also operates with less feedback, and has a more spacious and lively quality. I find the midrange is much more refined in character, and that causes me to choose to listen more.

At my work, we compared a Mesa Baron amplifier to the Dynaco. While the Baron had better bass slam (at three times the power), the modified Dynaco compared well in the mids and highs.

I will be happy to communicate regarding questions and comments on this design. I may be reached at deHavilland Electric Amplifier Co. ❖

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12. Noncompliance with any of these guidelines will result in your free personal-ad submission being refused for publication.

Book Briefs

***Back- to-Basics Audio*, Julian Nathan. Newnes (Butterworth-Heinemann), Boston, 1998, 344 pp., $39.95**

A thorough review of the basic underpinnings of audio, mixed with a wide-ranging treatment of audio reproduction in nearly all the places in our lives where sound is propagated. Starting with a short chapter on underlying theory, the author moves steadily through measurements, amplification, waves and their units, transmission, speakers, recording, mixing, acoustics, cabling, power, controls, assembly, installation basics, and home theater. There are excellent appendices of practical problems and solutions, as well as a glossary and index. The treatment is crisp and thorough. Excellent for systematic learning, review, or reference. (Available from Old Colony Sound Lab, 603-924-9464, FAX 603-924-9467, E-mail custserv@audioXpress.com, BKB55.)

***Complete Guide to Audio*, John J. Adams. Prompt Publications (Howard Sams & Co.), Indianapolis, 1998, $29.**

A quickie, extremely simple survey of the main outlines of audio options, written for your brother-in-law who likes your high end system and wants it explained. (Available from Old Colony Sound Lab, BK559.)

***Audio Measurement Handbook*, Bob Metzler. Audio Precision, Inc., Beaverton, Oreg., 1993, 177 pp., $12.95.**

A highly useful, comprehensive introduction to audio testing in all its forms, by one of the four founding partners in Audio Precision, Inc. It is hardly an accident that the company is in Beaverton, home of that other distinguished measurement company, Tektronix. An indispensable bargain for serious audiophiles. (Available from Old Colony Sound Lab, BKAP2.)

***Audio Systems Technology*, Vol. 1, Level 1. Larry W. Garter. Prompt Publications (Howard W. Sams & Co.), 1998, $34.95.**

This is the first of a two-volume set published under the auspices of the National Systems Contractors Association as a basic handbook for training new technicians. The book begins with a chapter on correctly written communications, basic grammar, and spelling with some vocabulary. With that out of the way, from there the book goes the usual route to math, physics, and electronics basics, and proceeds to hardware, mostly speakers and microphones. If you have dreams of becoming an audio systems technician, this is the ideal place to begin. ❖

Test Tracks

I believe that subtle differences between components can only be detected during extended listening, as opposed to rapid A/B comparisons; thus, my preferred test tracks tend to be musical pieces that I enjoy. Most of the following albums are enjoyable in their entirety, and I typically play them straight through. However, for this discussion, I have identified specific tracks from these CDs that exemplify the best features of the recording.

I'm sure that these are not the seven "best" tracks in the world, or probably not even the "best" recordings of these pieces available on CD. However, these are recordings that I find useful in reviewing equipment. Similarly, I have identified only CDs because we do not own a turntable. LPs may sound better; in fact, I think there's a good chance that they do, but a lifestyle that includes young children makes CDs a "no-brainer" decision.

1. Charlie Daniels Band, *Fiddle Fire*, Blue Hat Records BLH-9703-2; track #2, "The Devil Went Down to Georgia." Although the Charlie Daniels Band is best known for country music, many of the tracks on this album would be equally at home on a rock recording. The transitions from the snare-drum solos to the entire band present a significant dynamic contrast.

The fiddle is recorded quite realistically and sounds clear and crisp without screeching. The vocalist is immediately identifiable as Charlie Daniels. Finally, the placement of the various instruments is distinct; the "evil hiss" produced by the "band of demons" is panned smoothly across the entire soundstage.

