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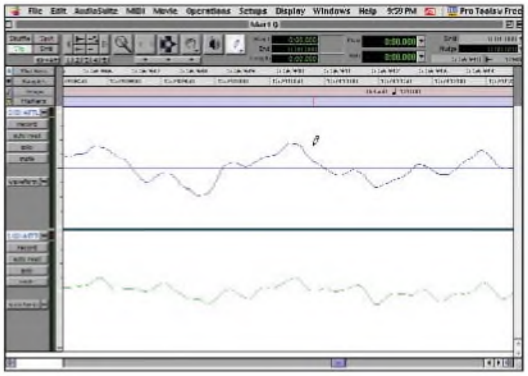

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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■ CATALOGS AND BROCHURES

All Electronics Corporation Spring/Summer 2001 new and surplus electronic parts and supplies catalog is available. For more information, call their customer service and information line at 818-904-0524.

McFeely's (Square Drive screws) also recently mailed their latest catalog, with advertised prices good until October 1, 2001. For more, call 1-800-443-7937.

CE Distribution, serving the guitar and audio market, announces its premier wholesale dealer catalog. CE Distribution offers a full line of parts to dealers and OEMs for replacement and production, including replacement parts for Fender, Marshall, and other amplifier manufacturers, as well as Jensen and Celestion speakers, JJ and Svetlana vacuum tubes, transformers, other electronic parts, and a large assortment of books. For more information, call 480-755-4712.

Scantek, Inc. released a new sixteen-page color brochure describing a complete line of environmental noise monitoring meters and systems from Norsonic. It discusses sound level meters, weatherproof enclosures, weather stations, masts and microphones, and analysis software, while also featuring the newest meters. For more, call 410-290-7726.

■ FENDER PRO REVERB AMPLIFIER

Fender Musical Instruments announces the release of a new 50W, all-tube, pro reverb combo amp featuring Jensen C12N vintage speakers. The Jensen C12N speaker was selected to match the circuitry used in the pro reverb. The pro reverb amplifier represents a first for Fender in that they have never released an amplifier with both tremolo and high-gain channel switching; it also features true tube reverb and a full-featured effects loop. For more, contact CE Distribution, 6221 South Maple Ave., Tempe, AZ 85283, 480-755-4712, e-mail: info@cedist.com, website: www.cedist.com.

■ BOSTON ACOUSTICS® MICRO SYSTEM9000 II

Boston Acoustics announces the addition of the Micro System9000 II complete home theater speaker system to its home speaker line. This system combines two Micro 90x II front satellite speakers, two Micro 80x II surround speakers, a Micro 90c II center channel speaker, and a matching subwoofer based on the PV600. It is also available in a new buffed silver-gray finish. For more, contact Boston Acoustics, Inc., 300 Jubilee Drive, Peabody, MA 01960, 978-538-5131, FAX 978-538-5100.

■ MARTINLOGAN LOUDSPEAKER TRANSDUCER

MartinLogan announces the introduction of Aeon™, a new reference loudspeaker system. Aeon features a new woofer, modern enclosure materials, improved construction techniques, and the MartinLogan ESL transducer. It also features advanced crossover topology, audiophile-grade components, and point-to-point wiring. For more, contact MartinLogan, 2101 Delaware, Lawrence, KS 66046, 785-749-0133, e-mail: dzell@martinlogan.com.

PROTON A/V PREAMPLIFIER

Proton Corporation introduces the AS-2631 Dolby Digital, dts 5.1 channel A/V preamplifier. The AS-2631's advanced multi-channel digital sound processor reproduces six independent audio channels: left, right, center, left surround, right surround, and independent subwoofer. It offers Dolby Digital, dts and Dolby Pro Logic Surround decoding with DSP. It also features eight different on-screen surround modes, six video inputs, and video monitoring and record output in both composite and S-video. For more, contact Proton USA, 13855 Struikman Road, Cerritos, CA 90703-1031, 562-404-2222, website: www.proton-usa.com.

BRAUNER VM1KLAUS HEYNE EDITION MICROPHONE

Microphone designer and manufacturer Dirk Brauner and microphone modifier Klaus Heyne teamed up to build a large-diaphragm, tube, transformer-coupled microphone, the Brauner VM1 Klaus Heyne Edition. This microphone builds on the VM1, but the Brauner/Heyne team shortened electrical pathways, carefully tuned the capsule and shaped the grille, and mitigated phase shift through acoustics. Like all Brauner microphones, the VM1 Klaus Heyne Edition is completely hand-made in Germany. For more, contact Aadvert International, Inc., 1920 Waukegan Rd., Suite 212, Glenview, IL 60025, 847-998-0600, FAX 847-998-0260.

CERWIN-VEGA SUBWOOFER

Cerwin-Vega introduces their new RL-28W dual 8″ powered subwoofer to its RL Series of loudspeakers, designed for home audio/video systems. The RL-28W features dual-stacked 8″ woofers in a slim-profile cabinet designed to be easily placed or hidden within the listening environment. It is powered by a 200W RMS discrete amplifier and delivers bass response via two front-firing 8″ EX+™ Extended Throw woofers mounted on cast aluminum frames. For more, contact Cerwin-Vega, 555 East Easy Street, Simi Valley, CA 93065-1805, 805-584-9332, FAX 805-583-0865, website: www.cerwinvega.com.

B+K FREQUENCY COUNTER

B+K Precision Corporation announces the addition of the Model 1856C, a lightweight counter capable of frequency measurements from 5Hz–2.4GHz, to its line of instruments. This counter was designed for a broad spectrum of laboratory and service applications and features a special 50W-terminated input for high-frequency use, as well as a standard 1MW input for frequency measurements up to 100MHz. The counter utilizes an eight-digit display and three different modes. The Model 1856C measures 4.5″ high by 10.375″ deep by 11.75″ wide, weighs 3.98 lbs., and comes with operating manual and line cord. For more, contact B+K Precision Corporation, 1031 Segovia Circle, Placentia, CA 92870-7137, (714) 237-9220, FAX (714) 237-9214, www.bkprecision.com.

NEW CABLES FROM TERK

Terk introduces its first line of cables for A/V components. The line includes S-Video connections (TSV-6 and TSV-12), A/V connections (TAV-6 and TAV-12), and RG-6 (TRG-6, TRG-25, TRG-50, and TRG-100) for indoor and outdoor use. For more, contact Terk Technologies Corp., 63 Mall Drive, Commack, NY 11725, 631-543-1900, FAX 631-543-8088, website: www.terk.com.

NEUMANN M 150

Neumann's new M 150 tube microphone builds on their M 50 tube mike, but with improved electrical specifications and a titanium capsule. Like its predecessor, it features a unique low-frequency-omni, high-frequency-directional pickup pattern. For more, contact Neumann, 1 Enterprise Drive, Old Lyme, CT 06371, 860-434-5220, FAX 860-434-3148, www.neumannusa.com.

NEW SPEAKER TECHNOLOGY

Edge Technologies, Inc., through its subsidiary, Etrema Products, Inc., developed a speaker technology, known as the "Folded Shell," that transforms a traditional cone speaker (2-dimensional) into a novel shell (3-dimensional) speaker system. Two prototypes have been fabricated using an off-the-shelf driver with a 100mm coil, and Edge Technologies wishes to form a joint venture with forward-thinking partners to refine the Folded Shell into a series of commercial products. Edge Technologies, Inc., 2500 North Loop Drive, Ames, IA 50010, 515-296-8030, FAX 515-296-7168, e-mail: ssv@etrema-usa.com, website: www.etrema-usa.com.

CINEMA PROPACK™ 600

The JBL Cinema ProPack 600 is designed to be easy to install and operate, while still maintaining sound and build quality. It includes a five-disc DVD600 DVD changer with precision optics and advanced digital video circuitry, a DCR600 A/V receiver delivering 100W × 5 channels power output, and the SCS135 loudspeaker system. The JBL Cinema ProPack 600 also features Dolby Digital, DTS, and MP3 decoding, with Logic 7® and VMAx® processing. For more, contact JBL Consumer Products, 250 Crossways Park Drive, Woodbury, NY 11797, 516-496-3400, FAX 516-682-3523, www.jbl.com.

MODEL 412 AUDIO ATTENUATOR/AMPLIFIER

TDL® Technology, Inc. announces their Model 412, a general purpose audio attenuator and amplifier. The 412 features include 80dB of attenuation in four decade steps, fine attenuation in each step with a single-turn pot, 40dB of gain in steps of ×1, ×10, ×20, ×50, and ×100, wide bandwidth, low noise, low DC offset, and low harmonic distortion. The instrument uses the new OPA227 and OPA228 low-noise op amps from Texas Instruments' Burr-Brown division. For more, contact TDL® Technology, Inc., 5260 Conchise Trail, Las Cruces, NM 88012-9736, 505-382-3173, FAX 505-382-8810, www.zianet.com/tdl.

Build the Audax/D'Appolito Home Theater System

Follow one audiophile's journey in building the enclosures for this six-speaker project. **By John Calcote**

PHOTO 1: The author with the finished home theater system.

I'm a woodworker. I had little doubt when I started building Joe D'Appolito's Audax home theater speakers that I could construct accurate and good-looking enclosures. But, I had no idea it would be so much fun (*Photo 1*).

Though I've enjoyed woodworking as a hobby for the last 15 years, my interests are actually quite diverse. I used to love to buy those Radio Shack breadboard kits (20 years ago when they were available) and build something with my soldering iron that worked on batteries and electronic components. Building speaker enclosures has allowed me to combine my woodworking interests with my old electronics interests.

Before I focus on the enclosures for this six-speaker kit, I'll spend a few initial words on my experiences buying the kit and then a few paragraphs at the end about installing the crossovers and drivers.

PLANNING

You can find complete instructions and documentation for the kit on the Audax website at http://www.audax.com/doit/us_ht01.shtml. I ordered my drivers from Madisound Speaker Components in Madison, Wis., whose website points to a page containing information about the parts they sell for the kit—http://www.madisound.com/audaxhometheater.html. The Madisound kit comes with all of the drivers, ⅛″ flat-surface acoustic foam, Dacron pillow stuffing, fully assembled crossovers, gold-plated terminal cups, screws, wire (you should have a little extra 16-gauge speaker wire on hand, because they don't give you quite enough, in my opinion), weather stripping (for sealing the ported enclosures), and plenty of rosin core solder.

In addition, the subwoofer plans call for an integrated 150W amplifier. Madisound sells a line of subwoofer amplifiers from Keiga. They supply the KG-5150, 200W subwoofer amp as part of their kit. It seems to be very well built from quality materials.

However, the documentation is limited to a single sheet of paper that contains a few installation tips. In fact, Madisound's website has a specification sheet in Adobe PDF format for the Keiga amps that contains more information than the page sent with the amp. Fortunately, subwoofer amps are just not that complicated.

The Audax plans call for using 2″ convoluted ("egg crate") acoustic foam for internal damping in the ported enclosures. However, Madisound doesn't sell 2″ convoluted foam. Instead, they sent me the ⅛″ flat-surface acoustic foam mentioned earlier, and a lot more Dacron pillow stuffing than I needed for the surrounds and the center channel midrange/tweeter enclosure.

When I called them, they seemed to become a bit defensive. I received the distinct impression this was not the first time someone had called them on this point. In fairness, I understand their position here, and I assured the support technician that I was not upset, only curious to know what my options were. He suggested that using the ⅛″ foam in conjunction with the Dacron stuffing will have the same effect as the 2″ convoluted foam without stuffing.

Well, perhaps that's true, but the fact is I'm a purist who will spend twice the money to do it according to specifications. So on the Internet I found a supplier of 2″ convoluted acoustic foam. Auralex manufactures a relatively inexpensive brand called Sonomat. Their website at http://www.auralex.com lists all of their authorized dealers. I found one close to me and paid a hundred bucks for a couple of 4′ × 8′ sheets—three times as much as I needed, but that's the smallest size in which they sell it.

WOOD SELECTION

For lumber, I chose medium density fiberboard (MDF) because of its stability and its relatively good internal damping qualities, and because I've worked with it quite a bit and feel comfortable with it. It glues up nicely, machines nicely, and sands nicely. About the only bad thing you can say about it is that it looks ugly. You can paint it, but I want-

ABOUT THE AUTHOR
John Calcote is a systems software engineer for Novell. He enjoys woodworking and building his home theater system in his spare time. He can be reached at jcalcote@novell.com.

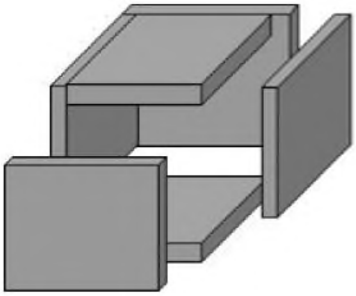


FIGURE 1: Exploded case drawing.

ed a natural wood look, so I used a peel-and-stick veneer product that my cabinetry supplier carries.

You could use other materials, such as furniture grade plywood, but make sure you get true cabinetry-grade plywood so that any internal voids are filled. The only real reason to use plywood is so you can get a veneer finish without having to use actual veneer.

This method actually requires more work than first appearances might suggest. You must use hardwood corners on your cabinets so you can join the plywood in a way that won't show the edge grain. You can use splined miters, but that's a lot of unnecessary work, in my opinion.

MDF is manufactured in ⅛″ thickness increments from ¼″ to 2″ thick; however, most lumber yards carry only the sizes most demanded by home builders—½″, ¾″, and 1-⅛″. I have an account with a wholesale cabinetry supplier who orders from the distributor, so I can get any size I need.

I used 1″ thick MDF for the front baffles and internal port supports, and ¾″ for the sides, back, top, and bottom, as per the cabinet plans. You can use the more common 1-⅛″ MDF for front baffles and port supports, if you wish. The cabinets will become ⅛″ deeper, but that's OK as long as the internal volume doesn't change. Additionally, the plans indicate that the entire subwoofer enclosure should be made from the thicker material.

GETTING STARTED

Lay out all of the fronts and internal port supports on the 1″ thick MDF sheet, drawing them at least ¼″ to ½″ oversized in order to account for saw blade kerf. Lay out some 3″ wide strips for the subwoofer bracing. You will need eight pieces approximately 18″ long. The length isn't critical, for you will cut them to fit together later at assembly time. Glue these pieces together in pairs to form four 2″ × 3″ braces.

Do the same layout and rough-cut procedure for the sides, tops, bottoms, and backs of the enclosures on the ¾″ sheet. Don't forget to lay out the sides and back of the center channel's internal enclosure. I found that you can easily cut all of the pieces from one 1″ sheet and one ¾″ sheet.

Try to align pieces with similar dimensions along rows or columns of your layout. If you don't have a long fence on your table saw, use a handheld circular saw to cut the sheet down into smaller pieces along these long layout lines, then use the table saw to cut pieces out of the smaller chunks and to clean up the edges.

Since I used simple butt joints on my

PHOTO 2: Squaring up the reference corner with a crosscut sled.

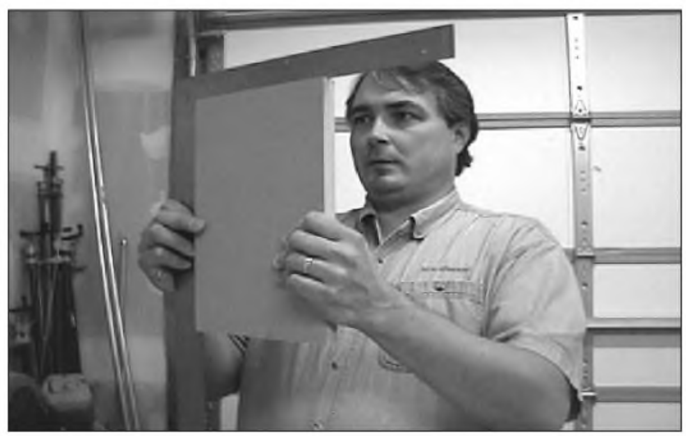


PHOTO 3: Checking the square corner.

PHOTO 4: Marking the reference corner.

PHOTO 5: Cutting a tweeter hole with a hole saw.

enclosures, the sizes of the fronts and backs are taken right from the external dimensions on the plans. The sides, tops, and bottoms are another matter. There are good reasons for gluing the backs and fronts on top of the edges of the open-ended box produced by the other four sides.

On the fronts, you need all the surface area you can get for laying out and cutting the driver holes. The veneer will cover everything except the backs. In order to have a clean look on the exposed back, it is best to have the back overlay all the sides that it touches.

One dimension of all four sides will be the external case dimension, minus the thickness of the front and back. The other dimension of two of the sides will be the same as the external case dimensions. The other dimension of the remaining two sides will be the external case dimension, minus the thickness of the first two sides (*Fig. 1*).

CUTTING THE PIECES

One of the secrets to woodworking—especially cabinetry—is carefully choosing a cutting order that will allow you to use the same tool setup for as many

PHOTO 6: Sanding the template with the drum sander.

PHOTO 9: Routing the driver recess.

PHOTO 7: Cutting out a driver hole with a jig saw.

PHOTO 10: Routing the round-over on the back of the driver hole.

PHOTO 8: Routing a driver hole with a template.

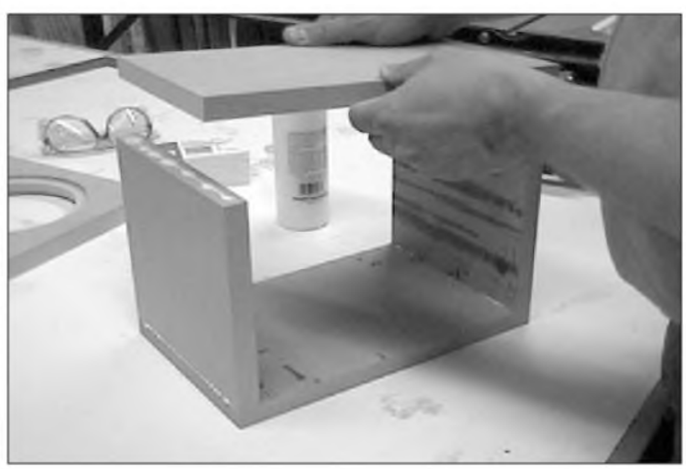


PHOTO 11: Assembling the sides, top, and bottom.

parts as possible. The basis behind this thought is that the tool setting is a much more accurate gauge than any measuring instrument. That is, two pieces cut without moving the saw fence will always be the exact same size. As a result, a methodology is critical to professional-looking results.

If you have a jointer, set the depth to 1/32″ and join one of the four edges of each piece. This gives you a clean, straight, reference edge. Then use a cross-cut sled or a miter gauge on your table saw to trim off one of the edges adjacent to the edge you just joined (*Photo 2*).

Make sure you place the joined edge against the miter-gauge fence. The idea is to get a single corner that is a perfect right angle with two clean, straight edges (*Photo 3*). This is your reference corner. Mark it with a pencil so you don't forget where it is on any given piece (*Photo 4*).

Now, set the table-saw fence for the smallest size in your cut list and make every cut in that size class, running the reference edge against the table-saw fence. If two dimensions on a piece are the same size, go ahead and cut both dimensions before moving on. This procedure can become a little tricky with this many pieces because you must keep

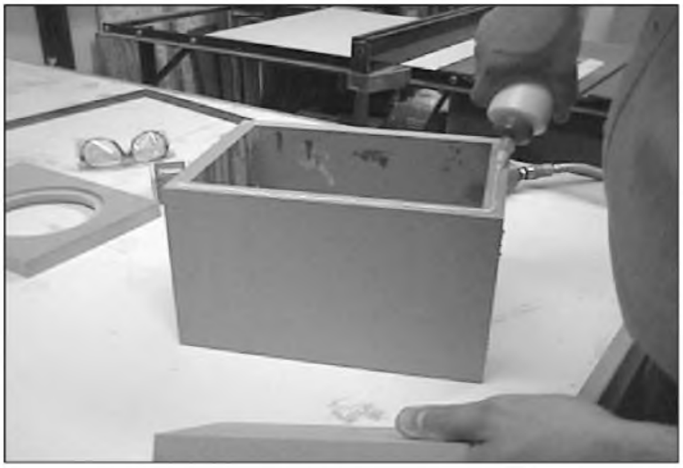


PHOTO 12: Applying glue for the back.

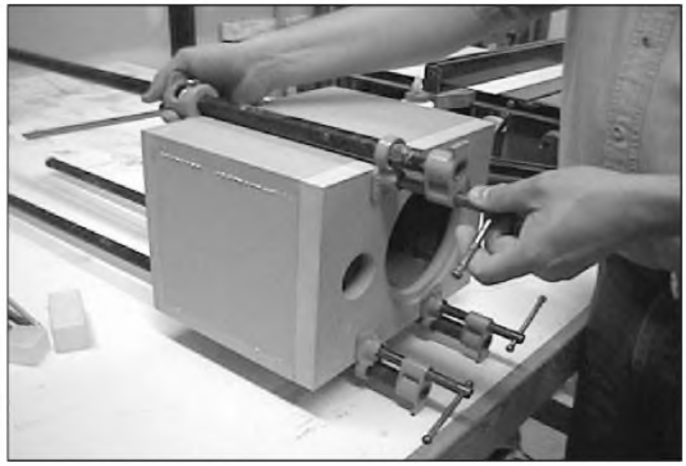


PHOTO 15: Clamping the side edges of the front.

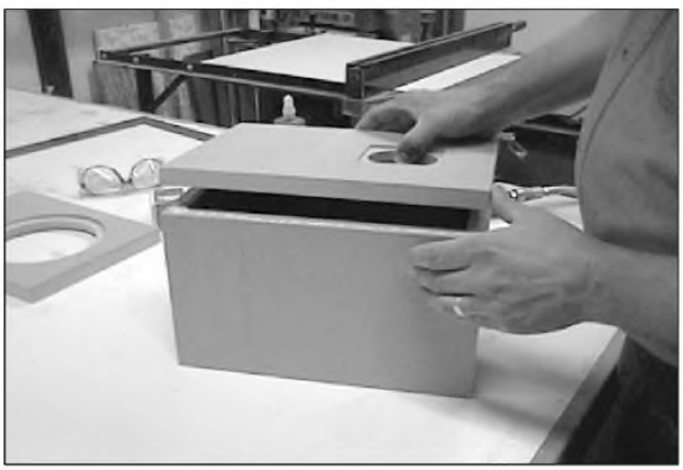


PHOTO 13: Attaching the back.

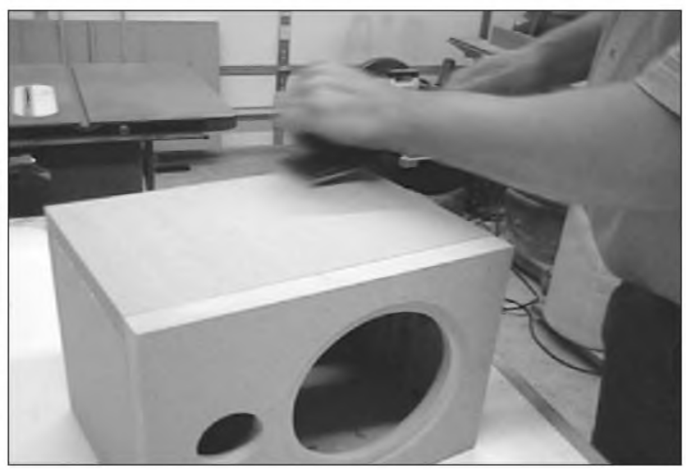


PHOTO 16: Using a glue scraper to remove excess dried glue.

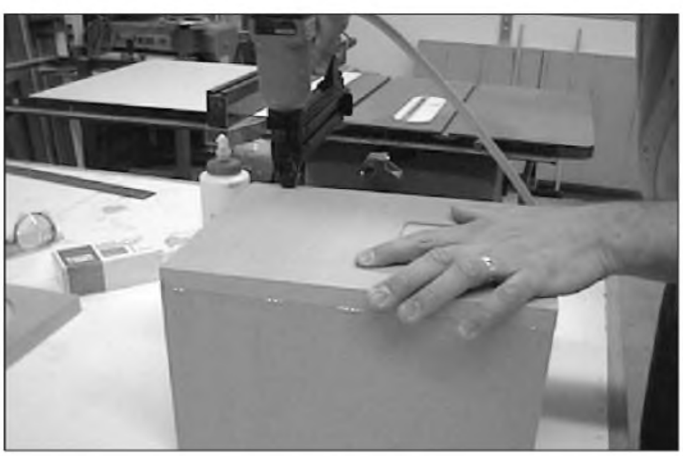


PHOTO 14: Fastening the back with a brad nailer.

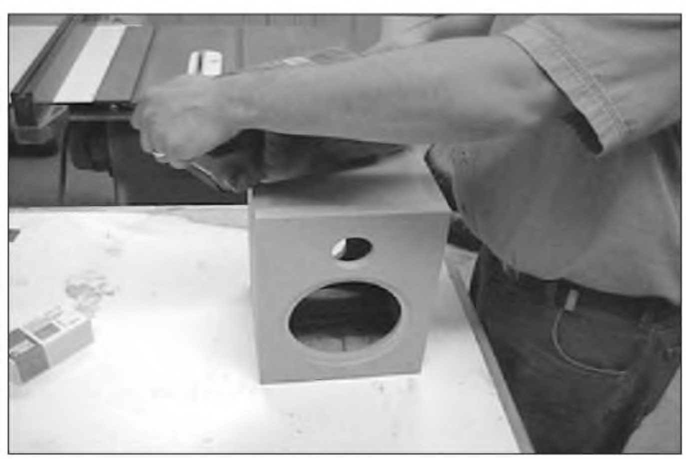


PHOTO 17: Using a belt sander to level the joints.

PHOTO 18: Routing a round-over on the side edges of the front.

PHOTO 19: Cutting the veneer.

checking both dimensions of each piece to make sure you haven't missed any.

I like to keep two output piles—one for pieces that are completed and the other for pieces that have been cut for only one dimension. In the end, you should have only a single pile of pieces cut for both dimensions. Using this method, you will always get perfect boxes.

DRILLING HOLES

When you have cut all of the pieces, it's time to consider the holes—in the fronts for drivers and in the backs for terminal cups and ports. The only exception is the subwoofer, which is ported in the front. Internal port supports also need holes.

I used a good-quality 2″ hole saw for the 2″ ports and the tweeter drivers, and a 2¼″ hole saw for the terminal cups, but you will need to gauge this hole to the size of your terminal cups. I used 3″ and 4″ hole saws for the 3″ and 4″ ports.