2. The Modern Jazz Quartet, *Blues on Bach*, Atlantic 1652-2. Track #2, "Blues in B Flat." This soft jazz quartet recording includes piano, vibraharp, acoustic bass, drums, and percussion. The entire frequency range is well reproduced, from the clean highs exhibited by the percussion throughout (particularly the high hat cymbals in the introduction) to the definition of individual notes produced by the acoustic bass. The piano sound is clear and well-defined; the transient "attack" of individual notes is excellent and crisp throughout. In short the piano sounds like a piano. The vibraphone has a "shimmering" quality and presents a lot of spatial and ambiance information.

3. The Doors, *The Doors*, DCC Compact Classics GZS 1023; track #1, "Break on Through (to the Other Side)." This album is in many ways typical of '60s hard rock with drums, organ, vocals, and lots of loud guitar. The dynamic contrast exhibited in the introduction is impressive (and potentially dangerous to your speakers). The first bars are cymbals alone—the guitar that enters after a few seconds is both much louder and much lower in frequency. Unusual for a rock recording, the definition of individual instruments within the "loud" mix is very good and the highs are clean throughout. (Note: the imaging is typical of this type of recording, i.e., terrible.)

4. *Beethoven Piano Sonatas*, Alfred Brendel, Philips 411 470-2; track #7, "Appassionata (Allegro assai)." This is an excellent recording of excellent piano music played by an excellent musician. The piano sound is realistic with each individual note well defined. A piano is capable of reproducing huge micro- and macro-dynamic contrasts; both have been effectively captured in this recording.

The transition at approximately

nine minutes between single notes and thunderous chords is particularly revealing. The image and definition of the piano remain stable and uncongested, even in the most complex passages. This recording is also a good test of frequency-response smoothness over the range of the piano.

5. The Weavers, *Reunion at Carnegie Hall*, Analogue Productions APFCD 005; track #1, "When the Saints Go Marching In." This recording of fast-paced folk music showcases a number of male and female voices. Both the space around the voices, or ambiance, and the placement of individual voices, or imaging, are extremely well defined on this recording. Due to the clarity of the recording, each voice can be identified uniquely. This clarity is also displayed in the applause segments at both ends of the track. You can distinguish individual claps within the soundfield. The percussion instruments produce notable transients, especially during the opening passages of the piece.

6. Doc and Merle Watson, *Pickin' the Blues*, Analogue Productions CAFPG 026. Track #8, "Hobo Bill's Last Ride." This recording of bluegrass/blues/vocal music features a number of acoustic stringed instruments and Doc Watson's vocals. Strengths of this track include the smooth guitar sound, the definition of individual plucked strings, and the realistic reproduction of Doc's voice. In addition, the spatial definition of the individual instruments is quite precise. Bass is present but not particularly well defined.

7. *Brahms Symphony Number 4*, Fritz Reiner, Chesky CD6; track #5, "Allegro energico e passionatto." This recording of a full symphony orchestra is so "huge" that it can be overwhelming in a smaller listening room. The dynamic range of this recording is also impressive. The bass response of the tympani and strings and the clarity of the horns are other important features.

OTHER CDS OF MERIT

On reviewing this list, I noticed several significant omissions.

I have not mentioned any of the Beethoven Symphonies. The entire five-CD set of the Beethoven Symphonies played by the Tonhalle Symphony Orchestra with David Zinman conducting (Arte Nova Classics) is well worth a listen, both for the performance and the recording.

None of my recommendations has really deep bass. However, most music (except some organ pieces) doesn't have really deep bass either. I think that the Modern Jazz Quartet and Brahms pieces have ample bass. If you prefer really deep bass, Telarc has Grofé's digital thunder (CD80086) and Tchaikovsky's digital cannons (CD80041), either of which can shred woofers if desired.

I have not included any large choral pieces. I will admit that I don't like choral music very much, so I don't own many examples. Although I'm sure that there is some good choral test material out there, I'm not familiar with it. ❖

Duncan MacArthur
White Rock, N. Mex.

New Chips on the Block

National Semiconductor LM3886 Power Amp IC

By Charles Hansen

(National Semiconductor informs us that "The LM3886 was released to production back on 8/26/93 and we have been selling it since." –Eds.)

The LM3886 is a high-performance audio power amplifier capable of delivering 68W of continuous average power to a 4Ω load and 38W into 8Ω with 0.1% THD+N from 20Hz–20kHz.

The performance of the LM3886, utilizing its Self Peak Instantaneous Temperature (SPiKe™) protection circuitry, provides an inherently, dynamically protected Safe Operating Area (SOA). SPiKe protection means that these amplifiers are completely safeguarded at the output against overvoltage, undervoltage, overloads, including shorts to the supplies, thermal runaway, and instantaneous temperature peaks.

The LM3886 maintains a signal-to-noise ratio of greater than 92dB with a typical noise floor of 2.0µV. It exhibits THD+N values of 0.03% at the rated output into the rated load over the audio spectrum, and provides excellent linearity with an IMD (SMPTE) typical rating of 0.004%.

The amplifier is also available in a TO-220 package (LM3886TF) offering complete electrical isolation without insulated mounting washers, and with no degradation of thermal performance. Device pricing is $2.20 US (5k pieces, LM3886T) and $2.25 (10k pieces, LM3886TF). National Semiconductor, Santa Clara, CA, 1-800-272-9959, www.national.com.

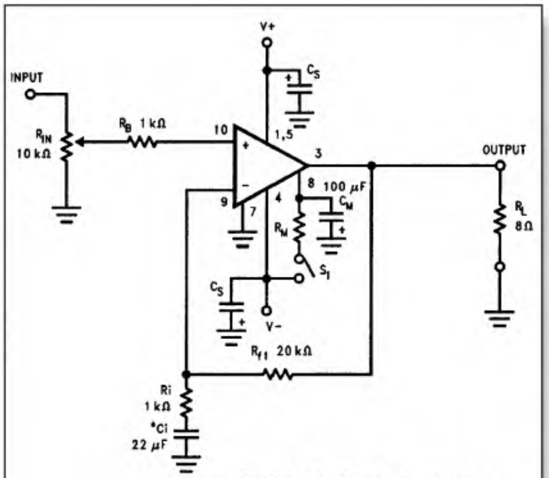

FIGURE 1: Typical audio amplifier application circuit.

FEATURES

- 68W cont. avg. output power into 4Ω at $V_{CC} = \pm 28V$
- 38W cont. avg. output power into 8Ω at $V_{CC} = \pm 28V$
- 50W cont. avg. output power into 8Ω at $V_{CC} = \pm 35V$
- 135W instantaneous peak output power capability
- Signal-to-noise ratio ≥92dB
- An input mute function
- Output protection from a short to ground or to the supplies via internal current limiting circuitry
- Output overvoltage protection against transients from inductive loads
- Supply undervoltage protection, not allowing internal biasing to occur when the sum of positive and negative supplies is less than 12V, thus eliminating turn-on and turn-off transients
- 11-lead TO-220 package (insulated and non-insulated)
- Wide supply range 20V–94V

APPLICATIONS

- Component stereo
- Compact stereo
- Self-powered speakers
- Surround-sound amplifiers
- High-end stereo TVs

(A number of application notes and the data sheet are available at National's website) ❖

Cornell Dubilier DSF Capacitor

By Charles Hansen

The Cornell Dubilier type DSF film capacitors are made with stacked metallized polyester, resulting in high volumetric efficiency and a very economical solution for general-purpose DC applications. They are ideally suited for blocking by-pass, coupling, decoupling, and filtering circuits, and are specifically designed for applications where high-density insertion of components is required. Ammo box style or reel taping available.

The stacked-film method produces substantial size reductions while providing greater pulse-handling capabilities than larger-wound metallized polyester sections, with dV/dt up to 55V/µs.

Typically, the 100nF 50V DC model costs 5 cents each.

Voltage range: 50–100V DC
(63V DC optional)
Capacitance range: 10nF –2µ2
Capacitance tolerance: ±5% (J) standard
Operating temp: –40 to +85°C
Dielectric strength: Rated V DC × 150%, 60 s
Dissipation factor: 1% max (25°C, 1kHz)
Insulation resistance:
C< 330nF : 3000MΩ min.
C> 330nF : 1000MΩµF min. ❖