Hole saws clog and overheat easily on a drill press when cutting through MDF. To alleviate these problems, I blew a stream of compressed air into the slots of the hole saw while I was making the cut (I had my son hold the air hose for me while I made the cut) (*Photo 5*). To keep both edges clean, I cut only half-way through, then turned the piece over and cut the other half from the other side. The pilot hole from the first cut will guide the matching cut nearly perfectly.

The larger holes and recesses require a few router bits; ¼″ and ½″ rabbetting bits, short and long template bits, a laminate trimmer, and a ½″ round-over bit. You can usually find a rabbetting set for less than the cost of two separate bits. My rabbetting set cost 50 bucks and came with eight different bearing diameters, giving me the ability to make ⅛″ to ½″ rabbets.

A typical template bit has a 1″ cutting length and a bearing between the shaft and the blade so that the bearing rides against a template that sits on top of the work piece. The short template bit should have a ¼″ cutting length. A laminate trimmer looks like a template bit with the bearing on the bottom end of the blade.

Make all templates from ¼″ Masonite, making sure you use the kind that has a hard surface on both sides. Cut your templates oversized. You want as much flat surface as possible for the router to ride on.

Use a drafting compass to draw holes that are 1″ less in diameter than the size of the driver recess. Leave yourself enough room when you calculate this diameter to set the driver down into the recess. The last thing you want is to have to hammer your expensive driver into a hole that's too small at assembly time.

Drill a ½″ pilot hole near the inside edge of the circle and use a saber saw to cut out the circle. Stay at least 1⁄16″ inside the line so you don't accidentally cut away too much material. Then use a

PHOTO 20: Peeling veneer backing.

PHOTO 23: Using a laminate trimmer to trim the top veneer.

PHOTO 21: Applying veneer to the top.

PHOTO 24: Drilling holes in the laminate to trim the driver holes.

PHOTO 22: Using a J-roller to apply pressure to the veneer.

PHOTO 25: Trimming the driver holes.

PHOTO 26: Sanding the laminate.

PHOTO 27: Applying urethane finish.

drum sander on a drill press (or an oscillating drum sander if you have one) to carefully sand away the remaining material to the line (*Photo 6*). It's a really good idea to test your templates on scrap pieces before committing to them on your baffles.

Lay out the driver hole on the baffle with the compass. Drill a pilot hole and use a saber saw to cut out the circle (*Photo 7*). This time, stay a good ⅛″ inside the circle to keep the wandering saber saw blade from angling outside the true circle at the bottom of the cut. Use ¾″ double-sided carpet tape (available at any home store) to hold the template to the baffle. Two 3″ pieces—one on either side of the hole—provide plenty of holding power.

Set up the router with the long template bit such that most of the bearing is riding on the edge of the template. Your blade will not cut all the way through the baffle since the baffle is an inch thick and the cutting edge is only an inch long. This is OK because you will use the ½″ round-over bit on the inside edges of the holes. The round-over bit has its own bearing that will ride against the clean edge (*Photo 8*).

DRIVERS AND PORTS

Next, set up the rabbetting bit for a ½″ rabbet at the required depth to recess your driver properly (*Photo 9*). Don't forget to take into account the thickness of the weather stripping used to seal the driver flange against the recess. Since this driver has a polymer chassis, you must avoid cranking on the screw to seat the driver properly into the recess, which will bend the chassis. Instead, add at least 1/16″ of additional depth for weather stripping and then seat the driver gently against the foam. Rout the front face of the hole.

Again, it's important to cut as much as you can with the same router setting. Cut and clean up all of your template-based holes first, then rabbet them all at once. The midrange woofer in the center channel has a slightly shorter recess than the 6½″ woofers. Finally, use the ½″ round-over bit to round the edge on the back of the woofer and tweeter holes. According to the plans, you do not need to recess the tweeters.

For the terminal cup on the back and for the recessed amplifier opening in the subwoofer, you will produce square-holed templates with rounded corners. Quarter-inch radius corners seem to work nicely with the terminal cups and amplifier I received. Again, the more carefully you make these templates, the more professional your results will appear.

Both of these templates are fairly simple to get right. Use the terminal cup and the amp itself as a guide. Just set the template down over the device. If it fits nicely around the outside edge, then you're done. Look for tight spots and gaps between the device edge and the template hole, and try to clean these up with a ½″ sanding drum.

The hole in the back of the subwoofer enclosure for the amplifier need not be perfect. Just cut it about a half inch smaller all around than the template opening. Carefully align the template over the cup or amp hole—use double-sided tape to hold it in place. Use the short template bit to cut recesses for the amp and terminal cups.

For the ports, you can use 2″, 3″, and 4″ hole saws to cut holes in the enclosure backs and internal supports. These saws will produce holes that match the internal diameters of the PVC or ABS pipes that you use for the port tubes. Use a ¼″ rabbetting bit on the router to cut ¼″ by ¼″ rabbets for the tubes to sit in.

I found that the 3″ and 4″ tubes sat tightly into these recesses. The 2″ pipe

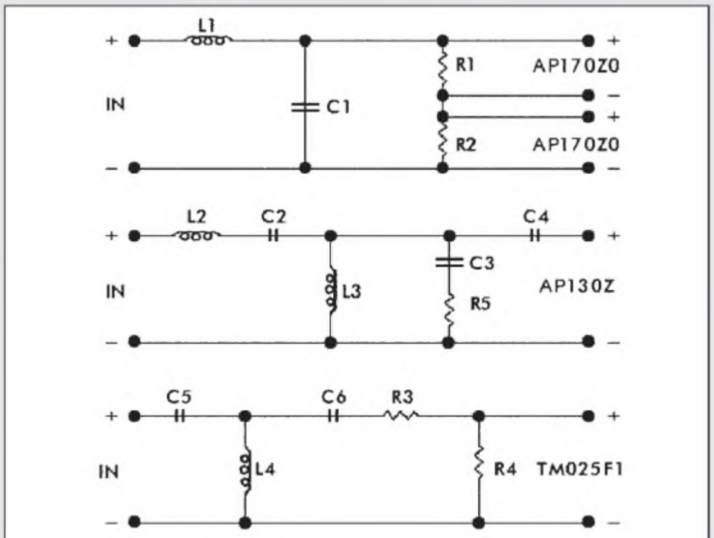


FIGURE 2: Center speaker crossover network diagram.

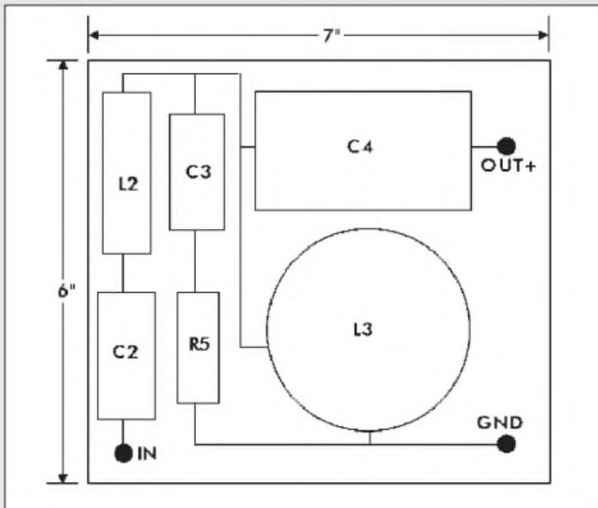


FIGURE 4: Crossover layout for center channel midrange.

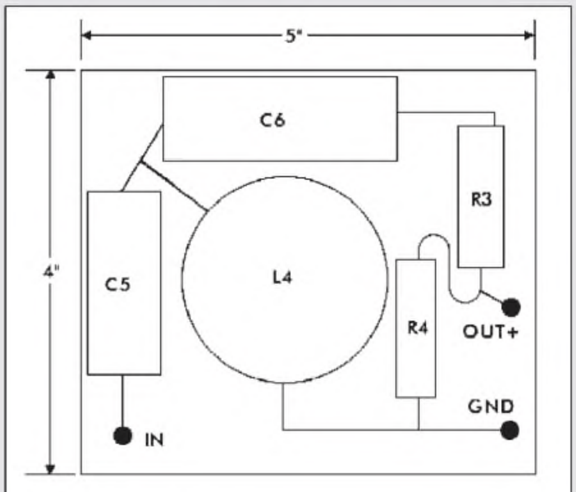


FIGURE 3: Crossover layout for center channel tweeter.

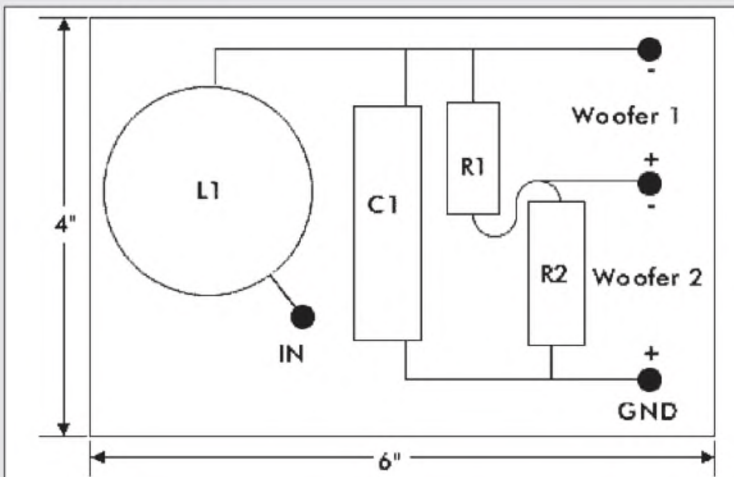


FIGURE 5: Center channel woofer crossover layout (note: woofer polarity is reversed).

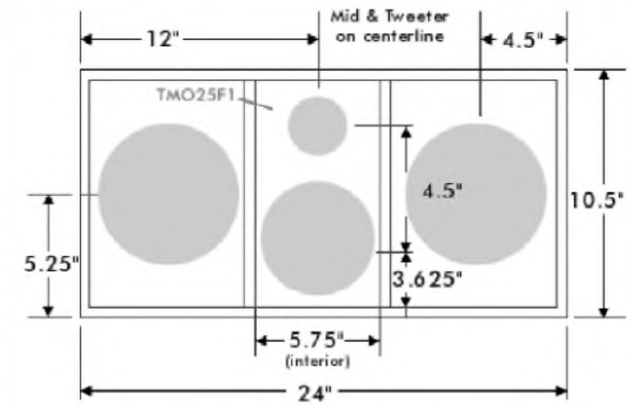


FIGURE 6: Audax center channel: front view. All material ¾″ MDF except front baffle. Front baffle 1″ MDF, edges rounded. Drawing not to scale.

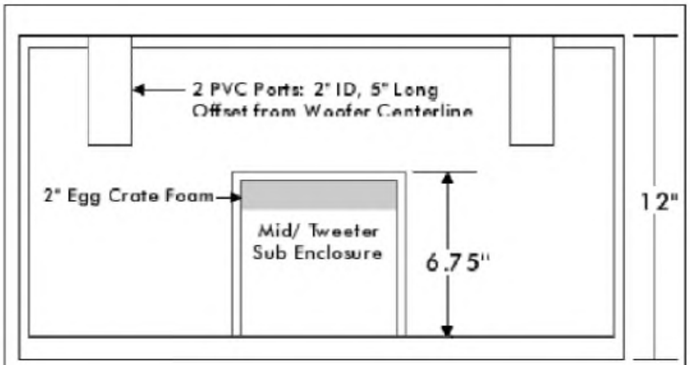


FIGURE 7: Audax center channel top view. Line rear, sides, top, or bottom (not both) with 2″ acoustic foam. Line rear of mid/tweeter enclosure with 2″ acoustic foam. Fill sub-enclosure with lightly compressed high loft Dacron. All material ¾″ MDF except front baffle. Front baffle 1″ MDF. Drawing not to scale.

is slightly less than ¼″ thick, however, and this caused me a little concern. It turns out to be a good thing though, because the 2″ ports are in the center channel enclosure and have no internal supports.

The extra space between the rabbet wall and the tube is just right for filling with five-minute epoxy. I used a disposable syringe to inject epoxy into the crack around the tubes. Glue these tubes in place before assembly and give them plenty of time to harden.

After you cut the port rabbets, rout the ½″ round-over profiles on the outsides of the ports. The bearing on the round-over bit will ride on the plastic tube, so have these in place when routing the round-over (*Photo 10*). The nice thing about this is that the holes will be widened to the exact inner size of the tube by the round-over bit. On the thicker material of the subwoofer enclosure, you might choose to use a laminate trimmer with a 1″ blade to accomplish the same effect before you apply the round-over to the opening.

ASSEMBLY

Finally, you are ready for assembly. This will go a lot quicker if you have a pneumatic brad nail gun. If you don't have an 18-gauge brad nailer, you can

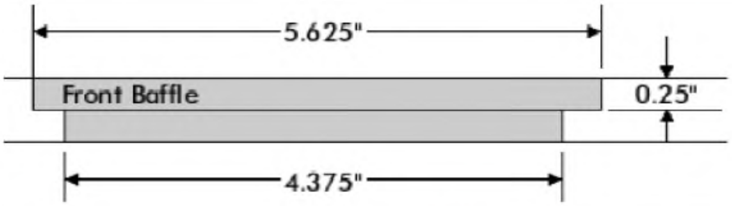
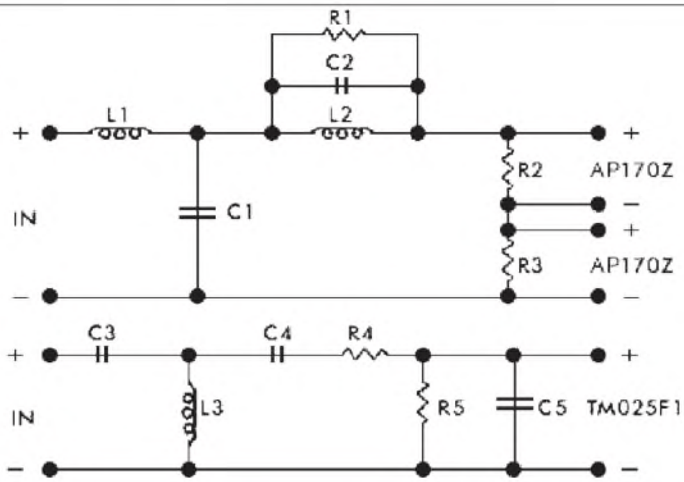


FIGURE 8: Cutout for 5.25″ AP130Z driver.

FIGURE 9: Left/right speaker crossover network diagram.



use clamps to hold everything in place while it dries, but this can be frustratingly difficult to do when assembling a small cube. It's much better to use drive screws and a drill.

MDF splits out easily when driving into an edge, so if you are going to use screws, take the time to drill a pilot hole the depth of the screw, and a countersink hole so the screw will sit slightly below the surface. This way, when you sand later, you won't hit the metal screw heads. Remember, the purpose of these screws is to act like clamps till the glue dries, not to hold the box together. It's the glue that really holds everything together.

First, assemble the sides, top, and bottom into an open-ended box (*Photo 11*). Use standard yellow carpenter's wood glue. You need to install the internal enclosure in the center channel. It works best to assemble these three

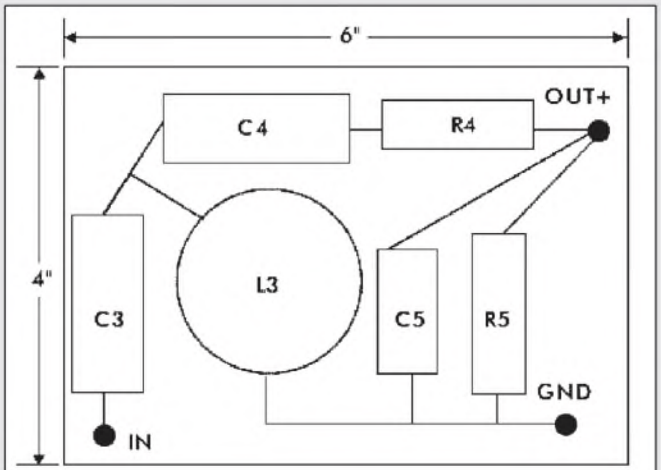


FIGURE 10: Left/right speaker tweeter crossover layout.

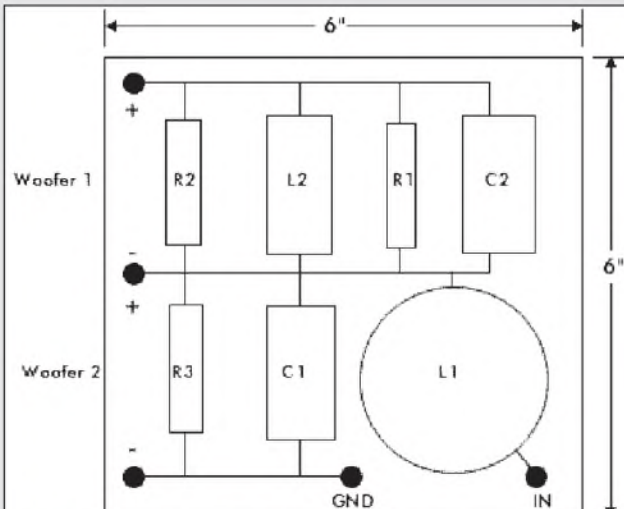
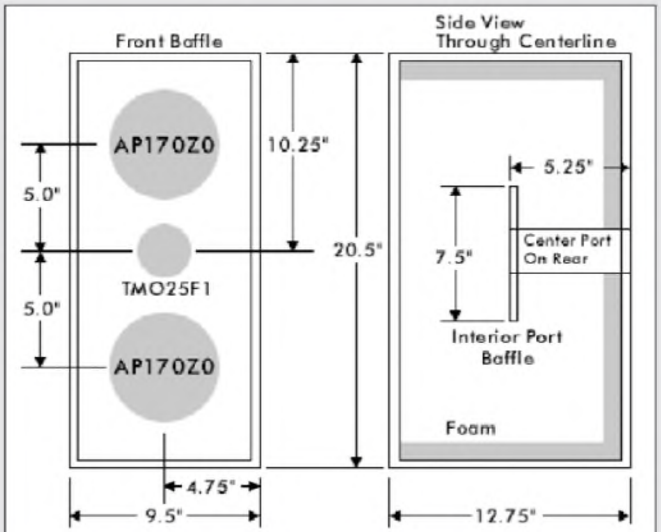


FIGURE 11: Left/right speaker woofer crossover layout.



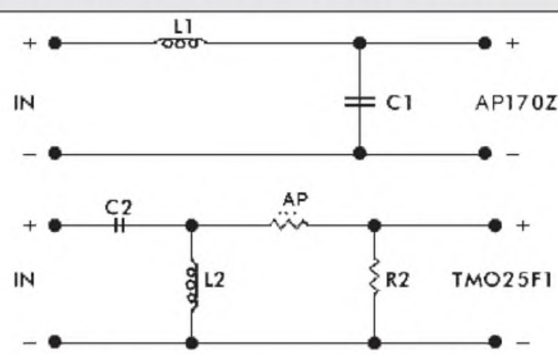


FIGURE 13: Surround speaker crossover network diagram.

FIGURE 12: Left/right channel enclosures. Note: Tweeter mounting hole is 2″ ID. Tweeter is mounted flush on front baffle. Top, bottom, sides, and rear lined with 2″ egg-crate acoustic foam. Drawing is not to scale.

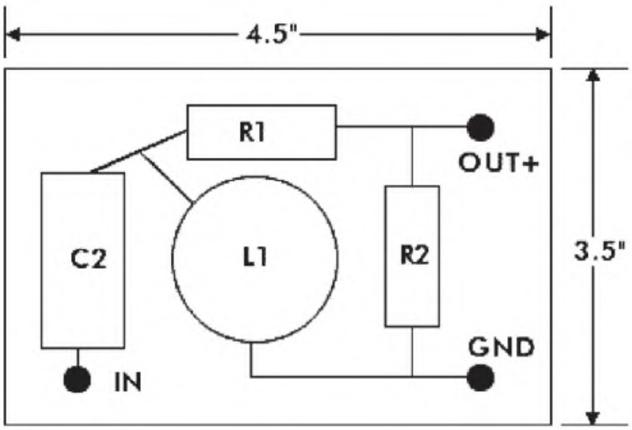


FIGURE 14: Surround speaker tweeter crossover layout.

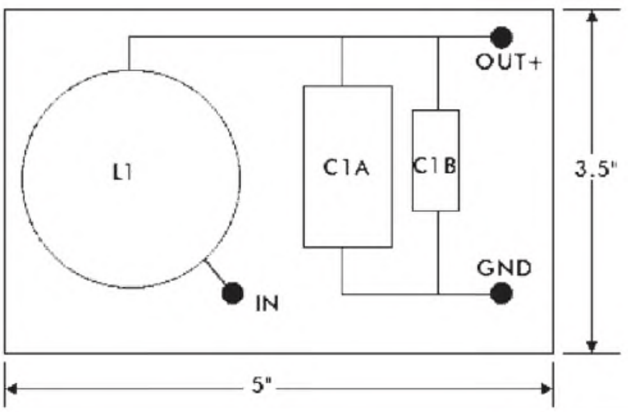


FIGURE 15: Surround speaker crossover layout.

pieces first, and then add the three-piece unit as you are assembling the four sides of the open-ended box. With this complete, glue and nail or screw on the back (*Photos 12–14*). Do a nice job of aligning and spacing the screws or nails on the back because it will be exposed.

In the left and right speakers, set the ports in place and glue and screw the internal baffles into place. You can add the subwoofer port after the cube is completely assembled, but you need to add the internal vibration bracing before gluing on the front baffle. The best way to do this is to cut two of the pieces opposite each other to the inside length of the box and glue and fasten them in place by nailing through the outside. Then cut the other two pieces to fit between the first two pieces, and glue and fasten them into place.

When you glue on the front baffles, don't use nails or screws on the left and right edges because you will round these edges over with the router. You don't really want to cut through these nails with your $30 router bit. Instead,

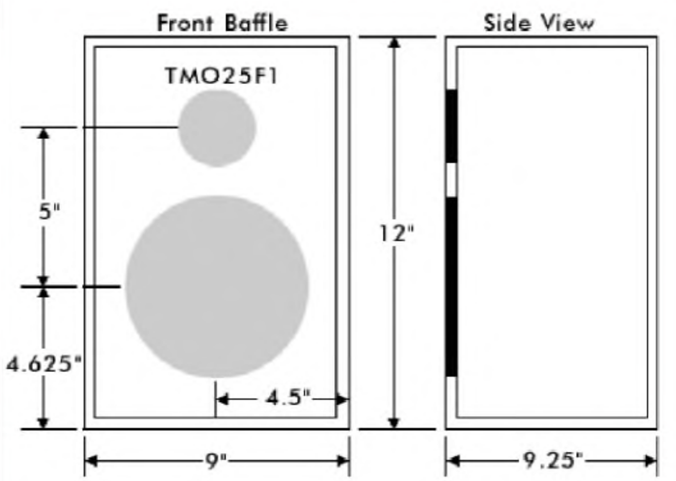


FIGURE 16: Audax surround speaker enclosure. 1. Front baffle is 1″ MDF, ½″ quarter round on front baffle, 2. Top, bottom, and sides are ¾″ MDF, mount terminal block on rear, 3. Mounting holes for tweeter and 6.5″ woofer are given on earlier drawings. Drawing not to scale.

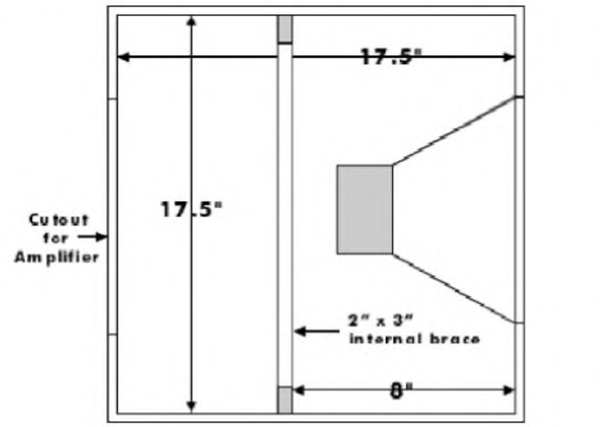


FIGURE 17: Audax home theater subwoofer (revised for HT300Z step 6 woofer). Notes: 1) Enclosure is a cube, interior dimensions (17.5″ × 17.5″ × 17.5″), 2) All material is 1″ MDF, 3) Line rear, top, bottom, and sides with 2″ acoustic foam, 4) Put 8″ × 8″ cutout for subwoofer amplifier on rear panel, 5) Port is 4″ ID PVC pipe 7.75″ long, 6) Place appropriate feet or spikes on bottom, 7) 2″ × 3″ brace across all sides.

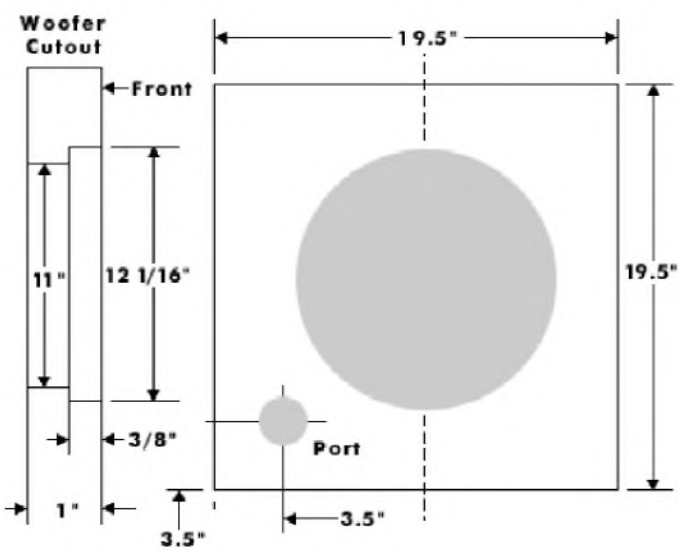


FIGURE 18: View of woofer baffle version A. Drawing not to scale. 7″ × 7″ baffle supports port tube inside opening. Apply ½″ quarter round to each port opening.

TABLE 1
CENTER SPEAKER

CROSSOVER PARTS LIST
L1 = 6.8mH, 0.48W, ferrite or iron core (#12AWG air core for the purist)
L2 = 1.2mH, 0.34W, #16AWG
L3 = 2.7mH, 0.53W, #16AWG
L4 = 0.27mH, 0.24W, #18AWG
C1 = 62μF
C2 = 24μF
C3 = 10μF
C4 = 82μF
C5 = 4.7μF
C6 = 8μF
R1, R2 = 10W, 25W
R3 = 8W, 10W
R4 = 15W, 10W
R5 = 2W, 10W

TABLE 2
LEFT/RIGHT SPEAKER

CROSSOVER PARTS LIST
L1 = 1.8mH, 0.43W, #16AWG
L2 = 0.15mH, 0.17W, #18AWG
L3 = 0.27mH, 0.24W, #18AWG
C1 = 8μF
C2 = 13μF (12μF and 1μF in parallel)
C3 = 6.8μF
C4 = 15μF
C5 = 2μF
R1 = 18W, 10W
R2, R3 = 10W, 25W
R4 = 5W, 10W
R5 = 15W, 10W

use a couple of clamps on each of these edges to apply some pressure while the glue sets (*Photo 15*).

Pay careful attention to orientation when you install the fronts and backs. I accidentally glued one of the front baffles on inside out. I didn't notice until I had finished and sat back to admire my work! Then I had to pry it off. The damage was mostly on the baffle, and I had enough material left over to manufacture another one.

Another area to watch closely is the orientation of the front baffle to the back of the subwoofer enclosure. It's really easy to get the front baffle oriented 90° off because the sub is a perfect cube and you must orient the amp a certain way for the heatsink to work properly.

APPLYING THE FINISH

I used a heavy-duty paint scraper to remove the excess dried glue (*Photo 16*) and a 120-grit belt in a belt sander to clean up the rest (*Photo 17*). Use the ½″ round-over bit to round over the left and right edges of the front baffles (*Photo 18*). This will allow you to wrap the veneer from the sides right around the front of the enclosure, producing a very professional-looking product. Remove as much dust as possible from the veneer surfaces with a hand broom and use a tack cloth to get the rest before you apply the veneer. The less dust you have on the surfaces to be veneered, the better adhesion you will get.

The veneer is simple to apply. Apply the top and bottom pieces first. Cut pieces so they hang over all four sides by at least ¼″ (*Photo 19*). Peel off the backing (*Photo 20*) and carefully align the veneer to the surface of the box (*Photo 21*). Use a J-roller to apply pressure to the entire surface (*Photo 22*).

You should pay careful attention to the grain of the veneer so you stay consistent. It doesn't really matter which way the grain runs on the tops and bottoms (I ran mine from side to side), but you should align the front pieces so the grain is vertical. The veneer wraps easily around a ½″ radius with the grain oriented this way. It's very difficult, if not impossible, to do it the other way.

After applying the tops and bottoms, use a laminate trimmer on the router to remove the waste (*Photo 23*). Clean up the surfaces with a hand broom and tack cloth and apply veneer to the sides and front. One piece all the way around would be ideal, but the veneer comes only in 2′ × 8′ pieces, so you have only a 24″ length to work with.

Make sure you have nice straight edges when joining two pieces together. Never join end-grain to end-grain, which looks terrible. Use the laminate trimmer again to trim off the waste. Use a ½″ drill bit to drill through the veneer in the driver holes (*Photo 24*) and use the laminate trimmer to trim to the edge of the holes (*Photo 25*).

For driver flange recesses, make sure the trimmer's bearing is set low enough to ride on the edge of the smaller hole. Then use the rabbetting bit to trim the veneer to the recess hole. If you set the depth to slightly less than the recess depth, you will trim off the veneer without cutting the recess any deeper.

The only problem I ran into was figuring out what to do with the 4″ port hole in the front of the subwoofer enclosure. The veneer is glued on over the top of this port since it's in the front of the enclosure. I used the round-over bit to trim the laminate to the edge of the port, but I set the bit 1/32″ low in the router so it produced a bead on the edge of the port.

This cut a neat circle in the veneer around the port. Now I had the raw edge of the MDF opening to deal with. After finishing the cabinet, I masked the circle and the ABS tube with masking tape and newspaper, and just sprayed the MDF edge with flat-black spraypaint.

Sand the entire box lightly with 120-grit sandpaper and then again with 220-grit using a random-orbit palm sander or a sanding block with the grain of the veneer (*Photo 26*). The veneer is only 1/32″ thick, so be careful not to sand through it with the heavier sandpaper. Also sand the back to flush-up the edges of the veneer with the back of the box and to remove any pencil and machine marks from the surface of the exposed MDF back. It's important to truly flush the thin edges of the veneer to the face of the box so it doesn't catch on something when you move your speaker around.

Apply a couple of coats of urethane finish (*Photo 27*). I used a water-borne urethane. The water base in the finish tends to raise the grain of the wood a bit, so you need to rub down the veneer surface with #00 steel wool (or the equivalent synthetic steel wool) after the first coat.

CROSSOVERS

When the urethane is dry, it's time to install the crossovers. Crossover layout and placement in the enclosure are shown in *Figs. 2–18*. They are taken from Audax information sheets. *Tables 1* and *2* are the crossover parts lists.

Here, I had a little trouble. The problem is that it isn't really clear in the plans how to install these. I e-mailed Joe D'Appolito about the issues I had, and he cleared up much confusion.

Cut a piece of ¼″ Masonite about ¼″ larger in both dimensions than the crossover boards. Be careful not to cut them larger than the driver hole, or you won't be able to fit the crossovers into the enclosures. It seems obvious enough, but it's easy to overlook when you are busy with something else.

Decide where you are going to place your crossovers inside the enclosures and measure lengths of wire to run from that location to the appropriate drivers and the terminal cup hole. Make it easy on yourself and cut lengths of wire that are a few inches longer than you really need.

Remember, you will need to solder the ends of these wires to the devices before installing the devices. Solder one end of each pair to the appropriate pads on the crossovers. Mark the other end so you know which device it goes to. You'll be covering up some of the crossovers with foam before you use the other end of the wires, so you should mark them now.

You can purchase a product called "Goop" from the hardware store for a few dollars a tube. One tube is plenty. Goop is funny stuff, though—until it's completely dry, it acts more like a lubricant than a glue. It causes the glued surfaces to slide easily on each other if you apply the smallest pressure in that direction.

Because of this, I found that it worked best to bundle up the wire leads with a rubber band and then glue everything into place and let it harden for a few hours before continuing. If the wires are touching any surfaces inside the enclosure, that pressure may cause the crossovers and Masonite mounting boards to slide out of alignment. Glue the crossovers to the Masonite boards, and then glue the boards in place inside the enclosures. I used four dots of Goop, about an inch from each corner under the crossovers, and then again under the Masonite boards. This stuff is very strong when it hardens, so don't worry about your crossovers falling off—it won't happen.

Madisound sent many extra screws—I believe they intended for the crossovers to be screwed to the inside of the enclosures. If you wish to be able to remove your crossovers, you might consider this approach. Screw the crossovers to the Masonite boards with the pan-head screws they sent, then glue the Masonite boards to the inside of the enclosures. In this case, the Masonite backer boards exist only to make mounting the crossovers easier. You can't get a drill inside the enclosure, or you could simply screw the crossovers right to the enclosure.

You mount crossovers in one or two steps, depending on the system you are working on. You can mount most crossovers in one step, but you must mount the two crossovers of the left and right speakers on opposite vertical walls. I turned the speaker sideways and mounted one, then turned the box over after a few hours and mounted the other one. Don't become anxious here and try to do the second one too soon.

For the rest of the units, choose a location that will not be covered by foam—for instance, the bottoms of the surrounds and center channel. The surrounds don't have foam and the center-channel plans call for foam on the top or bottom, but not both. I mounted the crossovers on the bottom and the foam on the top; however, the port tubes are low in the outer chamber and tend to get in the way. You might find it easier to mount the crossovers on the ceiling and place the foam on the floor.

The internal enclosure in the center-channel speaker is too small for both the tweeter and the midrange crossovers. Besides, the midrange crossover won't fit through the midrange driver hole. So I mounted the tweeter crossover on the floor of this chamber and the midrange crossover on the floor on one side of the outer chamber. I placed the bass crossover on the other end of the outer chamber. Then I drilled a ½″ hole in the back of the inner chamber through the terminal cup hole for the wires.

You can purchase from any hardware store a $6 can of 3M brand contact cement, which is more than adequate for holding the foam in place inside the enclosures. Remember that no one will ever touch this stuff once your kit is complete, so spot-gluing is just fine.

Spot-spray both surfaces and set the foam in place—if you wait too long you may not be able to slide the foam around if you need to, so I suggest you just spray and stick without waiting the usual few minutes for a tacky surface to develop. In many cases, I placed the foam in position inside the enclosure, then peeled it back from the wall, sprayed some adhesive on the surfaces, and pushed it back in place. This was easier than trying to position the foam after it had been glued.

Place the foam right over the crossovers in the left and right speakers. There'll be a hump there, but that's OK. Poke a small hole in the foam over the crossover leads and push the wires through as you position the foam.

FINAL TOUCHES

For the ported enclosures, you'll also need to apply weather stripping to the flange recess. Stretch it just a bit as you apply it, and it will easily form to the curve of the opening.

Applying the weather stripping to the tweeter openings is a little more difficult. You need a good base of stripping for the tweeter flange to sit on, but you don't want it to extend outside the flange. The best approach is to use a very sharp pencil to draw a light line around the tweeter flange. Then apply the stripping right to the pencil line.

Solder the drivers and terminal cups to the crossover leads. Then position the drivers and drill pilot holes for the screws. You can probably drive the screws into the MDF without pilot holes, but it will be much more difficult, and you might cause the flange recesses to crack.

Remember, the only thing that holds a screw in place is the threads, so there's no need to make a pilot hole any smaller than the screw body. I had my vacuum cleaner hose handy to suck up the sawdust from the driver flanges before driving the screws into place.

Ironically, the subwoofer is the easiest of the set to put together. The amp has a 50–100Hz variable crossover and summing circuitry built into it, so all you need to do after installing the acoustic foam is to fasten the amp to the enclosure, connect its red and black leads to the driver with the solderless connectors on the leads, and then screw the driver into place.

The polymer cones in these Audax drivers have a fairly tight sound. This is perfect for the detailed imaging you need for quality home theater. I'm quite pleased with the way they sound with my 20-year old JVC stereo amp. I can't wait for my new Denon five-channel amp to arrive in the mail so I can try them out on a real DTS processor! ❖

The Trouble With Screen Grids

This audio veteran shows you how to achieve the results you want with beam power tubes. **By Bill Chater**

PHOTO 1: Monoblock 100W cascode power amplifier. The chassis is an LMB UPS175-7-14.

Readers of the article entitled "Cascode Power Amplifier Circuits" (*GA* 5/98) will realize that the circuit discussed was probably compatible only with triodes. The thrust of that article was the careful control of push-pull cathode currents, and the introduction of a tube with a screen grid would throw all the careful work into a cocked hat.

But I was intrigued with the idea of extending the circuit into higher power. Part II of the article (*GA* 6/98) described the application of the circuit to the 6C33 tube, and provided a design able to reach 80W output. Although successful, the resulting amplifier was rather large, heavy, and ran hot (6C33s, you know). The accompanying circuitry was also a bit too complicated for the result I wanted.

Altogether, it was a hard way to get to 80W. I decided I could do better.

ROAD TO HIGHER POWER

One way to higher power is to use tubes able to pass as much as 1A without being limited to rather modest plate voltages. (The 6C33 can certainly pass the currents required, but is designed to operate at relatively low volt-

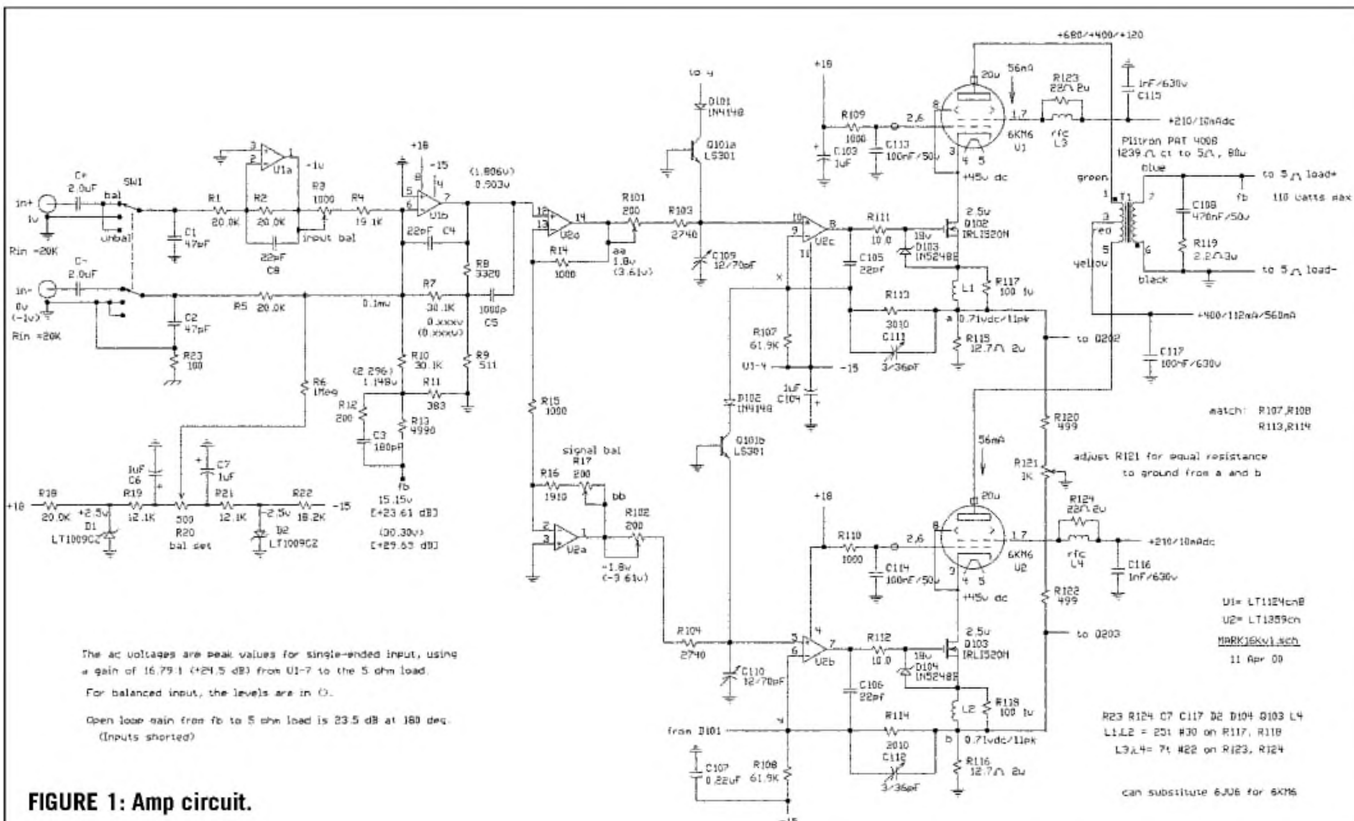


FIGURE 1: Amp circuit.

ages.) More capability is available if you go to the familiar TV horizontal output tubes ("sweep tubes") still available as NOS. These tubes are rated into the kilovolt range, and can pass as much as an amp, if you select the larger units. The 6KD6 and the 6KG6 (EL509) are probably at the top of this scale, with peak cathode currents of over 1A available, plus voltage ratings up to 7kV.

This surely makes enough reserve available to allow for a design to reach 150W. To prove this contention, one such amp has been on the market for several years, although it is no longer sold.[1] I turned my attention to the beam power horizontal output tube family.

And of course I ran headlong into the trouble with beam power pentodes: they have a screen grid, which draws current. In the cascode power stage I was using, this current would destroy the nice linearity of the cathode current by siphoning off a small and nonlinear part of the current, leaving the plate current waveform distorted.

There is another feature of the conversion of the cascode power stage to beam power: the tube screen potential will increase as the signal level increases, because of the active cathode voltage. Setting the quiescent tube screen voltage to +165 re the cathode means the screen supply DC level re ground is +210V. Therefore, the tube sees an increasing screen voltage as

AMPLIFIER CIRCUIT DETAILS

The circuit used in the accompanying article is very like the one in *GA* 5/98. The following is provided if you are interested in the small evolutionary changes in the design of this amp.

COMPONENT CHANGES

One improvement that seems useful is to increase the operating beta (current gain factor) of the transistors Q1 and Q2 in Fig. 13 of the *GA* 5/98 article. I replaced these two with the super-beta transistor Q101 in the new design. This part is an LS301 from Linear Systems—a dual matched-beta unit that has a beta of 2000, and is thus a nice part for use in this arrangement. With a very high beta, the base-current component is minimized, so the cross-connected correction currents in this circuit are more nearly ideal.

I have noticed many times the better performance that is associated with an op-amp circuit as an inverting amplifier—it helps to minimize common-mode problems. I decided to include this feature, and in the process was able to have the amp exhibit a constant 20kW input resistance at both of the balanced input terminals. The LT1124 operates well in this.

In many instances, the adjustment of pots R101 and R102 can be frustrating. Sometimes no amount of care leads to a happy minimization of the second and third harmonic components. It often appears necessary to make an adjustment of the high-frequency phase of these circuits. Only in this way can the right signal cancellations occur.

I found that small trimmer capacitors C109 and C110 help do this. Furthermore, the adjustment would sometimes appear willing to give a good second harmonic "tune," but require a different setting for a good third harmonic result. To break up this dependence, another variable seemed called for, so I also included the small trimmers, C111 and C112. This may seem like nothing but gingerbread in an audio circuit, but as it can turn out, just a fraction of a degree of high-frequency phase shift can lead to a much nicer result.

These parts, if you wish to experiment with them, show their effects when you are able to measure down to the –90dB harmonic level. (A calculation will show that for C109 = 30pF, there will be a phase shift of ⅒ degree at 3000Hz. It is remarkable that this small a correction should turn out to be important. This suggests interesting things for many other circuits. It might be that a lot of quasi-balanced designs could benefit from more careful attention to the high-frequency differential phase shifts in push-pull circuits.)

I changed MOSFET transistors Q3 and Q4 from IRF710s to IRLI520Ns. This allows for some extra headroom in the signal voltages at the U2c and U2d outputs, since these logic-level parts require less turn-on gate bias. This, combined with the rise to 18V of the 15V supply, gives yet more signal headroom, which seemed useful.

SNIVETS

Finally, there is a curious matter called "snivets." I first encountered this when I tried using the 6KG6 in the cascode amp. The technical data sheets on the tube mention "snivets" without giving a definition. Given the context of the '60s, perhaps losing this bit of knowledge is excusable.

This unfamiliar word comes from the TV repair jargon of the vacuum-tube TV era.[2] It refers to the appearance at the left margin of a TV raster, of a vertical black bar, or series of bars. This artifact of troubles in TV sweep tubes turned out to be caused by VHF oscillations of the horizontal output tube. The cause was usually the presence of some sort of resonant load in the tube's screen circuitry—maybe only an inch of wire, which can be associated with about 30nH of inductance. At 250mHz this looks like 47Ω. It only takes 13pF to tune this at that frequency.

Coupled with the tendency of the tube's screen current to develop a nonlinear relation with tube electrode voltages, the result could be a negative resistance region of screen or plate current action—a sure invitation to an oscillation. The trouble usually appeared at the left edge of the raster because it was at that point that the tube was driven into hardest conduction at the lowest plate voltage. If you happen to have a copy of the tube data on the type 6KG6, for example, look at the plots of screen current versus plate voltage—there is a very evident negative screen resistance region at plate voltages of 50–100V.

If you use such a tube in an audio amplifier, it can go wild in the VHF range anyway. Audio requirements might well see fit to drive it into that same region of low plate voltage. You might again be rewarded with a squawk from the FM tuner, or a meltdown in the amp. To avoid these problems, there are a few tricks worth knowing.

First, keep away from low plate voltage. That in itself is a success criterion (intentional or accidental) of some designs using the EL509 (6KG6). I pick the EL509 since it is one of the highest conductance sweep tubes used, and thus more likely to give RF problems.

Another good trick is the inclusion in the screen feed line of a series 20Ω 2W resistor wound with seven or eight turns of #20 wire. This forms a loss network for VHF that can reverse the negative resistance of the screen circuit. (This idea explains the presence of such parts in *Fig. 1*.)

Sometimes just replacing the 6KG6 with a different tube can reduce the VHF problems. Another ploy is to insert a series 1000Ω in the screen feed line, and bypass the power-supply side of this part to a good local ground with 1nF or more. This will, of course, lower the screen voltage with audio signal current components, thus lowering the power output.

Some combination of these tricks can usually alleviate the instabilities. The circuit of *Fig. 1* is probably less tolerant of VHF problems than other circuits, because of the earlier mentioned fact that the tube screen voltage is signal-dependent.

the audio signal drives the cathode toward ground.

The tube thus acts as a higher-conductance device for larger signal levels. This has some good and some adverse implications. The good news is the increased power capability—the peak current available is higher. The trouble might come because of the increased potential for RF instability. (See the sidebar on "snivets.")

REDESIGN

Figure 1 is the circuit diagram of the amplifier. The parts list is in *Table 1*. The amp is a monoblock design, using two 6KM6 tubes and the Plitron PAT-4008 toroidal output transformer, matched to the Plitron 454707 toroidal power transformer (*Photo 1*).

This is a slight redesign of the amp described in the *GA* 5/98 article. Some of the differences are mentioned in the sidebar. The output stage is the same (nearly) and produces the same action. This is visualized by the *Fig. 2* waveforms, which show the two output tubes' current pulses in response to a sinusoidal test input.

As described in the earlier article, adding these two currents to see the current delivered to the OPT results in a nice fully reconstituted sine, with no crossover problems. (This action was fully described in *GA* 5/98.) *Figure 3* shows the sum of the two waves of *Fig. 2* (solid trace), and, for reference, one of the *Fig. 2* traces as a comparison (the dotted trace).

Now you are in a position to superimpose the screen currents. *Figure 4* shows the measured screen currents required by the two tubes. As you can see, the currents are not of equal amplitude (a matter of tube inequality). Adding these together, as was done for the plate currents, gives the curve of *Fig. 5*.

This is the total signal that is siphoned off from the cathode current, and is clearly a source of distortion.

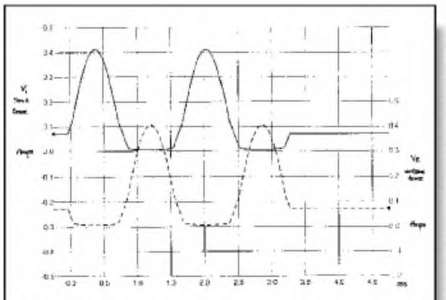


FIGURE 2: Tube cathode current.

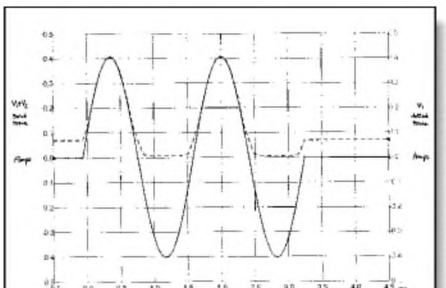


FIGURE 3: Tube currents combined.

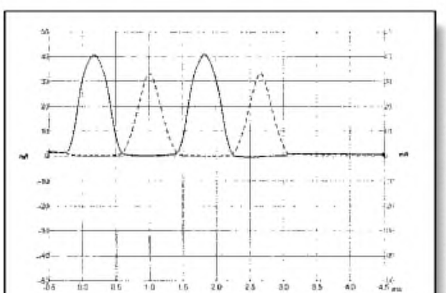


FIGURE 4: Screen currents.

TABLE 1
AMPLIFIER PARTS LIST

ALL SIGNAL PATH CAPACITORS POLYPRO AT 50V OR GREATER

ALL RESISTORS ¼W 1% UNLESS NOTED

SITE	VALUE
C+, C–	2.0µF/50V or greater, polyprop
C1, C2	47pF
C3	180pF
C4, C8, C105, C106	22pF
C5	1000pF
C6, C7, C103, C104	1µF/35V tantalum
C107	0.22µF/mono ceramic
C108	470nF/50V
C109, C110	12/70pF trimmer
C111, C112	3/36pF trimmer
C113, C114	100nF/50V
C115, C116	1nF/630V
C117	100nF/630V
D1, D2	LT1009CZ
D101, D102	1N4148
D103, D104	1N5248B
L1, L2	See circuit drawing
L3, L4	rfc see circuit drawing
Q101a, Q101b	LS301 Linear Systems
Q102, Q103	IRLI520N
R1, R2, R5, R18	20.0k
R3	1000 ¼″ square pot
R4	19.1k
R6	1Meg
R7, R10	30.1k
R8	3320
R9	511
R11	383
R12	200
R13	4990
R14, R15, R109, R110	1000
R16	1910
R17, R101, R102	200 ¼″ square pot
R19, R21	12.1k
R20	500 ¼″ square pot
R22	18.2k
R23	100
R103, R104	2740
R107, R108	61.9k
R111, R112	10.0
R113, R114	3010
R115, R116	12.7 2W
R117, R118	100 1W
R119	2.2 3W
R120, R122	499
R121	1k ⅛″ square pot
R123, R124	22 2W
SW1	dp3t rotary panel switch
T1	Plitron PAT 4008
U1a, U1b	LT1124cn8
U2a–U2d	LT1359cn

TABLE 2
POWER SUPPLY PARTS LIST

ALL RESISTORS ¼W 1% UNLESS NOTED

SITE	VALUE
C117	100nF/630V
C201	0.1µF/100V
C202	470µF/100V
C203–C206, C216	1µF/35V
C207–C211	470µF/250V
C212	33µF/450V
C213	0.1µF/400V
C214, C215	100nF/50V
D201	1N4747A
D202	1N4746A
D203, D204	1N4754A
D205, D206	1N4148
D207–D210	4 each Hexfred diode
D211	1N4737A 7.5V
DB1	DB104 Digi-Key
F1	1.2Asb
F2	1.25Afb
L101	0.5Hy/0.3A
Q201	IRF710
Q202, Q203	ZVN2106A
R201	20k
R202	18.2k
R203, R204	51
R205	4990
R206	51Ω surge limiter
R207–R210	51k 1W
R211, R212	140
R213, R214	39k 2W
R215, R216	200 1W
R217, R218, R221, R222	301k
R219, R220	14.7k
R223, R224	15.0k
R225, R226	10.0k
R227, R228	10.0
R229, R230	500 ¼″ square pot
SW1	spst toggle
T2	Plitron 454707
U201	LT1354cn8
U202	LT1492cn8
reg1	78M18
reg2	79M15

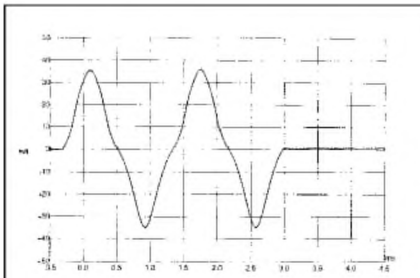


FIGURE 5: Screen currents added together showing the nonlinear shape of the screen currents which are stolen from the plate currents unless corrected for.

The beam power version of the amp should noticeably be worse for having this done to it! An amplifier made with this circuit using beam power tubes would require some correction for it to work in the league with the triode version.

WHAT CAN BE DONE?

My answer is to measure the screen current of each tube, and to force just that much current into its cathode again, so that in effect, the cathode current is "pre-distorted" with the exact signal necessary for the plate current to be error-free. If this sounds like positive feedback, you're right. But there is no problem with instability, because the gain in such a circuit is much less than unity.

The answer in circuit design parlance is given in *Fig. 6*, the power supply for the amplifier. The parts list is in *Table 2*. In the lower right, you can see this circuit includes a dual op amp, two small MOSFETs, and some resistors. The screen currents flow to the tubes via the 200Ω resistors R215 and R216. The drop in these resistors is used to drive the op amps in a local feedback loop, which adjusts the current in the 10Ω resistors R227 and R228 to be equal to the current in the screen feeds. This is so because the resistor attenuators R217–R224 have a 20:1 loss, which is the ratio between the 200 and the 10Ω parts.

To adjust for slight op-amp offsets, you should set pots R229 and R230 to bring the idle current through the 10Ω resistors to about 1mA, which is easily done by making this adjustment under no-signal conditions. (Alternatively, you can measure the no-signal drop in R215, divide this voltage by a factor of 20, and adjust R229 to set the drop in R227 to this number. This will set the R227 resistor current equal to the screen current value in V1. Then do this same procedure for R230.)

I arranged to connect these recovered screen-current signals back into the tube cathodes by tying the drain leads of Q202 and Q203 back to circuit points "a" and "b." The FET drain currents thus flow from "a" and "b" to ground in parallel with the R115 and R116 currents, enhancing the tubes' current by the correction amounts.

The circuitry of the output stage normally operates with +0.7V across the resistors R115 and R116 (i.e., at points "a" and "b" on *Fig. 1*). This is plenty of potential to operate the MOSFETs Q202 and Q203, which at no-signal draw only a milliamp or so. As the power level rises, points "a" and "b" rise to as high as 10 or 11V, when the screen currents reach about 50mA, which requires only 0.5V at the MOSFETs' source terminals. The signal headroom across Q202 and Q203 is thus always enough.

SMART SCREEN SUPPLY

Now, how can you tell whether the connection of this "smart screen sup-

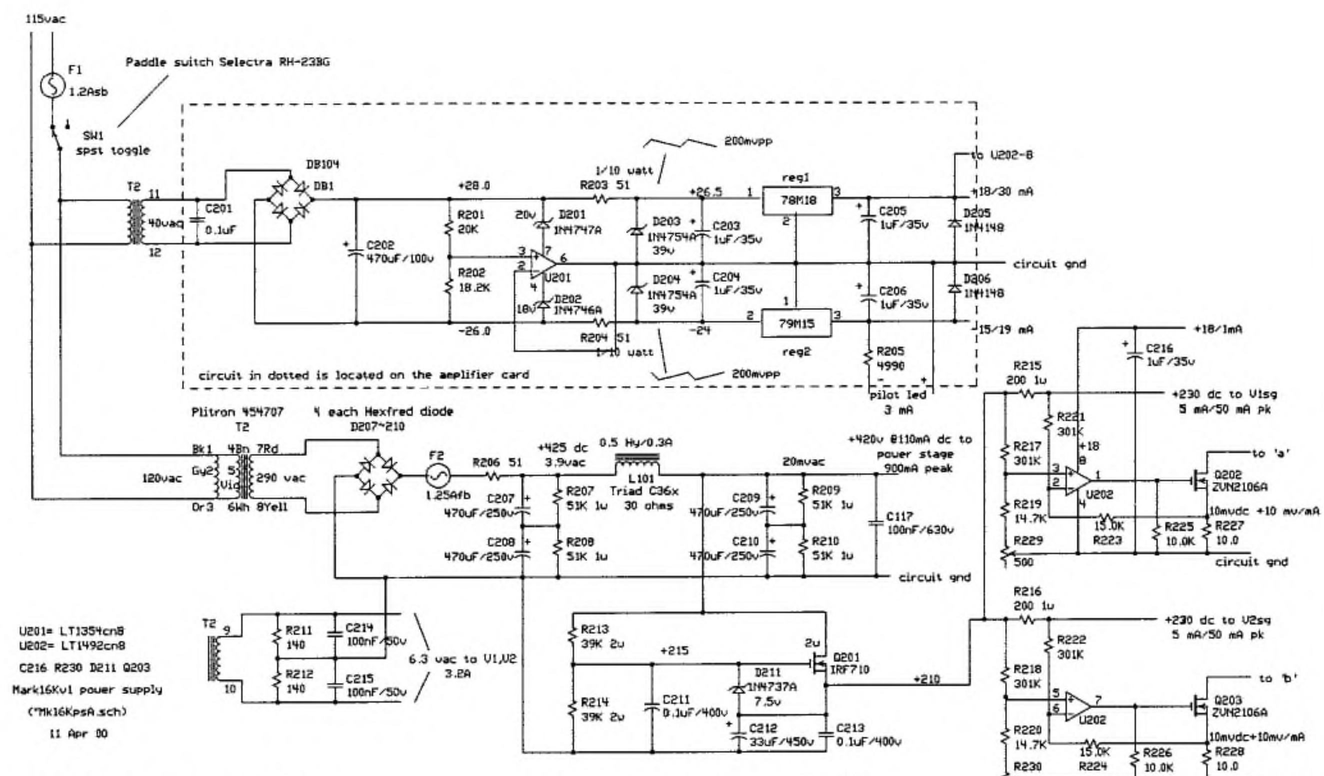


FIGURE 6: Power-supply circuit.

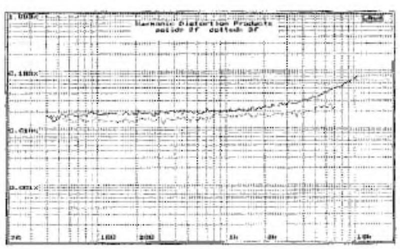


FIGURE 7: Harmonics without smart screen supply.

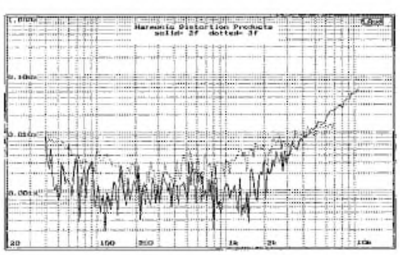


FIGURE 8: Harmonics with smart screen supply.

ply" is really helpful? You can estimate the residual nonlinearity of the circuit by measuring the harmonic distortion of the amp output, both with and without the correction. The easiest way to do this is first to re-

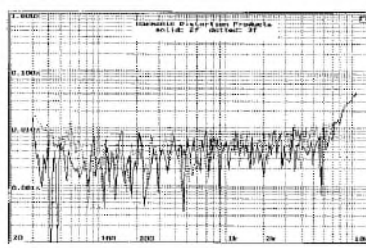


FIGURE 9: Harmonics with phase-corrected screen supply.

move op amp U202 from its socket. This de-activates the screen current correction entirely, as resistors R225 and R226 will pull Q202 and Q203 into cut-off.

In this condition, the amp gives the harmonic plot of *Fig. 7*. I then reinstalled U202 and ran another curve (*Fig. 8*). Clearly, there is a nice improvement. In fact, this result puts the amp in the class with the triode version (see Fig. 18 of the *GA* 5/98 article). (The reason for the increased "noise" in *Fig. 8* is the much lower level of signal processed in the computer analysis—the round-off errors in the signals are a larger fraction of the lower signal amplitudes.)

The high-frequency region of *Fig. 8* shows a rise of harmonics with rising frequency, as is so often the case. I supposed that there might be a high-frequency phase error in the screen current correction that could be improved. A small capacitor (12pF) connected across R221 did help this, with the result shown in *Fig. 9*. This seems to have been a good idea, and now the result is quite acceptable, leaving a design that allows the use of beam power tubes without their usual error habit.

The tubes I was using were 6KM6s, which are somewhat smaller power devices than the EL509 mentioned earlier. The 6KM6 is able, however, to provide a plate swing of 280V peak at 900mA peak, which results in a peak power output of 252W, or 126W average. This seems a nice reward for the conversion to sweep tubes. ❖

REFERENCES

1. The Nestorovic Labs model NA-1 rated at 150W.
2. The details of the "Snivets" story have been supplied by W.F. Santelmann, by e-mail.

LP Transfer to CD

This article introduces the art of transferring your LPs to CDs in high-quality form via a personal computer, so that you can edit out the ticks and some distortion, and access tracks with your CD player's remote. **By Victor Staggs**

Computer users have discovered that it is practical now to transfer their LPs onto their computers' hard drives and then to convert them into MP3 files for transfer into an MP3 player, or to record them onto compact discs. Depending on the quality of sound you desire from your LPs, this may be quite time consuming, so what are the motivations for doing this? What kind of quality can you expect in a transfer to CD, and is all this work worth it? I shall attempt to give you a taste of this process so you can decide for yourself whether this kind of audio activity is for you.

WHY DO IT?

If, like some audiophiles, you have a substantial collection of LPs, you might consider it worthwhile to compress all the music on them into MP3 format so that you can squeeze ten hours of music onto a CD, and play it in an MP3-compatible player. There will be a noticeable loss of audio quality in this form, but it makes all those LPs much more accessible.

Or, as I have done, you might wish to do a high-quality transfer of selected LPs to the CD format, one LP per CD, so that you can edit the sound and get rid of annoying ticks and pops in the process. With perseverance, you can make a CD transfer that sounds as good as a commercial CD, or so close that you won't care about the difference.

The high-quality transfers won't save you much space, because now you will have CD jewel cases instead of LPs. But they will sound less noisy, and you will be able to use your remote control and switch between tracks at will without putting your vinyl at risk.

Moreover, many fine recordings on the reissue brands have not survived into the CD era, because their brand owners either discontinued the label or have begun a process of rerecording digitally to start all over again with a new catalog. So, some of your treasured recordings simply will not be available on CDs unless you so record them yourself.

REQUIRED RESOURCES

Obviously you will need a computer with an audio-digitizer card inside it, or an external box with an umbilical interface cable. There is an entire website devoted to resources for processing LPs via PC and Macintosh computers at http://www.tracertek.com, including better-quality digitizing cards and software for audio editing. Hollywood editors generally shun these, preferring the expensive converters from Digidesign® or Apogee Digital®. However, these products cost as much as a substantial new CD collection, so most audiophiles are content to use the mid-grade digitizers.

A stereo file 72 minutes long takes up 650 megabytes (MB) on your hard drive at 16-bit 44.1kHz resolution, as you know from the CD format. In addition, during editing your software will produce copies of your tracks as you make changes to them, so that the hard drive may require 2 or 3 gigabytes (GB) to produce a finished CD. This is fairly small compared to modern hard drives, which are capable of storing home movies in MPEG-2 (DV) format occupying 13GB per hour.

If you wish to configure your computer for audio editing, you can install a two-drive RAID (redundant array of independent devices) using a SCSI 160 or ATA/66 (or ATA/100) controller and appropriate RAID software. This will speed up the editing and file-saving process considerably. In addition, a fast CPU is always welcome for filtering, which you will definitely need. My rig is a PowerMac® 8600/300, which is a RISC machine equivalent to a Pentium® 600. I have equipped it with an 18GB SCSI RAID that can make continuous transfers at 50MB/second, and peak transfers (cache hits) at 160MB/second.

Your LP turntable will need to be in good shape, because software is incapable of removing wow and flutter. If you spend more than an hour or two editing an LP file, you should make sure your turntable system is in the best possible shape so you will not need to redo the editing on a new digital transfer of the LP.

THE CD SOUND

In my case, I used my PowerMac's built-in digitizer, a device made by Apple to record sound and to digitize analog video into MPEG-1 (Quick Time®) format. It has the best of the computer digitizers of its era, even though it is not

ABOUT THE AUTHOR

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

on a par with an Apogee® converter, so I was able to produce CDs that sounded about 99% like the LPs, and that were superior in the sense that I could edit out ticks and certain kinds of distortion. Now I am very happy to be able to play these LPs as CDs, and to better enjoy some of those old bargain-brand LPs that contained great music but quite a few ticks.

The computer's digitizer does impart a noticeable flavor to the sound, but you can minimize this effect with judicious filtering so that only a slight loss of harmonic subtlety is the price of conversion. The preservation of acoustic space is remarkable. I cannot comment on the quality of other digitizers, so you must try your own and see how the result sounds to you. I will give hints later on to help you optimize your results.

In general, a good CD transfer of a quality LP—if properly edited and filtered—will sound much better than a mediocre brand of CD that was recorded and mastered entirely in the digital domain. With digital formats as with analog, the results of commercial recording depend on the skill of the engineer as much as on the quality of the hardware.

THE BEST LPS TO RESTORE

The beauty of doing your own restorations is that you can pick the LPs into which you wish to breathe new life. This will depend on how you regard the kind of music they contain, and how much you like to listen to particular ones. However, experience will soon teach you that it pays to be selective about which LPs you transfer.

The main issue, after you decide on your musical favorites, will be the practical one of how difficult it is to get adequate CD sound out of the transfer. If it is too difficult, you might as well care for the LP and play it as is. It is not practical to transfer bargain LPs that were pressed from noisy vinyl containing too much badly reground material, since the background noise will be intrusive when played from CD and it will be difficult to remove it without compromising the sound of the music. So because you didn't pay much for your bargain pressings anyway, you might as well stick to restoring the better-quality pressings.

There were at least three generations in the history of mastering LP records (I shall ignore the 78s). The first generation used a sapphire stylus to cut a lacquer master disc, which was then used to produce stampers through a process of plating and producing negative masters, interim positives, and negative stampers. This kind of mastering resulted in a fairly noisy groove that would be hard to transfer to a CD. The sound is even worse if a foreign label sent a tape copy to the USA for remastering, since then there is a copying loss in the master tape.

The next generation used an RF-heated sapphire stylus to do a better job of mastering on the same kind of lacquer master as the first generation. The lacquer masters were actually a lacquer coating on a dead-flat aluminum blank. However, the lacquer deposition method left a small irregularity on the otherwise shiny surface that still produced some background noise in the final pressings.

The last generation was the copper mastering process (DMM®-Direct Metal Mastering) developed by Teldec. This used an ultrasonic signal mixed with the audio signal to vibrate the stylus when cutting a master disc with a copper surface. Since the copper blank was extremely smooth, this resulted in a very quiet surface in the vinyl pressings.

When you think about it from the perspective of the digital age, the audio industry expended great financial and intellectual resources to perfect the art of mechanical analog recording to improve the sound of the final product delivered to the consumer. Each step along the way demanded devotion and fanatical precision on the part of the craftsmen who produced the consumer media for playback.

Nowadays a laser pickup from an optical disc takes care of all the hard work for us, so we are left with miking, mixing, and software with which to influence the final result.

INTRUSIVE HUM

When you make your first recording onto your hard drive, you will quickly notice the presence of intrusive AC power-line hum, and if you use a spectrum analyzer program to look at incoming "silence," you will see other frequencies as well. In my case the power-line hum consisted of 60Hz and all of its harmonics up to 540Hz. Also evident were frequencies related to the refresh rate of the computer monitor.

After some experimentation, I discovered that the culprit was the monitor, so I isolated it via an isolation transformer. My 15″ monitor required a 100 volt-amp transformer (this roughly translates to 70W), so you might examine your monitor's power requirements and choose an isolation transformer accordingly. It did no good to float the phono preamp on an isolation transformer.

In fact, the last part of the setup was to connect a cable with alligator clips between the ground post on my phono preamp and a metal part on the chassis of my PowerMac 8600. This reduced the hum to inaudible levels, until it was only slightly greater than that produced by connecting the computer's audio output back into its input, and recording silence.

Audiophiles are used to flipping AC plugs (if they are not polarized) to minimize AC hum. This worked with the preamplifier plugged into the wall, and the improvement was 3dB. This is difficult to check without a spectrum analyzer, and it was not a huge improvement.

So the final setup was my LP player plugged into my phono preamp, the tape output of the preamp plugged into the audio input of the PowerMac, the monitor isolated on a transformer plugged into its own wall outlet, and the preamp-turntable combination plugged into another wall outlet. The computer was on an outlet-strip/transient-suppressor plugged into its own wall outlet. The turntable ground cable and the alligator clip were both attached to the preamp's ground post, and the other end of the alligator-clip wire was clipped to a PCI slot cover on the back of the PowerMac.

THE PHONO SETUP

Since I was mastering Baroque music for the first try, I knew that I would not need to worry about preserving killer bass, so I switched in the IEC filter in my phono preamp's RIAA stage. This rolls off the bass below 20Hz, which helps to keep rumble out of my electronics and out of my audio file. Even so, some rumble frequencies were noticeable on my computer screen, though they were too low in frequency to be audible through my speakers.

I am a devotee of Teflon®-insulated wire consisting of a twisted pair inside a shield, both for the wire connecting the tonearm to the phono preamp, and for connecting the tape output to the computer. The PowerMac 8600 has RCA jacks for audio, which made it easy to press my audio interconnects into service. The Teflon-insulated wire made an improvement in sound over a generic cable, even when making a test recording from an FM tuner, so I recommend using the best cables you have.

Of course, you have hand-tweaked the tracking force and the sideforce compensation to get the best sound out of your LP player, haven't you? If not, this is the time to do it. You should do this while the LP player is installed in your stereo system, because the monitoring electronics built into your computer are not of very high resolution. The Shure test record for its V15-V cartridge is useful for this.

RECORDING SOFTWARE

Your editing software should let you control the recording of sound onto your hard drive. I used MacroMedia's SoundEdit 16® software, which, though a bit dated now, does almost everything I need. If you buy Roxio's Jam® recording software for mastering Red Book Audio CDs, you will receive a copy of Bias Peak LE®, which you can use for recording also. Generally you will see buttons on your screen, looking like a tape transport's controls, that start, stop, pause, and rewind your files. You will also see a bar-graph level display, which is excellent for setting the audio recording levels.

Phono preamps connected via a tape output have a fixed gain in the 30–36dB range (more if for moving-coil cartridges), so you cannot use them to set the recording level. Usually the computer's sound card will have a hardware level setting feature that you access via a slider of some sort on the transport controls you see on your screen. If the signal level increases above some nominal safe level, yellow or orange bars will appear on the bar graph to warn you. If you come within an ace of saturating the A/D converters, or if you go over the top, a bar will turn red and leave a red marker on-screen for a few seconds to make sure you see it. Some displays will leave a permanent red marker on the screen to let you know that you went over the top somewhere in the recording.

You will need to play what you think are the loudest portions of your LP and set the input level on the computer so that the red warnings either don't appear at all, or appear only briefly. Evidently these bar-graph displays give you some safety margin. Even if you do get a bit of red, you can examine the waveform of your recorded file and check to see it is not flattened on top, and that you have some headroom left. You should preserve about 3dB of headroom in your file in anticipation of later filtering.

Another aspect of level-setting is that if your digital file often comes very close to 0dB (the top)[1] it may sound harsh in some CD players because their electronics are vulnerable at these levels. Your computer probably will be the worst offender in this regard, so you really must leave at least 3dB of headroom.

Being a die-hard LP fanatic, I am careful to use a Discwasher® brush and D4® cleaning solution to make sure the record is dust-free before recording. In addition, since this usually leaves a static charge on the vinyl, I discharge it with a Zerostat® antistatic gun. I noticed that if I was careful to discharge my vinyl, I ended up with many fewer mysterious-looking ticks in my audio waveform, and the files were easier to edit.

So the final step was to put the stylus in the groove and start the recording software to do a transfer to my hard drive. It goes without saying that you will need a recording situation that is not vulnerable to the sound of footsteps being transmitted into your phono pickup. Once I got my file onto the hard drive and started editing, I discovered that my practical education was just beginning.

WORKFLOW PLANNING

To do a successful job of editing LP audio for CD replay, you must plan in advance. Commercial CDs usually contain dead silence before the music starts, and Jam will complain if you do not include at least two seconds of it before the music starts. However, it is more problematic to consider the sound between the music bands, between LP sides, and after the final cadenza.

My treatment consisted of forcing the lead-in groove to two seconds of dead silence via editing. But if you fade to silence between tracks, the change in the way the rumble sounds will be obvious, so it is better to leave the rumble between bands intact. The only treatment I attempted between bands was to low-pass-filter the rumble, which helps the quality of the presentation without being obvious. The only way to make this work is to preserve all of the "silence" between bands so as to avoid introducing any discontinuities by editing some of it out.

The sound between LP sides is more of a problem, since it will not be continuous between the end of the last band on Side 1 and the beginning of the first band on Side 2. In this case you can low-pass filter the rumble and fade it into silence at the end of Side 1, and then start Side 2 just as you started Side 1, with dead silence.

If you wish to play your CDs in a CD jukebox, you should leave the three bands that make up a concerto (for instance) together so there are no CD track breaks between them. Then, if you play the concerto directly, or by a random-play feature, you will hear the whole thing instead of an isolated movement. For this kind of access, you will need dead silence at the beginning of the concerto, and low-pass-filtered rumble between movements and at its end, followed by a fade to silence.

When low-pass filtering the rumble between bands, you must recombine separate files that you may have produced with individual bands on them so that the low-pass filter will not encounter any discontinuity in the rumble. You can separate the bands into individual files again after filtering the rumble.

In the case of my 8600, I found that I needed to somewhat equalize the frequency response of the sound of the whole LP to neutralize the characteristic flavor that the hardware imparted. The software I used for this, Pro Tools Free® by Digidesign®, is very cumbersome to use on individual LP bands, so it would be wise to carry out this step on files consisting of entire LP sides before cutting them into individual tracks for CD access. I will say more about equalization later.

After applying any overall equalization to your sides, you can separate them into individual files corresponding to the album-band information, using the timing on the LP cover or the center label as a rough guide to finding the edit points in your files. These files will be the tracks to which Jam will assign access points to when you record them onto a CD, so you can skip around with your remote control.

WHY EQUALIZE?

After recording a few LPs onto CDs and listening to them through my stereo system, I realized that my PowerMac

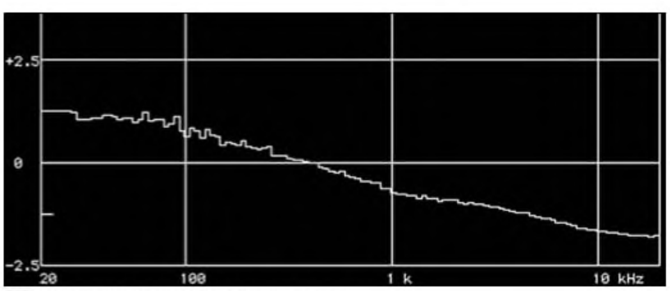


FIGURE 1: "Linear slope" equalizer.

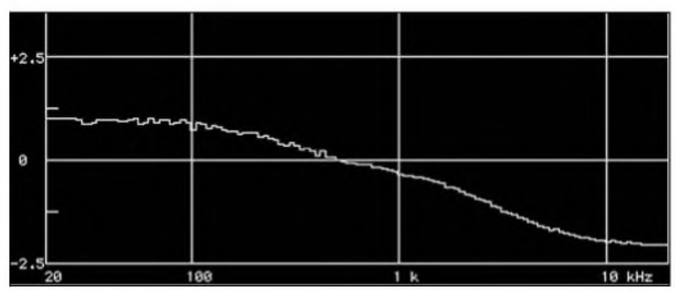


FIGURE 2: Equalizer for PowerMac 8600.

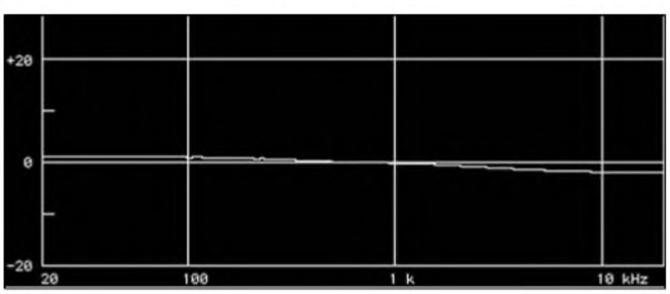


FIGURE 3: Equalizer for PowerMac 8600.

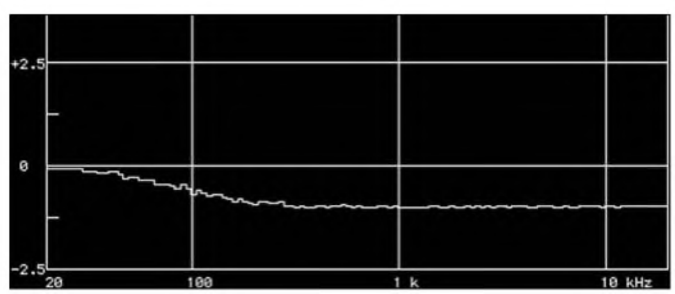


FIGURE 4: Second equalizer for PowerMac 8600.

8600 was imparting a characteristic flavor to the LP sound that was unflattering. I believe that most of this is due to the use for the sound input of surface-mount-device (SMD) capacitors on the motherboard, which are likely to be some form of electrolytic capacitor, rather than a high-grade film capacitor as used in high-end audio.

I admit that the sound was not as obviously tainted as it is from many mid-fi stereo receivers, which are filled with aluminum electrolytic capacitors, but it did seem overly bright at the top and lacking dynamic energy in the low frequencies.

Theoretically, it first seemed that an equalization with a linear rolloff of 3dB from the lowest audible frequencies to the very top might be ideal for this. After all, the frequency response of many movie-theater loudspeakers consists of a 5dB rolloff from 40Hz to 16kHz[2], and my curve would be a milder version thereof. Such a curve, as best approximated by combining three of the equalizers built into Pro Tools Free, is shown in *Fig. 1*. But it is much more practical, and sounds better, if you use only a two-band equalizer to achieve the curve shown in *Fig. 2*. I obtained these graphs by measuring the frequency response of the Pro Tools Free filters using pink noise, so they are not perfectly smooth.

Before purists complain about the radical shape of the curves, they should refer to the same curve as in *Fig. 2*, presented in *Fig. 3* on the same dB scale that would be used to display the effect of tone controls. It is really pretty mild.

BASS-LIFT COMPENSATION

The 1dB bass lift compensates for the lack of energy at low frequencies, and since it flattens out below 80Hz or so, the rumble frequencies are not unduly amplified. In addition, the brightness of solo instruments is preserved, while any biting quality of the high frequencies is civilized into subtlety above 8kHz.

The tricky part of this equalization is that you can't hear much of it through the monitoring electronics in the PowerMac 8600—the headphone amp is too crude dynamically, and I was listening through some quite smooth Grado SR60 headphones. So it was necessary to process the same LP band several times with different EQ settings in Pro Tools Free, burn the files onto a CD, and then pick the best settings by listening to the CD through my stereo system. I was quite astonished to find out what an excellent restoration was possible by picking the right EQ curve, since capacitor-induced distortions are nonlinear, and equalization is a linear process. This is the best you can do in working with real-world audio.

About halfway through this project I fine-tuned the antiskating (sideforce) compensation on my Shure/SME arm using the Shure V15-V test record, and I refilled the damping trough with new silicone fluid from SME in England. This wrought a great cleaning-up of the LP sound, and I settled on a new EQ curve (*Fig. 4*). This curve abandoned the high-frequency rolloff and boosted the bass by only 1dB. In addition, with the arm's fundamental resonance well damped, I switched out my preamp's IEC filter and left the RIAA curve flat below 20Hz.

Even though this was the best I could do, it was apparent that some natural dynamics of plucked harp strings were being lost, and that the bass boost was not compensating perfectly for this. Even professional recording engineers must compensate for the capacitors built into mixing consoles and preserve whatever dynamics they can, but since I was not sure whether the loss was due to the PowerMac or simply to my CD player, I left the EQ at 1dB bass boost.

You must accompany equalization by redithering the sound file, which I shall explain later.

WHAT MAKES RECORDS TICK?

Vinyl LPs are pressed between two stampers with microridges that will form the microgrooves. Any small, hard impurity in the vinyl compound will result in an irregular surface in its vicinity if it comes into contact with a stamper's ridge. When a phono pickup stylus traces this shape, it produces a wild but very short peak in the waveform, usually confined to one channel because the mechanical irregularity occurs in just one groove wall. Occasionally you will see a peak in the waveform in one channel plus its mirror image in the other that is caused by an impurity in the bottom of the groove or by a

FIGURE 5: A tick waveform, nearly full amplitude.

dropped tonearm.

The leading edge of the peak generally is a complete discontinuity, with the voltage swinging up hugely in just one digital sample interval. Such a peak is shown in *Fig. 5* (this one swings down first).

As previously mentioned, a small static discharge occurring during the playback of an LP charged up by cleaning, but not discharged before playing, can produce what looks like an impurity-caused discontinuity. But when you discharge the LP and rerecord it, you will find that the discontinuity has disappeared. These flaws can be minimized by simple care during recording.

Another cause of discontinuities in the waveform is mistracking of the stylus in the groove. This is caused by early LP masters being cut with styli not faceted to conform to modern standards for playback by elliptical styli, nor perhaps to any standard at all. Such distortion seems to make itself evident during end-of-side cadenzas, although I have heard it in violin tracks at the very beginning of an LP side.

Removing ticks and other discontinuities requires the most time, persistence, and creativity on the part of the editor. It is at once the most frustrating and the most rewarding part of the process, at least after you are finished and you find out that your LP really sounds much better than you imagined it could.

HOW HOLLYWOOD DOES IT

Hollywood sound-editing studios remove imperfections from recorded material with sound-editing software, particularly from recordings made using portable equipment at on-location events. Even professionally done master files contain ticks due to grounding problems or to static discharge, and you must remove these before the material is mixed for broadcast.

In this situation, there is no second recording that would provide material to replace the obviously damaged portions of a file. Therefore, it is up to your skill and ingenuity to fake in the proper waveform so that even an experienced listener will never detect the edit.

The usual tool for this is software with a waveform editor. SoundEdit 16, a semipro program, displays a sound waveform as a wiggly line on the screen. When you zoom in to maximum resolution on the time scale (*Fig. 6*), the display changes, representing the samples as small discrete diamonds that can be pushed up and down in amplitude by clicking and dragging with the mouse.

This is a crude but highly effective way to rework an obviously flawed waveform so that the new wave blends well into the surrounding material. It is useful mostly for ticks that do not disturb more than a few tens of pixels, because moving them around on your screen is picky business. This kind of editor is called a "fatbits" or "fatpixels" editor.

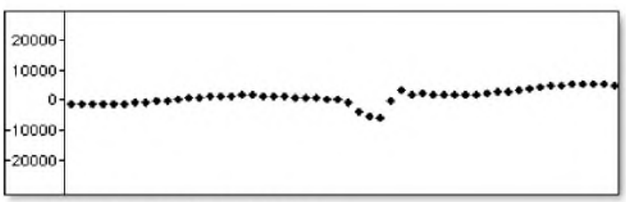


FIGURE 6: A click in fatbits mode.

You can zoom out and look at the adjacent waveform to get an idea of how to fake in the repaired waveform. Sometimes you must add small wiggles to continue high frequencies present in the surrounding material. Since the repair is only a few tens of samples long—amounting to a few units of 0.23ms—the ear will not notice that any repair has been made. *Figure 7* shows a click in a periodic waveform that I edited in fatbits mode, resulting in the restored version shown in *Fig. 8*.

MORE TRICKS AVAILABLE

More problems result when the damaged area is on the order of 50–100 samples or so. No one has the time to repair this with a fatbits editor, but in this case, there are still tricks to try. Often one channel will resemble the other quite closely, and only one will be damaged, so you can copy a selection from the good channel and paste it into the damaged one. Even if you need to adjust the gain of the pasted-in material and slide some of its pixels around in fatbits mode, this is a very time-effective repair.

One feature of music that facilitates editing is that instruments spend some time resonating or otherwise producing a continuous periodic waveform. In this case, you need only copy a selection consisting of one complete period and paste it into another similar period that is damaged. This kind of repair is justifiable even on theoretical grounds, and I defy anyone to tell when it has been made.

You can even find part of a waveform having good material to use in repairing a defective one, but where the good

FIGURE 7: A click in a periodic waveform.

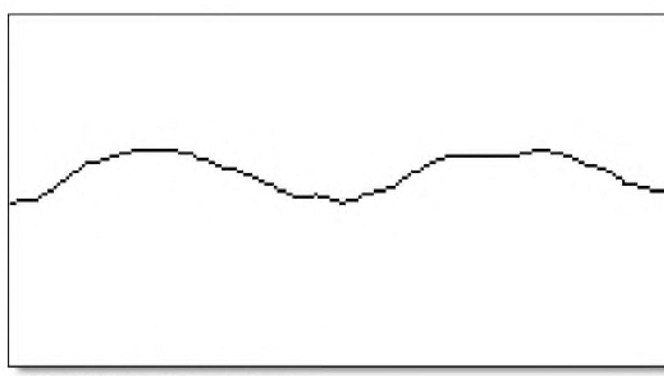


FIGURE 8: The click restored.

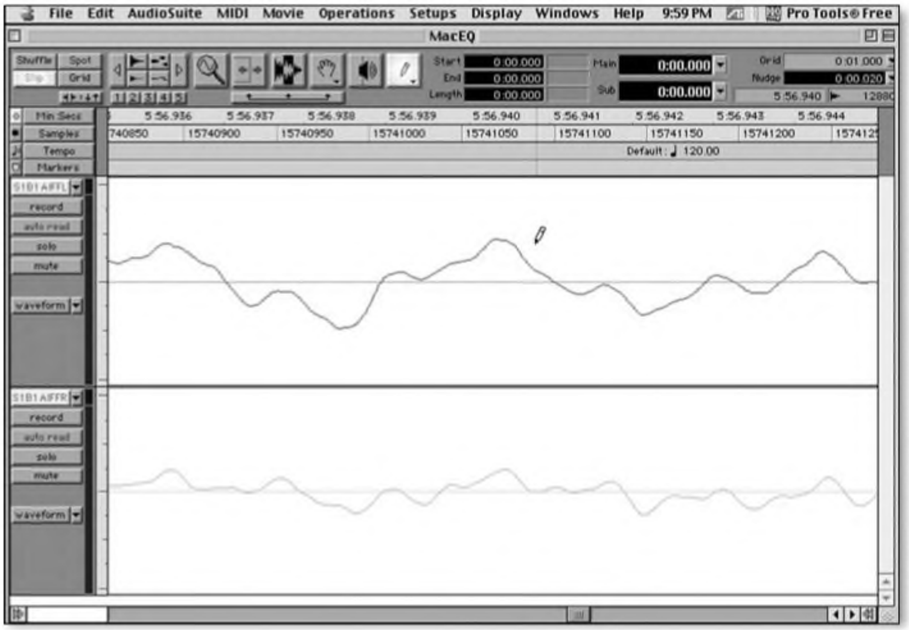


FIGURE 9: A screen shot of Pro Tools Free's edit window, with the pencil poised to draw.

waveform is flipped over on the time axis. It is easy to copy and paste the good section into a new document, reverse it in time, and then paste it in to patch the defective one. This is another theoretically justifiable edit.

The only way to preserve the stereo effect between channels is never to change the relative number of samples between channels by editing. Whatever you paste into a channel must match exactly the number of samples you are editing out of it. This also preserves the rhythm of the music.

My approach to editing a music file has two areas of focus. First, I edit the supposedly silent areas between tracks to remove large ticks before low-pass-filtering them. I even copy and paste these portions into new documents so that I can amplify them by 800%, edit them, shrink them back down to their original amplitude, and paste them back into the master document.

Next I tackle the music itself. Selections of music tracks containing low-amplitude sound, such as soloists, can also be copied into a new document, amplified by 400%, checked for ticks and repaired, and then shrunk back to normal amplitude and pasted back in.

HIGH-AMPLITUDE MUSIC

You cannot temporarily expand normal or high-amplitude music in a new document, so you must listen carefully with your monitoring setup to detect imperfections. Usually when you hear one, it will not be apparent in the corresponding waveform display, so you must zoom in on the small selection that ticks to make the discontinuity visible. Then you can use trickery to edit it out.

The PowerMac 8600's headphone output connected to Grado SR-60 headphones did not readily reveal ticks, and I could hear some in my stereo system after I had supposedly removed them all and burned a CD. So, I suggest that you hook an amplifier and a pair of near-field speakers with good tweeters to your computer's sound output and use them for monitoring.

You will become adept at this kind of editing after discovering and eliminating a few ticks and determining the "before" and "after" quality of the sound. Not all ticks are large-amplitude, but they impart a mushy sound to the music, which will improve greatly if you discover and edit them out. You can clean up mistracking distortion just like vinyl clicks if you discover it.

Flaws in sound, whether they are hum, harmonic distortion, ticks, or mistracking distortion, can have a subliminal effect on your perception of the music even if they are small. So as you develop skill in removing small flaws, your listening impression of the music will become more agreeable.

Higher-end software, such as Bias Peak and Pro Tools Free, includes a waveform editor that does not force you to move pixels around on your screen one at a time. Instead, they provide a tool that turns the screen cursor into a pencil, letting you redraw a waveform into a convincing curve without zooming in to fatbits resolution. Pro Tools Free also low-pass filters your new curve when you release the mouse button so that the repair blends in imperceptibly. I have not used full Bias Peak, so I do not know how its pencil tool works, but it does have one. *Figure 9* is a screen shot of Pro Tools Free's edit window, with the pencil poised to draw.

Once you have taken care of the ticks in your file, you can consider running a high-pass filter (above 20 or 30Hz, for instance) to filter out subsonic rumble if it is a problem. Doing this before removing the ticks will make them harder to edit.

DIGITAL SCIENCE

Digital science is a cousin to "rocket science" for reasons I explain now. You may have noticed that the only editing for which I recommend using SoundEdit 16 is for repairing ticks, low-pass-filtering the "silence" between LP bands, and temporarily changing the gain by powers of 2 (2, 4, 8—or 200%, 400%, 800%). This was to avoid performing a floating-point operation on any selection of audio with a substantial amplitude for a substantial amount of time.

Changing amplitude up and back down by powers of 2 (multiplying or dividing) is equivalent to sliding the bit pattern in an integer word of sound data to the left or the right, without introducing any new data or losing the old data off the end of the register. This preserves all the digital data intact, except for individual samples that you edited.

If you do any operation not involving a simple power of 2, then you might be performing a floating-point operation on the sound, which forces some of the data to occupy positions below the "decimal point." Generally digital filtering is done with floating point arithmetic, which adds a decimal point and many fractional digits. But sound data is stored as integers, and programmers must carefully convert floating-point registers back to integers to preserve the sound quality.

I have experimented with processing 16-bit sound files using 64-bit floating-point arithmetic and converting back to 16-bit integer sound-data storage by writing my own program in the C language. The only way to preserve the sound quality is to add a good form of dither, such as triangular-probability distribution-function dither with a 2-bit amplitude. This produces as much as 20-bit effective resolution with 16-bit files, according to Blesser and Locanthi[3], and, believe me, you won't like listening to any file that has only 16-bit effective resolution or less.

Pro Tools Free lets you add to your tracks processing inserts that do all the frequency equalization and amplitude adjustment in one pass, adding the dither at the same time, so that you can do a 16-bit-integer-to-16-bit-integer process with floating-point arithmetic that preserves 20-bit audible resolution. If your equalization software cannot add dither when taking processed data out of its floating-point registers and requantizing it, then the sound quality of the file will be ruined permanently. Do your experiments on an expendable copy of your sound recordings!

Most CD purchasers are aware that the sound is in 16-bit format, but few realize that the audible resolution attained is actually about 20 bits, achieved by various clever forms of dithering when going from the analog (or floating-point) domain to the integer digital domain. This is an area where the skill of the lacquer-mastering engineer has been replaced by the ingenuity of the mathematician and the electrical/software engineer.

This discussion points out that some software will be good for some tasks but not for others, as in the case of the dated but still useful SoundEdit 16. It can handle everything but subtle equalization on full-amplitude audio files, an area where more modern programs such as Pro Tools Free or Bias Peak shine.

THE MUSICAL TRUTH

I have been discussing the restoration of LPs to CDs as an attempt to preserve the musical truth recorded onto the LPs via various forms of digital-editing trickery and equalization. I have not suggested noise reduction, which introduces some audible artifacts and alters the low-level dynamics. Nor have I suggested equalization to make a recording compensate for loudspeakers or to cater to your personal tastes in sound coloration.

But considering all of the flagrant scams I have suggested to "restore" the audio, you wonder what the truth is in all of this. The best I have been able to do, and what I suggest that you do also, is to compare the sound of your CD copy with your LP, and decide how you wish to edit and equalize your sound files to get the best match. You must make your own truth.

FULFILLING YOUR BURNING DESIRES

It may take one or two LP transfers to CD before you develop your skills as a remastering engineer to the point that you desire. But eventually you will possess a fully edited file that sounds promising via the playback monitor, and that you greatly anticipate burning onto a CD-R. This is not the time to rush.

My modest but decent-sounding Yamaha CD player, a horse-trade from my friend and erstwhile coauthor Chuck Crawford, is not that great at playing back audio files burned at 4× writing speed onto a CD-R blank. Evidently its playback laser and optical pickup require the superior eye pattern provided by the 1× writing speed in my CD-R drive. I can record at 4×, but then the Yamaha unit can play the music but it will not skip tracks properly using the remote. So, as in the days of half-speed mastered LPs, I force the burn to occur at 1× speed for my own CDs. You may need to experiment with your own consumer CD player; computer CD-ROM drives seem to have no trouble at all playing 4× mastered CD-Rs.

SOFTWARE TOOLS

Good software for two-track sound editing is Bias Peak from Berkeley Integrated Audio Software. You can download a limited-time free-trial version of it from http://www.bias-inc.com/. It is mildly expensive, but it will save you much time compared to other software. Bias Peak has a click detector that will find the ticks and give you the choice of repairing them yourself or of letting the software do it for you. However, an equalizer function is an extra-cost plug-in with this program.

If you don't mind the intimidating user interface and cumbersome workability of Pro Tools Free, you can download it from http://www.digidesign. com/. It is really designed for multitrack use, and it does not lend itself to efficient two-track operation in the restoration of LPs, but it does include an equalizer.

SoundEdit 16 is "in maintenance mode" according to Macromedia, but it is still available on special order. If you buy this, you need Pro Tools Free to round out your arsenal of tools. You can order SoundEdit 16 from a software purveyor, or from http://www.macromedia.com/.

This isn't a complete list of software tools for the Macintosh; you can do some sound editing in Adobe Premiere®, Apple's QuickTime Pro®, Apple's Final Cut Pro®, Strata VideoShop®, and in other video editing programs. But the ones I've mentioned previously are practical for LP sound restoration, and I have some experience with them. You can check out http://www.tracertek.com for more sound-editing software.

CADENZA

I discovered that it is possible to make CD-R versions of LPs that sound so much like the LPs that I am happy with them. Getting rid of ticks, static discharge, and mistracking distortion can truly breathe new life into a favorite recording. And I am talking about Baroque music that contains the delicious blending of tonalities and overtones of the different instruments.

Some music, such as Baroque strings and compressed rock, is not laden with abrupt dynamics nor difficult to transfer. However, a very fine Baroque harp recording by Marisa Robles on Argo ZRG 930 is a severe test of my 8600's digitizer, especially with a freshly optimized tonearm. The plucked strings really do sound live from the LP, but not so live from the CD version.

One LP I transferred to CD is L'Oiseau-Lyre 414-339-1, containing the Clarinet Concerto in A major and the Oboe Concerto in C major by Wolfgang Amadeus Mozart. Christopher Hogwood conducts The Academy of Ancient Music. It was originally a digitial recording, and it captures a particularly rich sound. To check on the quality of my LP transfer, I purchased a new copy of it on CD, L'Oiseau-Lyre 414-339-2, and played both of them on the same CD player.

As expected, the sound from the manufacturer's CD was slightly more harmonically rich and subtle than the transfer from LP. There was some frequency balance difference due to the small amount of EQ that I applied to the LP transfer. But the difference was slight when heard through a Yamaha CDX-480 player into a modified Hafler DH-200 (no electrolytic capacitors in its audio chain) through Teflon-insulated cables. I expect that a higher-quality sound digitizer for computers may sound better, although most of commercial grade use electrolytic input capacitors. Your LP transfers to CD will fall on the analytical side of the quality continuum, not on the lush romantic side.

Since I live in Southern California, where the electric power grid is in big trouble, it was necessary to listen late at night and on weekends to make sure that what I heard was coming from the LP transfer, and not from our noisy grid.

If you have music with sledge-hammer bass and powerful cymbal crashes, or a full symphony orchestra, then you might find that the analog-to-digital transfer is too much for your computer's converter; but you might give it a try to see whether the result is something that you can live with. In addition, my EQ curves work for me, but they should be only a suggestion for your own experimentation, which may yield a different curve for your own best results. ❖

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No-Feedback Voltage Regulators

This author shares the results of his extensive work on regulator circuits to control output voltage. **By Andy Nehan**

When most people think about regulators, they usually have in mind three-legged devices such as the LM, 78, or 79 series. These have a few features in common. First, they work at lowish voltages, usually less than 40V on the input; second, they have three legs; and third, they all employ feedback. Another group of regulators, sometimes described as though they used no feedback, are based on the shunt element.

Both these types of regulators, diagrammed in *Figs. 1* and *2* respectively, share certain features brought about by the use of an error amplifier. All amplifiers have finite gain and limited bandwidth. In an ideal world, you would use an error amplifier with a constant gain and phase shift (group delay) from DC or low frequencies through the audio spectrum and beyond.

The reasoning behind this is that the output impedance of a feedback regulator is set by the characteristics of the pass element and the error amplifier such that:

- the higher the error-amplifier gain, the lower the output impedance; and
- the turnover frequency—after which the output impedance usually rises monotonically—is set by the turnover frequency of the error amplifier. In practice, this can be in the region of 100Hz–10kHz.

Figure 3 shows a typical output impedance of an LM317-type device. My aim was to design a circuit with fundamentally different parameters, namely, a stable output impedance across the whole audio band with room to spare, and high attenuation of input noise and general grunge on the rails. Looking at these items in more detail, I'll start with the rectifier and work towards the regulated output voltage.

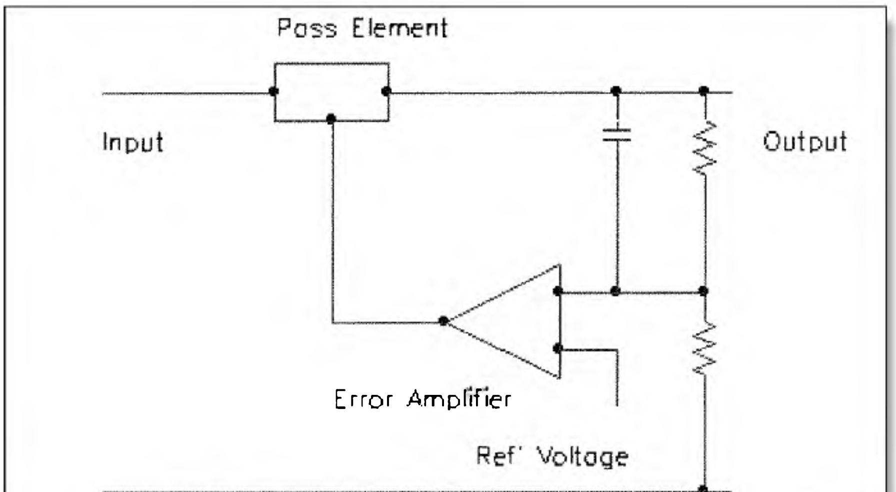


FIGURE 1: Simplified diagram of voltage regulator. G-1480-1

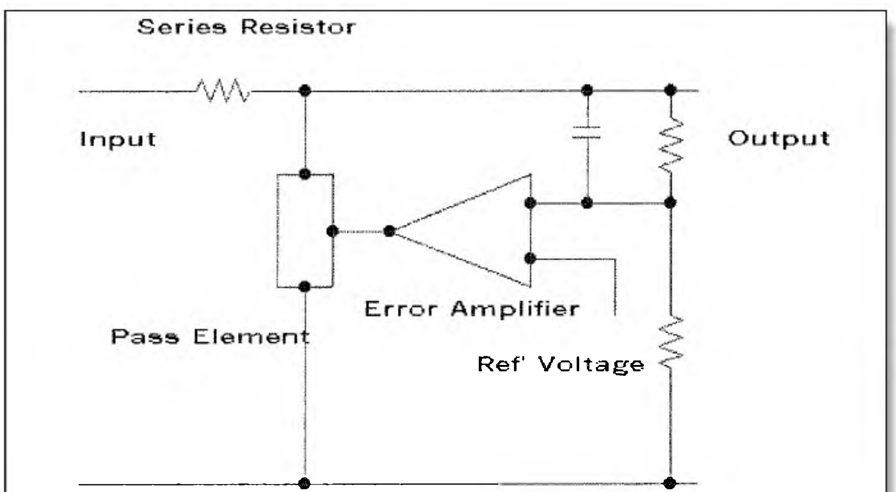


FIGURE 2: Simplified diagram of shunt regulator. G-1480-2

GENERATING THE DC

Not only must you consider the effects of power-line harmonics brought about by rectification, but also the power-line input signal itself, which is typically far from a pure sine wave. The effects of transformer rectification, as well as many switch-mode power supplies, in-

ABOUT THE AUTHOR

Andy Nehan was trained as a chemist, but then discovered computing. He has spent his whole working life in IT and is currently working as a project manager for an IT reseller. Nehan had an early interest in Ham Radio and has been building audio electronics for the past 25 years.

troduce a decidedly flat-topped sine wave, so much so that I understand the electric-power industry is introducing new regulations requiring manufacturers to fit rectification circuits that draw power from the whole cycle rather than near the peak.

Figure 4 shows the measured results I obtained using an EI frame-type transformer. I was surprised to see signals only 60dB down at 850Hz and 240mV in amplitude, quite a reasonable size. I made these spectrum analyses using the PICO[1] instrument. I further examined the line voltage up to 10MHz, which is the upper frequency limit of my equipment.

I expected to see the effects of TV line-frequency components, which for the PAL system used in the UK would be at 15,625Hz. In fact, I observed none; the trace was clean down to better than −60dB, the limit of resolution of the equipment. The process of running FFTs (fast Fourier transforms) with averaging does impact the final result because, in addition to the limitations the FFT itself already imposes on the capture of transient events, these events are eliminated in the averaging process. So perhaps the signals were there, but were too transient to capture.

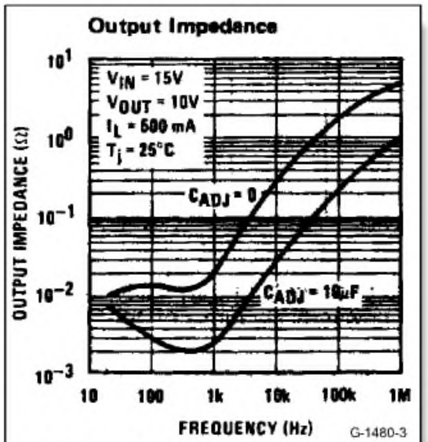


FIGURE 3: LM317-type device output-impedance plot.

I use frame-type transformers because their sound is more to my liking. They are not as fast as toroids, but I find they have better detail and a more natural presentation. Now, looking at the output from the rectifier stage to see what's really there, *Fig. 5* shows the output using a standard bridge circuit. Nasty—even worse than the power-line supply voltage. Now there are signals out at 2kHz that are less than 60dB attenuated relative to the 50Hz ripple voltage.

NEED FOR CLEAN INPUT

I wished to remove as much of the harmonic signal content as possible and, at the same time, remove any transient RF that might be present. The reasoning behind this was that all regulators have capacitance between the input and output terminals, which reduces the effective isolation between these two ports. This is in addition to any shortcomings in the feedback or reference network. Therefore, it follows that, for the output of a regulator to be clean, the input must also be clean! This presents a bit of a quandry; how do you get a clean input?

RC or LC filtering can make a significant reduction in the harmonic content of the waveform entering the regulator, the noise floor, and the RF content. However, possibly the most important question is one of definition—just when is a clean signal really clean? After all, the designs in glass audio articles in *audioXpress* are aimed mainly at tube circuits that, on the whole, are single-ended and have poor PSRRs.

In order to bottom out this area, I experimented by fitting both single- and double-pole RC filters, after the reservoir capacitor, in the rails to the voltage-regulator stage. As expected, adding the first pole made by far the greatest audi-

TABLE 1
COMPARISON OF ORIGINAL AND ATTENUATED AMPLITUDES

FREQUENCY	ORIGINAL, FROM RECTIFIER	AFTER TWO STAGES OF FILTERING	SIGNAL AMPLITUDE AFTER TWO STAGES OF FILTERING
100Hz	+6dB	−23dB	>100mV
200Hz	+2dB	−38dB	~10mV
300Hz	−6dB	−52dB	<3mV
400Hz	−10dB	−62dB	<1mV

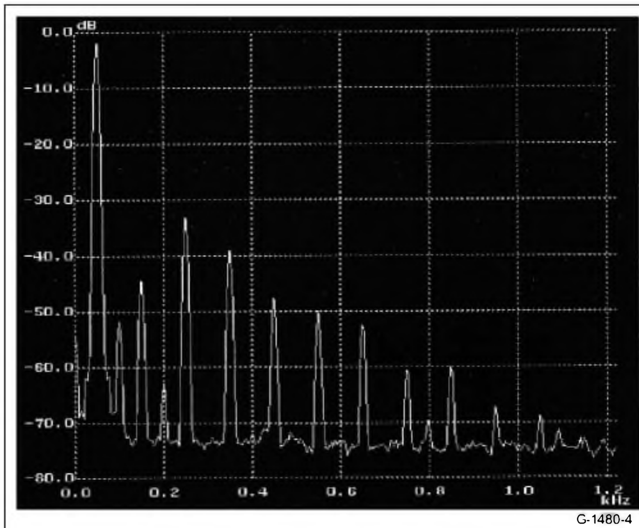


FIGURE 4: Frame-transformer AC spectrum analysis of line voltage, where 0dB equals approximately 20V RMS.

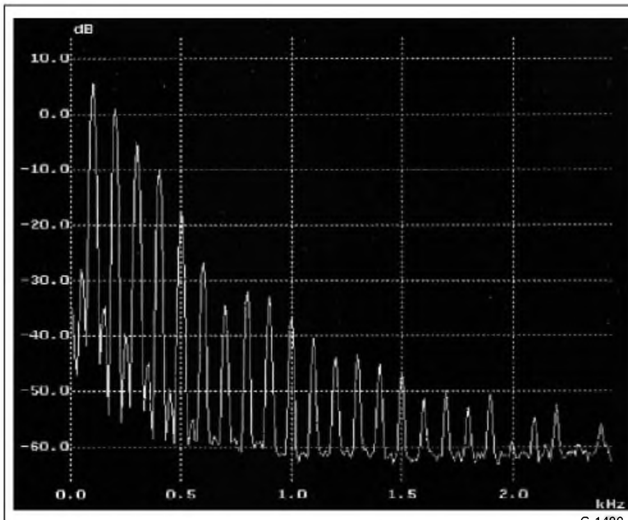


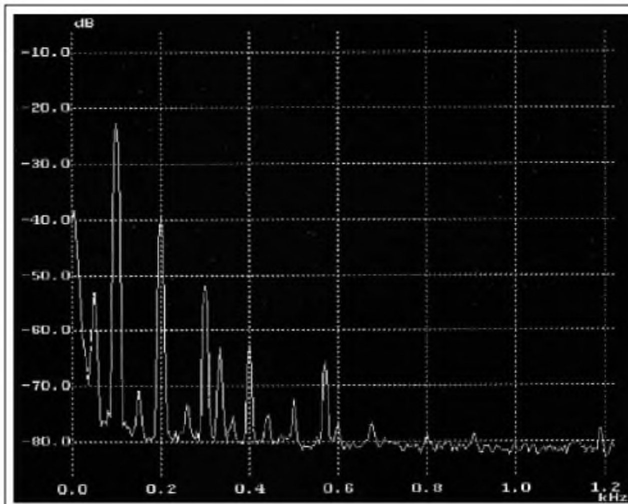
FIGURE 5: Output of rectifier, 2V pk-pk ripple.

ble difference. The second pole still makes a worthwhile contribution. However, I limited the design to two poles of filtering to save space and expense.

Figure 6 shows the output of the circuit of *Fig. 7*, and the data is presented in *Table 1*. What a difference! Not only have the upper harmonics been savagely reduced, but even the fundamental is heavily attenuated, just as I wished. My instrumentation does not allow me to accurately measure the background noise in the presence of signals, so I cannot comment on this. Apart from the fundamental, all signals are below approximately 10mV.

Another pole of filtering might reduce these amplitudes by some 20dB at 200Hz, and more at higher frequencies, but to be honest, they should then be rapidly approaching the noise floor. Simplified modeling of a power FET-based regulator (*Fig. 8*) showed 100dB rejection of low-frequency input signals and 40dB at 100kHz. These impressive figures are fine, but you're unlikely to achieve these in practice because of pick-up from wires in the vicinity of the PCB and the circulating ground current on the PCB itself.

So I decided to err on the side of caution and allow 60dB at low frequencies and 20dB at high frequencies. The implication of this is that a 100mV 50Hz signal becomes 0.1mV. Reduction of high-frequency signals is not now so important, since they have been attenuated by the RC filters.



G-1480-6

FIGURE 6: Spectrum analysis at the output of the two-pole filter network, 100 mV pk-pk ripple.

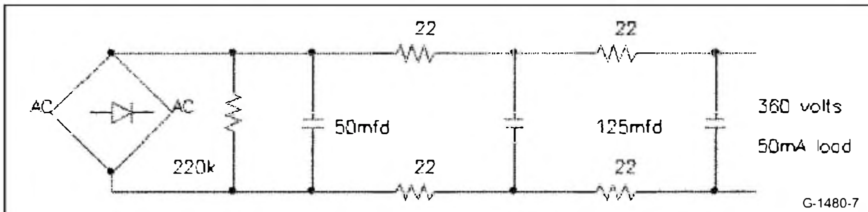


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FIGURE 7: Circuit diagram of rectifier network, all caps bypassed by smaller-value FKP1s.

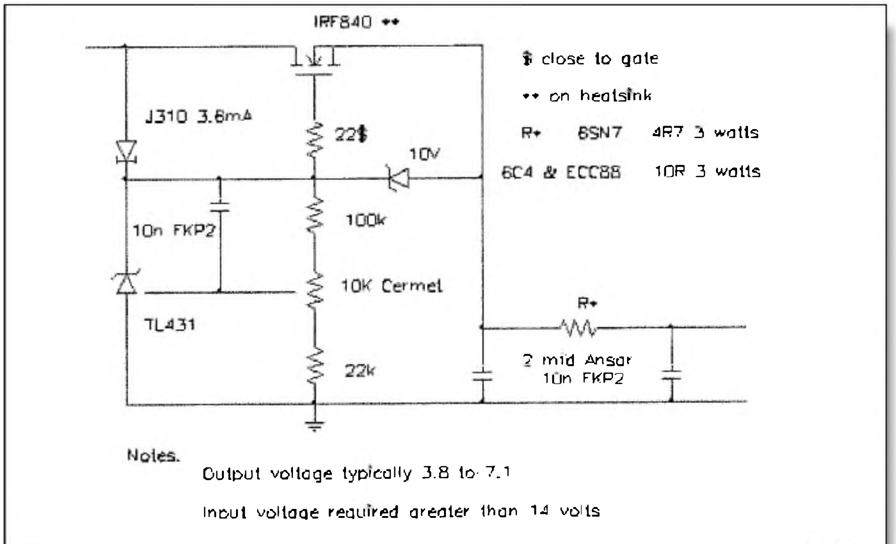

FIGURE 8: Heater/low-voltage regulator.

SMALL SIGNALS

Based on my experience, I believe that all non-fundamental AC signals should be as small as can be reasonably achieved, and as the frequency increases, their amplitudes should be further reduced. This is because the ability of the regulator to block them is itself reduced by the decreasing capacitive reactance between the input and output ports, and possibly by the impact of the gain bandwidth product of the error amplifier. As you can see in *Fig. 6* and *Table 1*, this has largely been achieved.

Adding a resistor/inductor in both the ground and live leads is important. It's all too easy to fall into the trap of doing a quick canonical rearrangement to show that the result is the same as having a single resistor in one lead. Unfortunately, it is not. A distributed capacitance exists between the primary of the transformer and the secondary, in addition to a distributed inductance. This is likely to result in one end of a winding having more signal fed through from the primary than the other end. I am sure that it is possible to measure these effects, but not with my current instrumentation.

Introducing an LC or RC network will significantly reduce the harmonic content going into the regulator, and, just as important, wideband noise will at least be bandlimited. I have never been happy with the sound of LC filters over the long term, since, to my ears, they slow down the sound of the system.

This is not to say that a properly designed choke input filter would sound badly, but I have not tried these during the last five years or so, since they don't seem to match well with my designs and systems. There are limitations on the size of the series resistors; if the values are too large, then, once again, the system slows down. I have settled on 22Ω per leg for low current loads, with larger current drains requiring smaller resistor values.

NEARING PERFECTION

Unfortunately, I have not quite reached perfection. Not only are there harmonics of the input line frequency, but when rectified, DC level shifts are brought about because the utility companies in the UK are required to maintain the AC voltage within the wide tolerance of −10% and +6%, and that's a lot of volts! I have not presented a graph or table of these, since they are very dependent on what's on TV during the evening (at the break in a football match, there is a nationwide mass turning-on of kettles and a resultant change in line voltage).

You can easily see this by connecting a scope to the rectified output, with the scope input set to AC. Then you can spot consistent level shifts on a regular basis (from less than a second to a few seconds). Short of regenerating the line voltage, there is little you can do about these, so it's time to grin and bear it. I do suspect that these level shifts may add

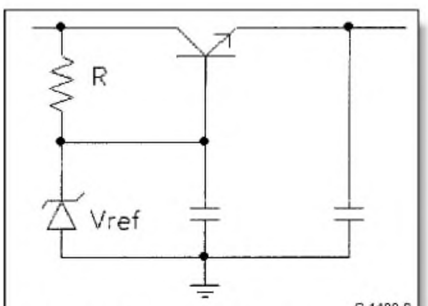

FIGURE 9: The basic configuration.

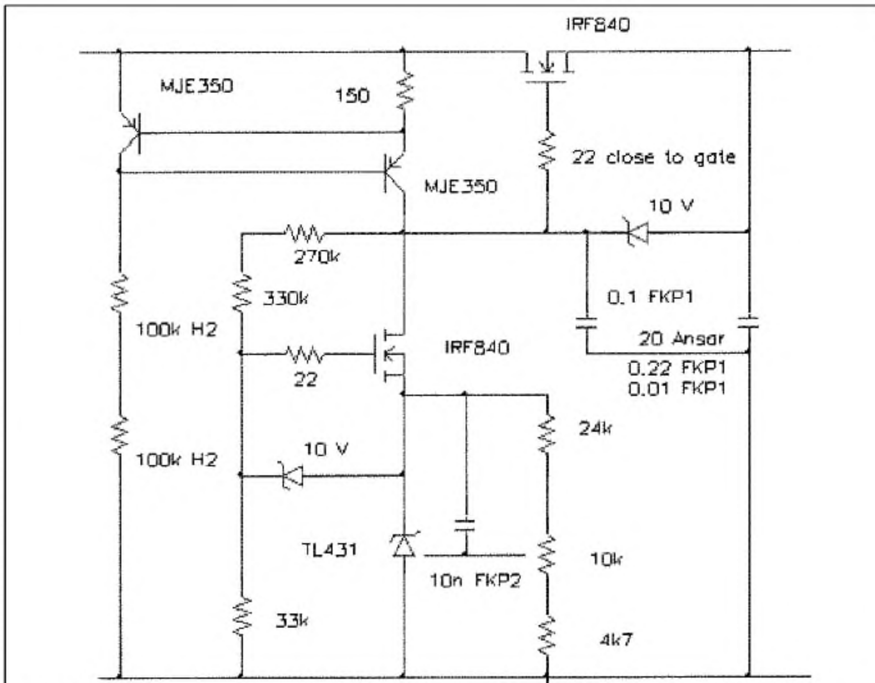

FIGURE 10: Circuit of HT regulator.

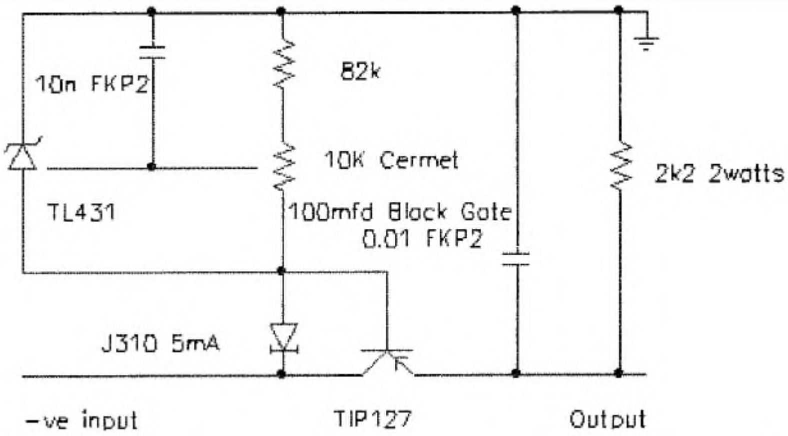

FIGURE 11: Negative regulator. G-1480-11

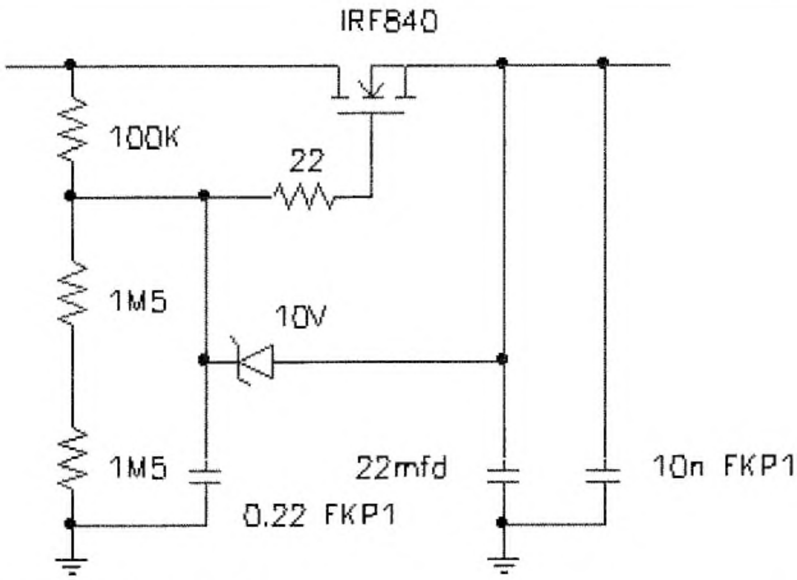

FIGURE 12: The buffer circuit. G-1480-10

to the low-frequency background noise in the system, but I cannot, at present, confirm this.

THE WAY FORWARD

I have worked on a zero feedback design that I believe fulfills my requirements and which, after listening tests, has replaced previous feedback designs of high repute. A design with a high degree of feedback will always be playing catch-up with the amplifier's varying current draw.

My method is to use a stable voltage reference, followed by a simple buffer feeding a storage device (a capacitor). The reference provides the attenuation, the buffer provides constant output impedance, and the capacitor provides an instant supply of current for the amplifier. Both designs were modeled to confirm design goals. High-feedback designs offer greater attenuation and have lower (but rising) output impedance, yet these designs seem to perform better when judged by listening tests.

THE BASIC CONFIGURATION

The main regulator must supply a stable output voltage and provide ripple rejection. The design is based upon the simplest of regulators, but with each component replaced by what I consider to be the best design for purpose (*Fig. 9*).

Maximum attenuation of input noise will occur if R is large and Z_{Vref} is small. Voltage references work best when fed from a high-resistance source. A constant-current source should therefore fit the bill quite well. For high-voltage regulators, I used a ring of two, since it is both simple and provides a better performance with a fluctuating input than the simple transistor type. Low-voltage regulators used a ring of two or a constant-current diode.

I set the current at about 5mA for high-voltage regulators (since both the reference voltage and its divider network draw current) and a little lower for the lower-voltage regulators. The TL431 itself needs 2mA to work properly, plus whatever the rest of the circuit requires. One small point to keep in mind is that the ring of two, like some other constant-current circuits, can oscillate if high F_T devices are used. For this reason, I have used the MJ340/350 which, in my experience, is fine.

Zeners are inherently noisy and have poor temperature coefficients. Their output voltage will vary with changes of airflow across an open chassis. In addition, their impedance characteristic is poor. I used a TL431 adjustable-voltage reference as the building block, since it offers low impedance and its output voltage can readily be set using a simple divider network. In the heater circuit, it is used as is.

THE LOW-VOLTAGE REGULATOR

The pass element could be either a bipolar transistor or a FET. In practice I used FETs, selecting those that had a high g_m, high wattage, and high voltage. Reliability has been superb. A heatsink is required for both high- and low-voltage regulators. The capacitor across the upper-arm components surrounding the TL431 is included, since it reduces the circuit noise. *Figure 8* shows the low-voltage regulator.

The output network is quite important. It works well even when added to an LM317, where it effectively negates the rising output impedance, but at the same time increases it by a few orders of magnitude. Its effect is to reduce the noise on the rail and to separate the heater from the regulator. Unfortunately, this means you must also be prepared to trim the voltage when changing tubes, because no two tubes ever seem to have exactly the same heater-current needs. I believe this is a small price to pay.

If I was building this again, I would probably split the output resistor in two, putting one half in the negative rail, since I suspect this would be audible and worthwhile. Listening showed that only a 2µF output capacitor was required, especially when the output network was added. I have not used the same network for high voltage regulators because the varying current requirements of a typical single-ended stage do not lend themselves to such an increase in output impedance.

The output impedance of the circuit in *Fig. 8* is flat almost from DC, and has approximately the same value as the series resistance. It then drops when the µF capacitor kicks in at around 10kHz. Without the output network, the output impedance is set by the g_m of the pass device, and would be in the region of 0.5Ω—once again flat across the audio spectrum until the capacitor operates.

THE HIGH-VOLTAGE REGULATOR

Figure 10 shows the circuit of the HT regulator. A TL431, with a maximum output voltage of 30V, drives the FET V_{be} multiplier, itself an IRF840, whose amplification factor is set by the ratio of the combined 270k and 330k resistors to the 33k lower arm, i.e., 15 times. This whole network is driven from a ring of two constant current comprising the MJE350s and the 150R resistor, which sets the current at about 5mA.

The pass element, an IRF840, then regulates the voltage. It's very similar to the low-voltage regulator, except that it uses no output network, and the output capacitor is quite large electrically and physically.

Note that capacitor quality is critical. Each needs to be fast and have low impedance; WIMA FKP1 film and foil types fit the bill. Ansars are OK, and other capacitors may well work very well in the output position.

NEGATIVE-VOLTAGE REGULATOR

The newest regulator is that for negative voltages (*Fig. 11*). Inevitably there must be some changes from the positive type, if only because components such as the TL431 do not come in a negative version. In fact, I still used the TL431, but made the pass element a Darlington transistor. Ultimate performance was not required from this circuit, so my goals were a little more relaxed.

THE BUFFER

Now that I have described the heater and HT regulators, there are only the individual HT stage buffers to consider (*Fig. 12*). Their function is simply to provide a low-output impedance and isolation. Once again, feedback elimination was paramount. After all, there's no point spending all this time getting rid of its insidious effects only to let them in by the back door!

Fortunately, the buffer is very simple. All that's needed is a low-impedance drive to each of the stages, readily provided by a simple FET whose output impedance is roughly 1/gm (quite small). Once again, the output impedance is flat-ish across the audio band. Choice of output capacitor is important. I have found that paralleling capacitors of similar-value does not deliver the goods. A 20µF sounds better than two 10µFs, from the same manufacturer, in parallel.

BUILDING

The construction is fairly standard if you use both sides of the PCB to limit the overall space. It is critical that the PCB material is double-sided PTFE, one plane of which is grounded, since this makes quite a difference in the sound. Mine have been in operation now for over five years without causing trouble, so reliability should not be a problem.

LISTENING

If you are used to listening to LM317-type regulators, the sound of the NFB (no feedback) regulator may be a bit of a shock. The first thing that will surprise you is the lack of apparent speed. I believe that this is misleading, since the LM317 creates the sensation of speed by inaccurate representation of the sound. Close listening will reveal that the NFB circuit has all the grunge removed that the LM317s put in.

Then spend time getting used to the sound—don't decide after only an hour's listening. I am confident that you will be happy with the overall presentation. For me it's like trying raw fish for the first time—you simply must forget your prejudice.

Following are some notes I made when auditioning the LM317, tube, and NFB regulator.

The LM317 (as heater regulator) with output network is considerably better than the straight LM317. However, you must get the values right; 10Ω and 2μF seem about right.

The LM317 (as heater regulator) with output network versus the NFB regulator: the NFB is much faster, the top is considerably extended, it has greater dynamics, i.e., the loud bits are louder, the soft bits softer. It has loads more resolution.

The NFB (as HT regulator) versus an advanced tube regulator using four tubes, one pass element, one voltage tube as reference voltage (85A2), one voltage tube as anode load for error amplifier (85A2), and error amplifier (EF91). Very similar in overall presentation, tonally the same, and frequency presentation also very similar. However, the NFB regulator is quicker, and the tube slightly thick in sound.

In my experience, varying the HT voltage in my preamps by as little as ±5V is quite audible. Indeed, each stage needs to have its individual HT voltage set as to optimize its sound. I realize that this appears controversial, but since each of my amplification stages has its own HT regulator, I can vary each voltage quite readily, and the effect is immediate and obvious.

I conducted many listening sessions on capacitors, and eventually opted for WIMA FKP1s, which are film and foil, although Kimber is also OK, especially for the larger values, and particularly since FKP1s are available only to 0.22μF. Clear differences exist between standard electrolytic and polypropylene type caps, with the polypropylene much preferred.

In my systems, I can readily distinguish the sound of the PP caps used to bypass the electrolytic caps in the heater rectifier system, and the caps I like in the signal path are the same as those I like in the rectifier system! So be prepared to use good stuff—cutting corners is just not allowed. ❖

SOURCES
PICO products
www.picotech.com.

A New Crossover for Cambridge Soundworks Model Six

Maintain domestic tranquility and provide for some great sound with these inexpensive bookshelf speakers.

By Jim Forte

PHOTO 1: Cambridge Soundworks Model Six Loudspeakers.

Remodeling. The word alone is enough to make most homeowners cringe. But when a sound buff is remodeling his listening room (living room to the rest of the world, including his wife), there is added stress. How can you remodel the room and keep all the family members happy? This very question will be answered by the end of this project.

INTERIOR DECORATION

My wife and I spent considerable time planning the new living room. And the only plan that would work involved a large wall unit that would cover the whole wall where the entertainment center and speakers now reside. So I need to give up my JBL Control Monitors and lead-filled speaker stands and go with a bookshelf model.

I am not very handy with woodworking, so I planned to purchase cabinets and design my own system. However, while searching the web one night I found a review for Cambridge Soundworks Model Six Bookshelf Loudspeakers. The review stated that they were designed by Henry Kloss (of KLH and Advent fame) to provide good performance when placed on a bookshelf. The dynamics and imaging were said to be very good and that they were a great buy at \$200. Hifi.com now sells them brand new for \$160.

I immediately hit eBay and sure enough reconditioned units were being sold by Hifi.com for \$99.95 on a regular basis. These reconditioned units have been returned to Hifi.com under their 45-day return policy and carry the full ten-year warranty. I decided to give them a try for \$99.95; if I didn't like them I could always resell them on eBay.

They arrived about a week later, and I anxiously unpacked the boxes and hooked them up. The sound was awful! The review had said that they needed to be broken in, so I moved them into the basement for a break-in period. It took about a month before they were fully broken in, but they now sounded fairly good, with a good overall balance except that the top end seemed to me to have a little too much sizzle. Also, the soundstage seemed to be all in a single plane within about four feet of each speaker.

ABOUT THE AUTHOR

James W. Forte holds a bachelor's degree in electrical engineering and has been designing and building loudspeakers for over 20 years. He is Engineering Manager for an electrical manufacturer and also runs Forte Acoustics Inc. (www.forteacoustics.com), which offers loudspeaker design and testing services along with custom-manufactured loudspeakers. Mr. Forte lives on Long Island with his wife Carol and their 2-year-old son Charles.

SPEAKER DESCRIPTION

The speaker (*Photo 1*) is a two-way unit with an 8″ poly woofer and 1¾″ paper cone tweeter. It is a sealed box design with an attractive teak vinyl exterior and a black knit grille. The cabinet itself is ⅛″ thick MDF with a ¾″ thick MDF front baffle, which also has a stiffening brace. Overall the enclosure is very solid.

The crossover is shown in *Photo 2* and the schematic is shown in *Fig. 1*. The inductors are bar-core types and the capacitors are non-polarized electrolytics. I get the feeling that the bean counters had a say in the choice of components for the crossover, because I am sure that Mr. Kloss originally had specified air-core inductors and film capacitors.

The crossover frequency is quoted at 2kHz. It is a first-order low-pass, second-order high-pass with no impedance compensation. There is a contour circuit for the tweeter (a 12µF capacitor in parallel with a 7Ω resistor) that operates below 13kHz and a 2Ω resistor to

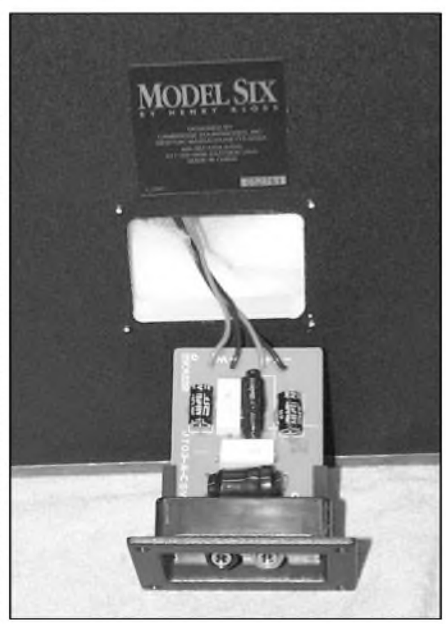

PHOTO 2: Original crossover for Model Six Loudspeaker.

match the output of the tweeter to the output of the woofer.

The crossover is definitely the weak link in the system, so I set out to design a new one. I needed to get a benchmark for the frequency response, so I measured from 200Hz–20kHz using my Cliolite measurement system (*Fig. 2*). The rising high end is readily apparent. (A word about Cliolite: Using a computerized measurement system really reduces the time it takes to measure your speakers. However, you can achieve the same results using good analog measurement techniques; it just takes longer.)

I then removed the woofer and measured the impedance (*Photo 3*). The woofer impedance curve (*Fig. 3*) has a typical shape, with a peak at resonance and a rise at higher frequencies due to voice-coil inductance. A filter cannot operate properly with this impedance curve, so I needed compensation.

I had a 10Ω, 10W resistor and some 5μF caps in my parts drawer, so I connected them across the woofer. I then ran the impedance curve again with Cliolite, which takes about 30 seconds. The combination of 10Ω and 10μF produced the flattest curve (*Fig. 4*). From the curve I calculated that the average impedance is 10Ω.

I repeated the process on the tweeter (*Fig. 5*). Note that there is no resonant peak, showing that this is a ferrofluid-cooled tweeter. There is an impedance rise at high frequencies that you can compensate for. After some experimentation, a 5Ω, 5W resistor in series with a 5μF capacitor produced the flattest curve (*Fig. 6*). The average impedance is 4.5Ω.

CROSSOVER DESIGN

The literature that comes with the speakers states that Henry Kloss spent many hours voicing the system. That, along with his 40+ years of experience, told me that I should not just throw his crossover design away, but use it as a tool to design the new one. I planned to keep the new crossover as a first-order low-pass and second-order high-pass design, as well as the 2Ω resistor in series with the tweeter to provide level matching. However, I intend to add impedance compensation for both the woofer and the tweeter, but not include the contour filter at this time.

I was concerned about power handling of the tweeter with a crossover of 2kHz so I bumped it up to 2.2kHz. I also aimed to use a Linkwitz-Riley configuration for the high-pass filter, as well as air-core inductors and film capacitors.

To determine component values I used 6.5Ω for the tweeter impedance and 10.5Ω for the woofer. I added 2Ω to the tweeter impedance to include the

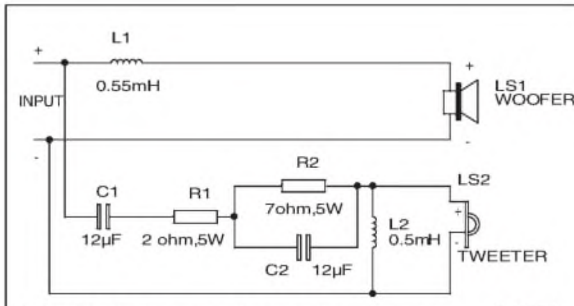

FIGURE 1: Schematic of original crossover.


PHOTO 3: Measuring impedance using ClioLite.

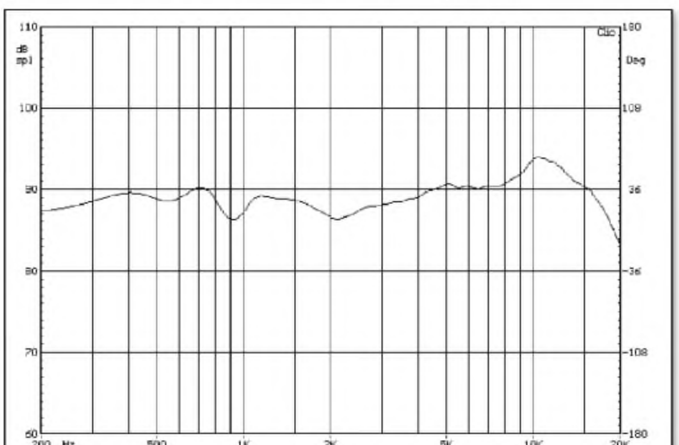
FIGURE 2: Frequency response—as received.

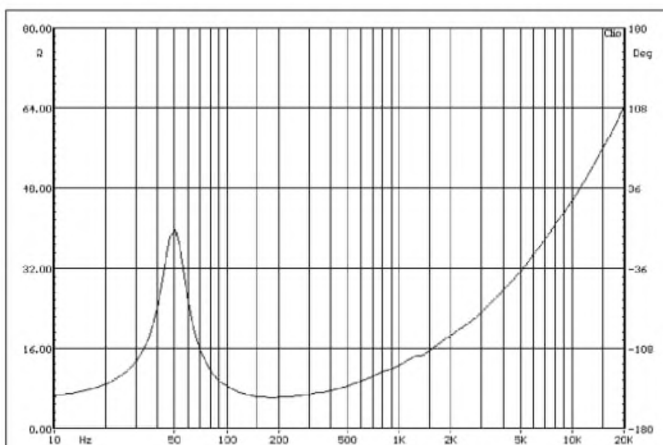
FIGURE 3: Woofer impedance—as received.

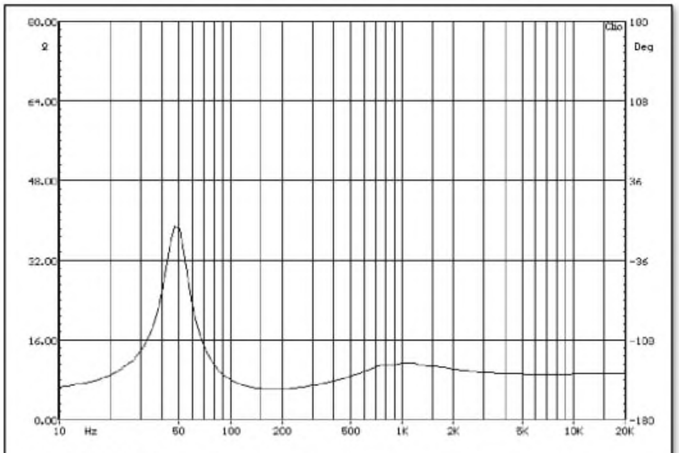

FIGURE 4: Woofer impedance—compensated.

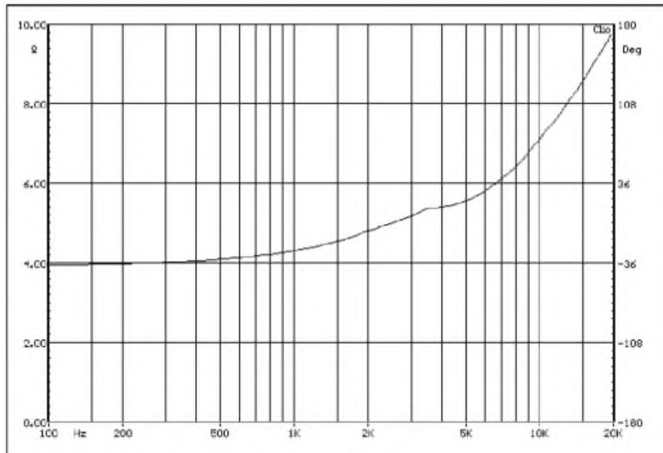

FIGURE 5: Tweeter impedance—as received.

leveling resistor and added 0.5Ω to account for the resistance of the inductor that is in series with the woofer. From the fifth edition of the *LDC* (*Loudspeaker Design Cookbook*), L2 for the low-pass circuit is 0.76mH; C1 for the high-pass circuit is 5.6µF; and L1 for the high-pass circuit is 0.94mH. I used 0.82mH (DCR = 0.47Ω) for L2, 1.0mH for L1, and 5.0µF for C1, as I had these parts in my bin.

The new crossover schematic is shown in *Fig. 7*. I estimate that it will cost about $35 to purchase all the crossover parts. *Table 1* is a list of crossover parts from Parts Express (www.partsexpress.com).

TABLE 1
CROSSOVER PARTS

ITEM DESCRIPTION	QTY NEEDED	PARTS EXPRESS CATALOG NO.
0.8mH (18 ga) air-core inductor	2	266-822
1.0mH (18 ga) air-core inductor	2	266-826
4.7µF film capacitor	4	027-422
10.0µF film capacitor	2	027-428
10Ω, 10W resistor	2	016-10
5Ω, 5W resistor	2	015-5
2Ω, 5W resistor	2	015-2

INSTALLATION

Building and installing the crossovers required a slight modification of the binding post plate (*Photo 4*). First, I removed the insulation from the cabinet and put it in a safe place for reuse later. Then I removed the binding post plate/crossover assembly from the cabinet.

I carefully removed the hot-melt glue from the nuts that hold the binding posts to the plate (*Photo 5*). Then I removed the nuts/washers and pulled out the binding posts. I saved the posts/nuts/washers for reuse. Then I removed the remaining glue and carefully separated the crossover printed circuit board from the plate.

Next, I carefully reinstalled the binding posts, being sure to line up the holes in the posts as shown in *Photo 6*. When finished, the binding post plate appears as shown in *Photo 7*.

I mounted the new crossover components on a plastic sheet, making sure that the board fit through the woofer opening. I mounted my components with wire ties rather than hot glue. The wire ties hold just as well and make rework/reuse of the components much easier. I reused the woofer/tweeter wires from the old crossover and added new input wires with ring terminals that will be installed onto the binding posts. The completed crossover is shown in *Photo 8*. Note that I mounted the inductors perpendicular to each other to minimize field effects.

PHOTO 4: Binding post plate—as received.

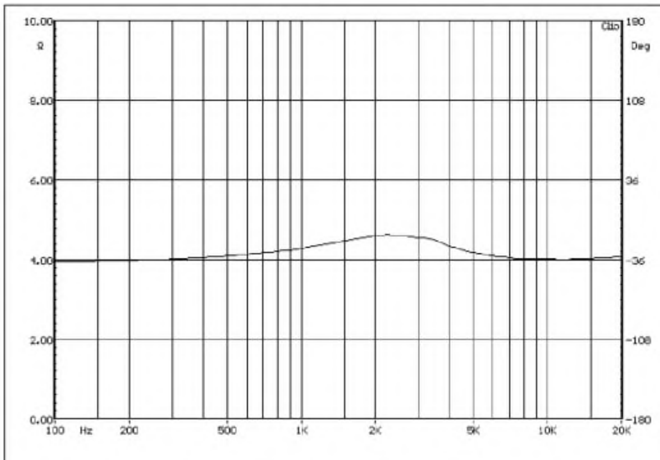

FIGURE 6: Tweeter impedance—compensated.

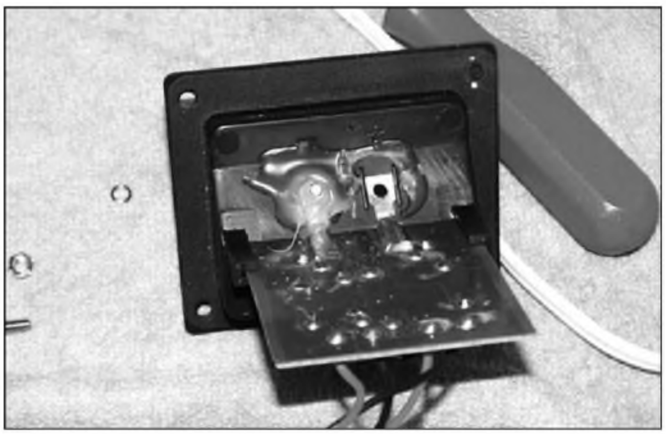

PHOTO 5: Hot glue removed from one binding post.

PHOTO 6: Proper alignment of binding posts.

I mounted the crossover to the back wall using three screws (½″ long). I located one screw at the top so that I could tighten it through the tweeter hole (*Photo 9*). I tightened the other two screws at the bottom through the woofer hole.

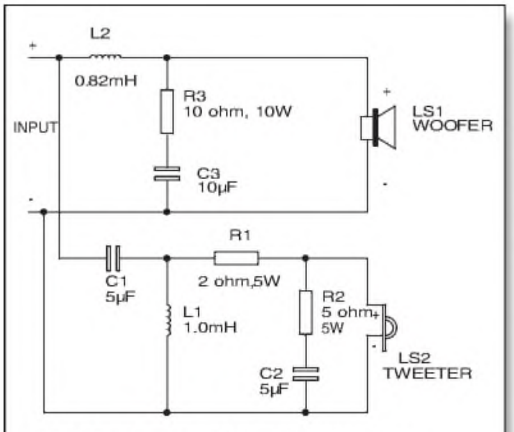

FIGURE 7: Schematic of new crossover.

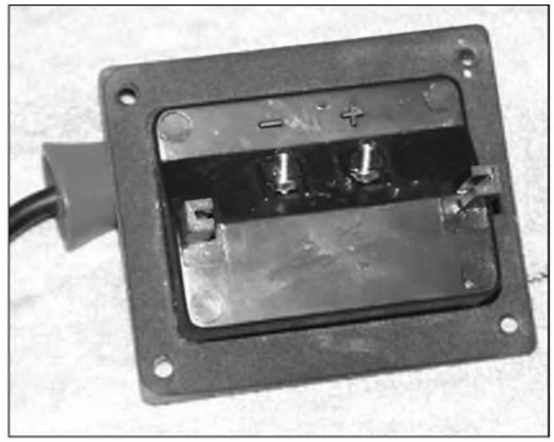

PHOTO 7: Fully modified binding post plate.

Make sure that the crossover board does not rattle at all. If it does, install some foam weatherstripping between the crossover board and the enclosure. The weatherstripping was not necessary for my application.

I turned the speaker over and pulled the input wires through the opening for the binding post plate. I attached the ring terminals (watch the polarity) to the binding posts using 4mm-0.7 nuts and lock washers (*Photo 10*). Finally, I installed the binding post plate to the enclosure.

I then turned the speaker face up again and re-installed the insulation. I ran the woofer and tweeter wires out of their respective holes and reinstalled the woofer and tweeter, making sure that I had the polarity correct. That's all there is to it.

TESTING THE UNIT

I measured the frequency response of the revised speaker (*Fig. 8*). The response has the same overall shape as the original but without the rising top end. I suspect that the impedance compensation for the woofer and tweeter is accomplishing the same task as the original contour circuit, but does a better job of it.

And how do they sound? I hooked them up, my hands shaking in anticipation. In a word, they sound superb. They are more dynamic than before; they don't sound as strained at high volume.

Listening to James Taylor (*Hourglass*, Columbia CK67912), I noted there is

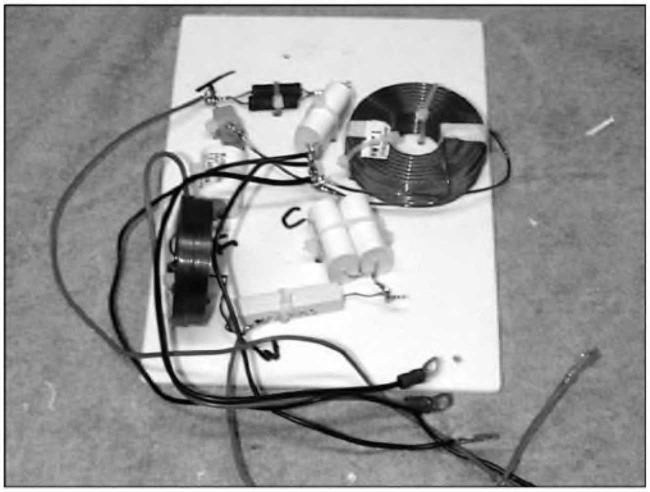

PHOTO 8: Completed crossover.

PHOTO 9: Crossover mounting screw visible through tweeter opening.

now space between the instruments and the vocals that wasn't there before. The sound is more balanced, more musical. I switched to "Return of the Jedi" by John Williams (RCA, RCD14748) and I was blown away. The soundstage is wide and deep; it no longer hovers around each speaker.

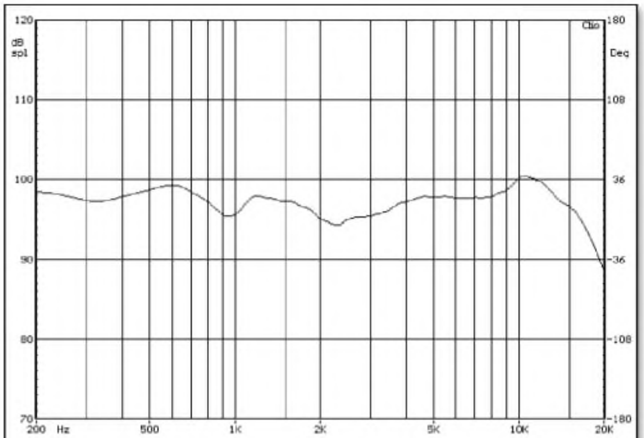

FIGURE 8: Frequency response with new crossover.

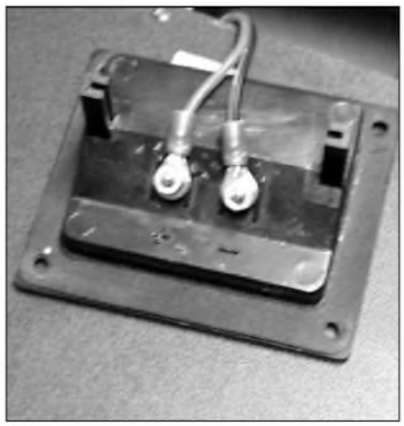

PHOTO 10: Crossover connected to binding posts.

Finally, I put on Joni Mitchell (*Blue*, Reprise 2038-2). Her song "River" came on and the speakers disappeared. It was just Joni Mitchell and her piano in my living room.

Are they the perfect speakers? No. They don't recreate a kick drum realistically, but they do have a nice bottom end. Certainly more than adequate for a bookshelf speaker. Incredible for a pair of speakers that cost me $135. I recommend this project to anyone who wants great sound in a small space on a limited budget. ❖

The Ultra Fidelity Computer Sound System, Part 2

Part one examined the computer listening environment and acoustics. Part two (of five) will use that knowledge to design the ultimate computer satellite speakers. **By R.K. Stonjek**

When you're listening up close—as you do at the computer—you'll pick up sounds that would be inaudible in the hi-fi environment. For comparison purposes, consider a listener at 4m from a single 94dB at 1W/1m hi-fi speaker, and a listener at 0.5m from a single 88dB speaker (roughly equivalent to a pair of 88dB hi-fi and 82dB computer speakers).

DESIGN CONSIDERATIONS

You could, for instance, hear noise from your hi-fi speaker when it should be silent and assess the level to be about the same as a leaf rustling at your feet—about 30dB. You can calculate the required level at 1m from the speaker to be 48dB, and so the amplifier output voltage must be 14mV for an 8Ω speaker. To hear the same level of noise at the computer, you need an output voltage of just 0.875mV.

This comparison gives you an idea of the extra problems at the closer listening distance. My Fluke Scopemeter (99B, series II) does not measure random noise below 10mV particularly well, making the debugging of this problem especially tricky.

Sound cards must be extra quiet. Add 18dB to make up for the difference in distance from listener to speaker and subtract 6dB to make up for the less efficient speakers, giving a total difference of 12dB. The computer sound system needs to have a noise floor 12dB lower than an equivalent hi-fi. Less efficient speakers are actually an advantage in the computer setup.

CABINET NOISE

Also of concern is the amount of noise coming directly off the speaker boxes. I have not found a consistent method of measuring this in relation to its impact on the overall sound heard by the listener. Experience indicates that the problem is more critical for the computer system.

I ran a simple experiment in an attempt to solve the problem of cabinet noise for close-in speakers. I played pink noise into a regular box-shaped cabinet and simply felt the vibration at various locations on the cabinet. The vibration increases as you move away from the corners. The point between corners forms a radiation point where you can hear sound directly from the cabinets. The walls of the box bow in and out to form a kind of speaker.

Based on this simple experiment, I reasoned that if the box had no corners—or at least very few—that it would have no accumulation points for direct radiation of sound from the cabinets. The speakers described here have few corners and seem to vibrate evenly all over. You do not notice any noise from the new cabinets, making the sound very clear and uncolored.

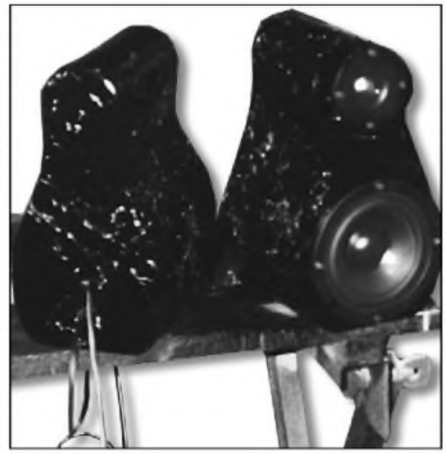

PHOTO 5: Completed satellite speakers.

I also considered Jim Moriyasu's work on midrange enclosures ("A Study of Midrange Enclosures, Parts 1 and 2," *SB* 7, 8/00) and the practical factors (such as size) of the environment in which the speakers were to operate. For computer speakers, you must think "information."

In early experimental versions I used locally badged polypropylene 4″ woofer-midrange drivers and matching Mylar dome tweeters. It worked well, but at the close-in computer environment you can "hear the cone." I have noticed this effect with cloth midrange and paper drivers when listening up close, but rarely when in the normal hi-fi listening position. This problem is unique to the close-in computer environment.

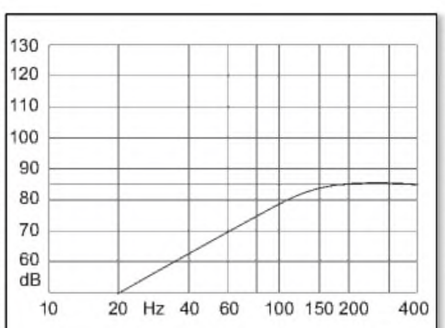

FIGURE 4: SPL at 1W/1m and driver sensitivity.

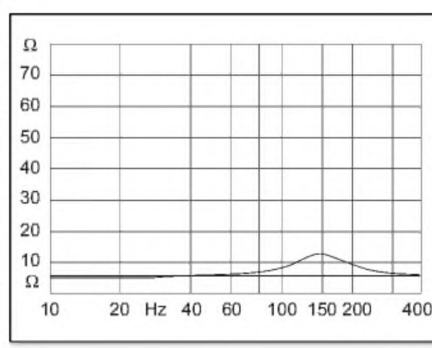

FIGURE 5: Impedance and DCR.

Changing to an Aerogel coned Audax woofer/midrange (AP100ZO) and a matching Titanium tweeter (TM020J7) fixed the problem. This also reduced the moving mass from 6.8 to 4.7g—comparable to cloth-dome

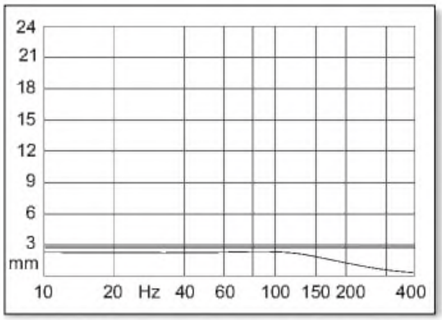

FIGURE 6: X_{MAX} **and driver excursion at 30W.**

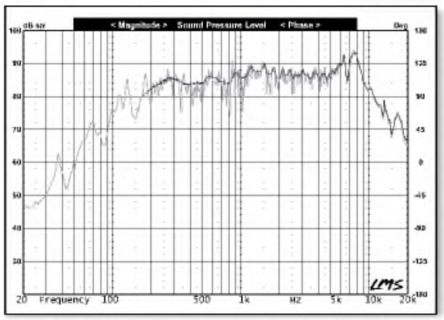

FIGURE 7: Bass at 0.5m, 1W. Bold line shows gated response.

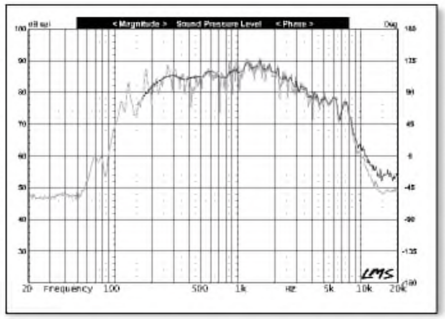

FIGURE 8: Bass and active crossover, 1W/0.5m.

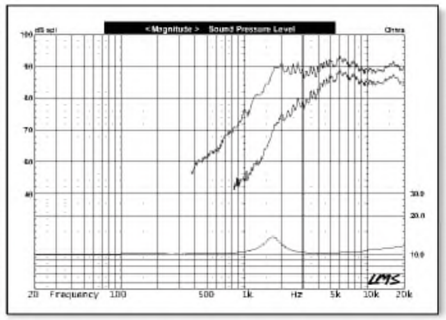

FIGURE 9: Gated tweeter response. Tweeter with active crossover and 4.7 series resistor. Tweeter impedance including 4.7 series resistor.

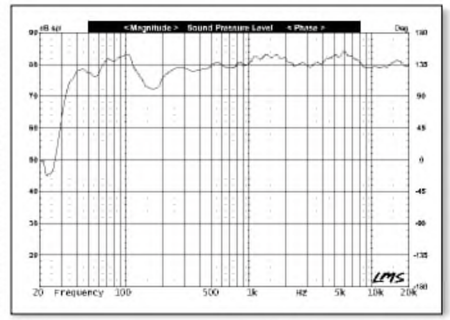

FIGURE 10: Full range, active crossovers.

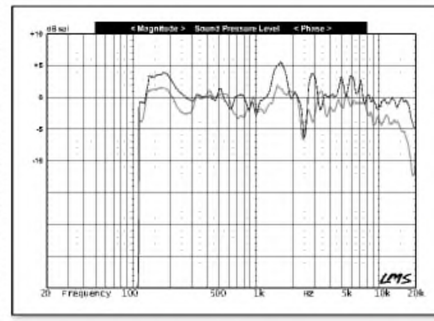

FIGURE 11: Left speaker measured from left and right listener's ear.

midrange drivers such as Dynaudio 76AF (4g) and Vifa D75 (3.3g). The lower mass helps to lower distortion and increase transparency.

PERFORMANCE

The cabinet design is made up of several layers of MDF that are laminated together (*Photo 5*). This technique is surprisingly straightforward, but first I want to discuss performance. These cabinets are designed to work with the active crossovers described in part 4.

Figures 4–6 are the design tool predictions for the 1.1 ltr cabinet; –3dB point is 135Hz. *Figure 7* shows the bass unit in the proposed box, while *Fig. 8* shows the same but with the active crossover.

I took measurements in open air with the speaker around 800mm off the ground and 500mm in front of a wall. This setup should simulate the average computer environment. The rear wall reacts with the bass at around 170Hz, causing cancellation. The cancellation frequency varies with the distance from the rear wall.

Figure 9 is the tweeter response with and without the active crossover. *Figure 10* shows the complete system including subbass and active crossover, 0.1 octave smoothing. *Figure 11* measures my own setup inside the house. My computer meets the wall at a 45° angle. Notice the dip at 170Hz has become a peak!

This graph also measures the difference in level as heard by the left and right ear to the left-hand speaker. With the mike only 80mm off center (i.e., between speakers ±80mm), you would not measure a consistent difference in level in the hi-fi environment. The graph shows a clear difference for the computer speakers, confirming the much greater effective channel separation.

Figure 12 graphs the satellite speakers with a passive crossover. *Figure 13* shows the passive crossover—note the tweeter is reverse phase.

CONSTRUCTION

Always wear a protective mask when working with MDF (medium density

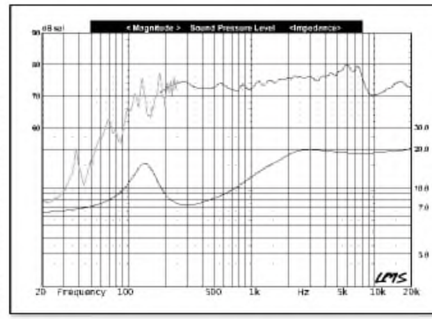

FIGURE 12: Satellite speakers with passive crossover.

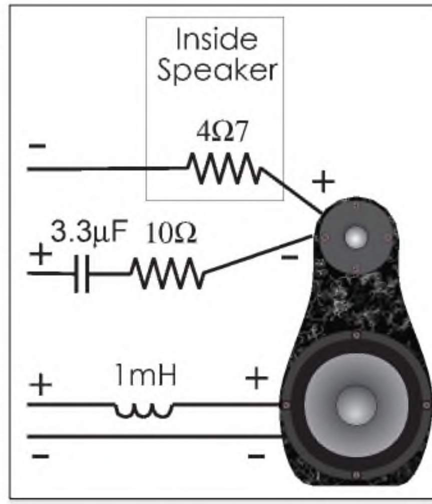

FIGURE 13: Passive crossover.

fiberboard). MDF dust can cause cancer and respiratory problems. The speakers are made by laminating eight pieces of MDF together. Using this technique, you can produce any shape that strikes your fancy. You can download the template for the design described here and for the baffleboard template from *audioXpress*' website at http://www.audioXpress.com/downloads/AX1001a1.zip.

Once you have done that, just print out the templates from your web browser or graphics program. Make sure they print at the correct scale—120 by 116mm. Cut out the printout so that you can trace around it.

Figure 14 shows the steps to construct these speakers. You must first make a wooden template. Use the downloaded template or copy the drawing from *Fig. 14* to a piece of MDF. Cut out the pattern using an electric jigsaw. Cut the pieces marked

“A” and screw and glue them together, then attach them to the piece marked “B.”

Now glue and screw the shaped piece to the rest of the template. Drill the four holes as shown. You should seal the edges of the template with MDF sealant or estapol varnish.

PHOTO 6: Template and pieces.

Using 18mm MDF, cut out fourteen 250 × 140mm blank pieces and two of the same from 9mm MDF. Place the template over one of the blanks. Through the holes in the template, screw the blank piece onto the template (from the template side). Clamp the template using the handle (A-A).

Using a router with a “flush trimming bit,” preferably a double-bearing type, trim the blank to the shape of the template. Four of the blanks will be left like this, while the other ten will need to be trimmed on the inside as well (*Fig. 14*/step 2).

Photo 6 shows the template and all of the pieces made using it. Using two of the blanks routed only on the outside, cut the holes as shown (you can use the baffle template to mark the holes). On the two 9mm blanks, cut the large hole only. Use a jigsaw to carefully cut the 9mm piece as shown in step 2 (*Fig. 15*).

Sand the front of the baffleboard (“F” and “E”) before assembly. You can round over the top of piece “E” (*Photo 7*). Check that the holes are the correct size by placing the speakers into them. In step 3, laminate the various pieces together using clamps as shown in *Photo 8*.

ASSEMBLY

Take care not to let the pieces skew during clamping. To help prevent skewing, start with the rear piece. Apply the glue to the first “C” piece and then press it firmly in place by hand and wait a couple of minutes. Continue up to the baffleboard in this way, then apply the clamps. No screws are needed.

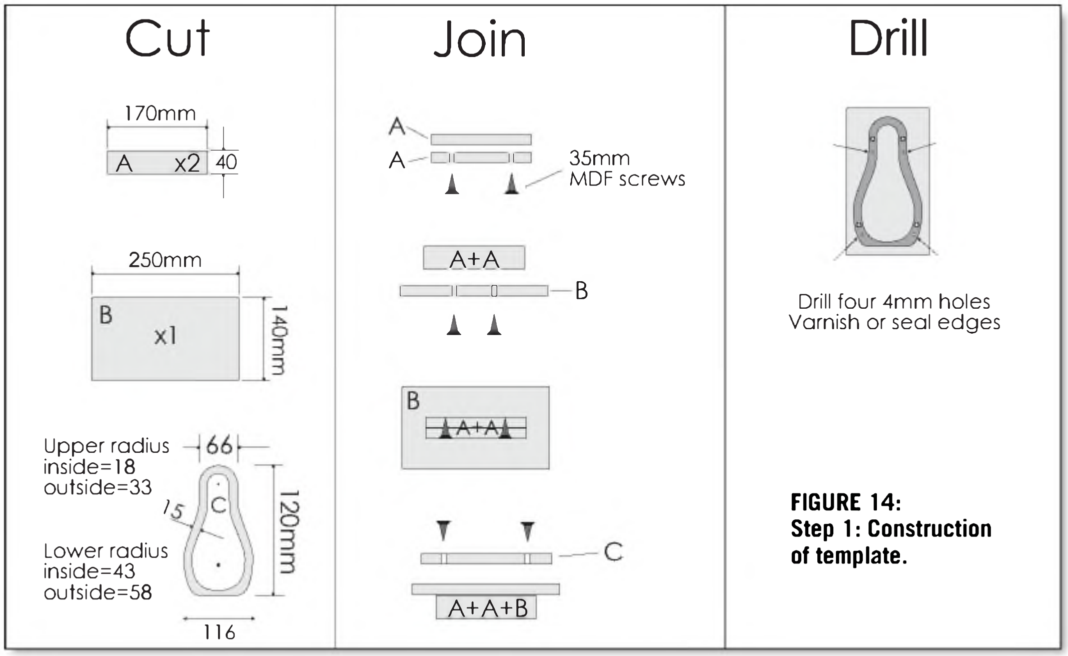

FIGURE 14: Step 1: Construction of template.

You should sand the sides once they are dry. Start by rasping any protrusions, then move to 60 grit sandpaper and work your way up to around 400g. Seal with MDF sealant if you have it; otherwise, you can use an appropriate undercoat. Drill two cable holes in the rear of the cabinet. The upper hole will be for the tweeter.

You can choose any finish that suits your fancy. I chose black with gold webbing. Always paint in a well-ventilated area and use an appropriate protective mask. First paint with matte black. Sand with fine sandpaper and recoat with matte black. The top coat should be gloss black (you get a very deep black this way). Spray the gold webbing on last (*Photo 9*).

When the paint is thoroughly dry, run the speaker cables in and seal them—I used hot melt glue. Make sure you have enough cable to run to the amplifier. If you're not sure, then you might consider speaker terminals at the back of the box—mark the tweeter and bass. Solder in the tweeter's 4.7Ω/5W resistor.

This resistor cuts down the damping factor between the tweeter and amplifier. With nothing between the

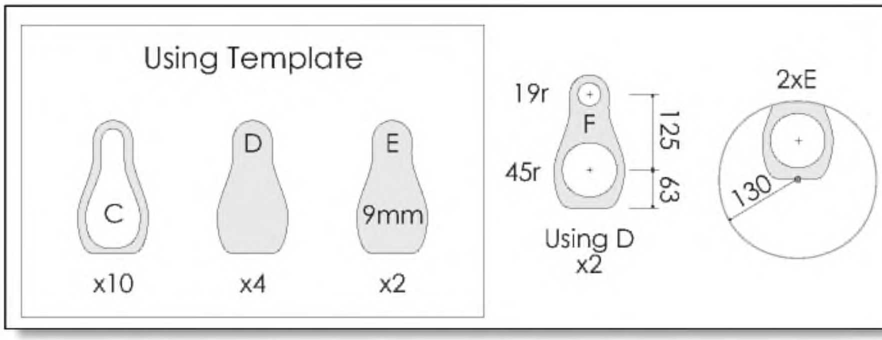

FIGURE 15: Step 2: Cutting list.

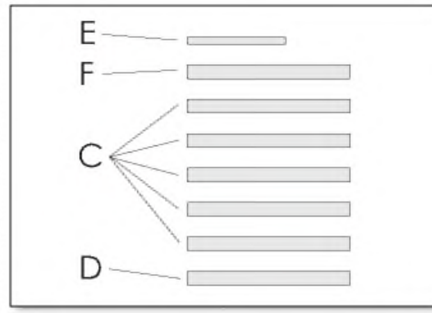

FIGURE 16: Step 3: Assembly. See *Photos 7–9.*

tweeter and amplifier, the amplifier treats the speaker as part of its own output devices and corrects for any overshoot. When a woofer overshoots, it produces a voltage. Because the amplifier's input does not have a matching voltage, the amplifier compensates for the overshoot. This makes bass and subbass very tight and controlled when using active crossovers.

The problem with the tweeter is that it does not really move in and out like a piston, but "flutters." There are still voltages produced by the tweeter and corrected for by the amplifier, but the correcting voltage is always heard as an extra tone, or "ringing," which can make some instrumental pieces sound sharper, but voices always sound gritty and sibilants sound horrible.

The series resistor effectively restricts the amplifier feedback to the amplifier, raises the damping factor, and protects the tweeter from the amplifier's switch-on voltage spike. If you're not happy with a wirewound resistor, use $3 \times 22\Omega$ and $2 \times 27\Omega$ 1W metal-film resistors (wired in parallel).

Fill the cabinet with acrylic* damping material. I always seal the driver using a black silicone sealant. To make later removal of the driver possible, rub Vaseline (petroleum jelly) di-

PHOTO 7: Ready for drivers.

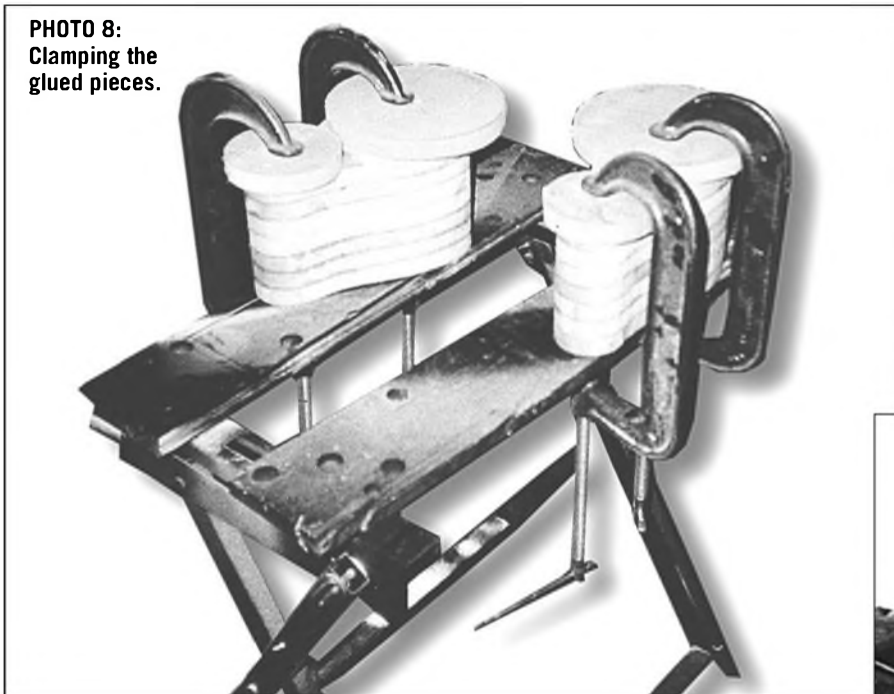

PHOTO 8: Clamping the glued pieces.

rectly onto the metal part of the speaker surround. I used decorative bronze screws. Pre-drill screw holes. If this all seems too much, then you could just make a regular square (golden ratio) box, around 2–3 ltr.

Figure 13 shows the passive crossover that will serve till the all-active system is finished. *Figure 12* is the response you should be getting from the passive setup. The passive crossover is designed to be simple and external for easy removal when the entire system is finished. Note that the tweeter is wired out of phase.

In part three I will detail the subwoofer construction. ❖

*See *aX* 8/01, "Box Stuffing—A Fistful of Fuzz."

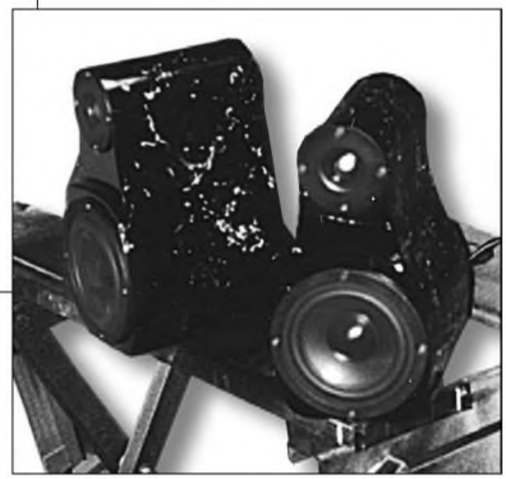

PHOTO 9: The finished product.

The AC Power Line and Audio Equipment

Part 2

This audio veteran examines the causes of interference and offers tips to ensure a clean AC power line. **By Charles Hansen**

Electromagnetic interference (EMI) is any undesirable electromagnetic emission or any electrical or electronic disturbance, man-made or natural, which causes an undesirable response, malfunction, or degradation in the performance of electrical equipment. EMI is broad-spectrum interference, either conducted on the power line, or radiated by wiring or circuits to susceptible equipment.

A utility alternator always produces some level of EMI. It is produced primarily by brush arcing in slip-ring alternators, or by rotating rectifier spikes in brushless alternators. The voltage regulator produces additional interference, which is normally a pulse-width modulation (PWM) switching regulator.

Radio-frequency interference (RFI) is any undesirable electromagnetic interference within the frequency spectrum assigned to radio transmission.

Conducted interference is noise injected by a device back into AC mains supply. It is distinguished from power-line harmonic distortion by its higher frequency range.

Radiated EMI comprises E-Field (electric) and H-Field (magnetic) components. Electric fields exist in space between two potentials, while magnetic fields exist around a current traveling through space or a conductor. You can easily control electric fields with

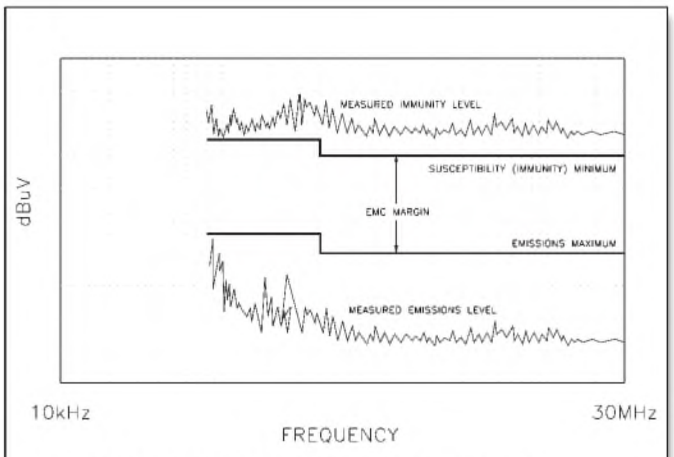

FIGURE 3: Principle of electromagnetic compatibility.

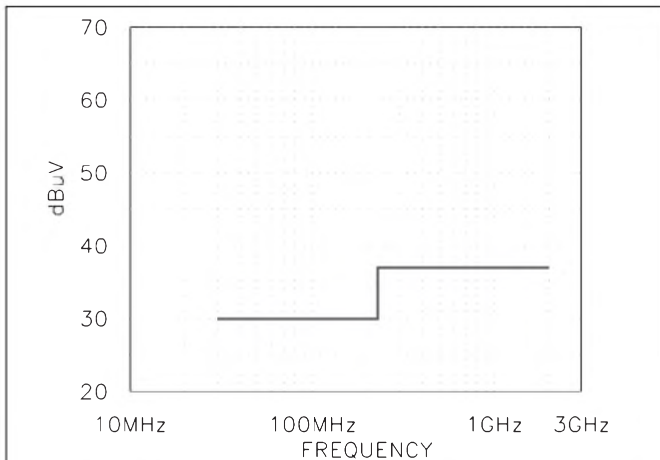

FIGURE 5: CISPR13 radiated EMI limits.

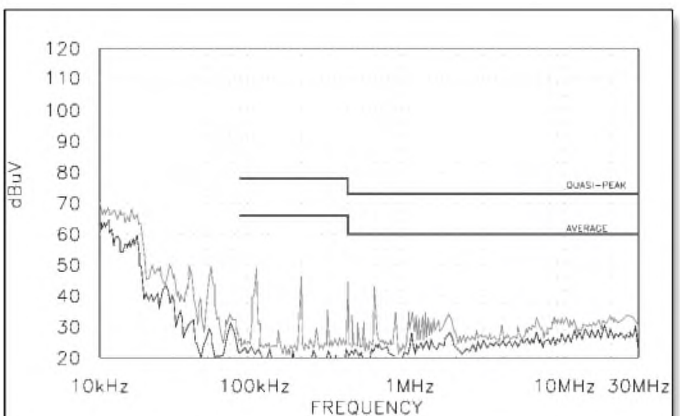

FIGURE 4: CISPR13 conducted EMI limits, with measurements.

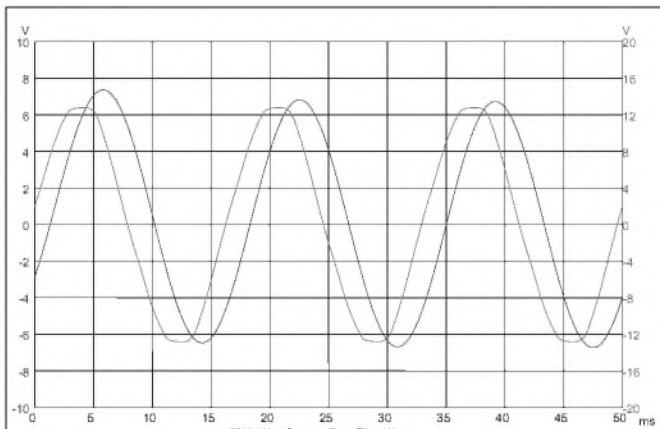

FIGURE 6: 60Hz AC line voltage waveform.

ground shielding. Magnetic fields can penetrate ordinary conductors and require high-permeability magnetic materials to contain them.

The most common EMI radiator is the AC power cord of an electronic device. The power cord or internal wiring may propagate any conducted interference generated by the device, especially if the conductor length is close to one-fourth the wavelength of the interference frequency.

EMI SOURCES

The rectifier diodes in power supplies produce spikes and ringing on the AC line as they abruptly turn off and on. The magnitude of the spikes depends on the diode recovery characteristics. Fast-recovery and fast/soft-recovery diodes are available to minimize RFI[4,5] produced by diode switching. Capacitor or RC snubbers across the diodes can further reduce RFI[6].

Switching power-supply semiconductors and their energy storage inductors and transformers can produce transients with lots of RF energy.

Light dimmers and motor speed controls operate by conducting for only a portion of the 60Hz AC cycle. When the SCR or Triac in the control is gated on, its fast turn-on pulse produces harmonics well into the RF spectrum.

RF signals are also intentionally placed on the AC line. X-10™ controllers place their digital codes on the AC line during the zero-crossing of each AC half-cycle (120 times per second for a 60Hz power line). Also, power-line-carrier coupling system RF signals are used to control remote unmanned power substation circuit breakers, providing a means for detecting faults on transmission lines. Power-line computer networking currently moves data along 120V wiring at 350kbps (kilobits per second), and promise speeds up to 11Mbps[7] (megabits per second). In the future, "smart" appliances may add their data to the power line.

INTERNATIONAL EMI LIMITS

EMI must be regulated in order to allow equipment to function properly without suffering degradation in performance due to interference generated by the equipment itself, or by other electronic

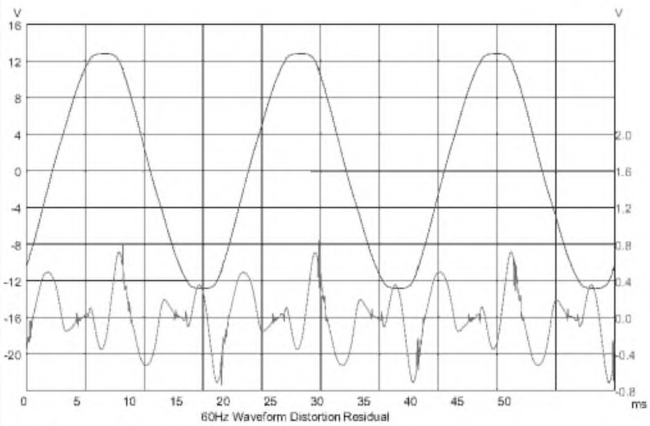

FIGURE 7: 60Hz AC line residual distortion.

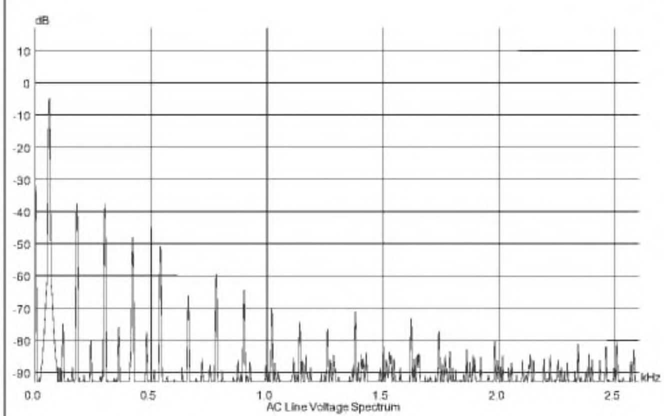

FIGURE 8: Spectrum of 60Hz AC line voltage.

devices. EMI regulations are designed to prevent real-world problems by providing a sufficient electromagnetic compatibility (EMC) margin between the worst-case emission and susceptibility limits (*Fig. 3*). The specifications change periodically, so the information presented here is a snapshot of what I believe to be the limits at the time of this writing.

Conducted EMI can be in the form of voltage or current. Power-line-conducted emissions are evaluated from 150kHz to 30MHz using an RF current probe or a line-impedance stabilization network (LISN) connected in series with the power lines to the equipment. The LISN establishes a consistent 50Ω impedance that allows for repeatability of test results. The conducted emissions are connected to measurement equipment through an RF connection port on the LISN. Some applications are capable of performing magnetic-field measurements down to 10kHz using H-field probes.

Radiated EMI is assumed to occur in the upper frequencies, from 30MHz to 10GHz. It is measured with a variety of broadband antennas at a standard distance (3–30m) from the equipment under test (EUT).

Radiated emissions are measured in dB above 1 microvolt per meter (dBμV). Conducted emissions are measured in dB above 1 microamp (dBμA), but commercial test data is presented in dBμV, measured across the standardized 50Ω measurement impedance. (Aerospace conducted emissions data is presented in dBμA.)

EMI emission limits are also placed on *unintentional emitters* in order to protect the radio communications spectrum. An unintentional emitter is a product that intentionally generates RF for use within the device, or that sends RF signals to associated equipment via shielded interconnecting wiring, but which is not intended to emit external RF energy by radiation or induction. Examples of unintentional emitters are TV and radio receivers, CD and DVD players, computers, set-top cable boxes, cable selector switches, and so on.

THE CE MARK

The CE Mark is mandatory for products marketed in the European Union. All manufacturers are responsible to see that their products conform with the appropriate safety, electrical, EMC and RFI emission, and immunity requirements for the EU common CE label.

Most equipment manufacturers tend to design to the worst-case conditions of all established standards, to reach the broadest possible market.

EU standard European Norms EN55013 (CISPR13) sets forth the limits for and methods of measurement of the interference characteristics of sound broadcast receivers, television receivers, and associated equipment. It also covers cable and satellite receiving equipment. Associated equipment is either intended to be connected directly to AM, FM, or TV receivers, or to generate or reproduce audio or visual information (audio amplifiers, active loudspeaker units, turntables, CD and DVD players, magnetic recording and playback systems, and electronic musical instruments).

The conducted and radiated emissions limits for CISPR13 are shown in *Figs. 4* and *5*, respectively. *Figure 4* also shows an example of the conducted interference measured with a resistive load (lower curve), and that of a typical switching power supply. Information-technology equipment such as personal computers are governed by a different standard (EN55022/CISPR22), even if they are intended to be connected to a TV broadcast receiver.

The EU and US Federal Communications Commission (FCC) limits for conducted and radiated emissions are based upon the end application of the device. Units intended for residential use must comply with the more stringent Class B limits, which are 10dB tighter than the industrial/commercial Class A limits.

Military limits (MIL-STD-461 and -462) are even tougher (almost 30dB tighter than industrial/commercial), cover a wider range of frequencies, and include more stringent immunity limits. Commercial aircraft use a similar EMI specification, called DO-160. These aerospace-radiated interference tests are measured 1m from the EUT, while the FCC and EU tests are performed with the antenna 3–30m from the EUT.

EMI IMMUNITY

In addition to the emissions limits, requirements exist that restrict the degree to which equipment can be adversely influenced by radiated and conducted noise. These specify the EMI levels that products must tolerate without misoperation. This is called *immunity*, or *susceptibility*. The degree to which power-line harmonics and EMI are audible in audio equipment, and even the definition of this misbehavior, is the subject of much debate, typically split along the subjective/objective dividing line. However, there is no question that the power supply in a piece of audio equipment is part of the audio signal path.

The EU immunity limits for audio and TV receivers and associated equipment are governed by EN55020/CISPR20. Conforming equipment may be required to show immunity to RF field strengths up to 2V per meter. In the US, the FCC specifies emission standards, but tends to be more hands-off toward immunity unless public safety is involved. This will undoubtedly change, what with the many IEEE committees at work to align the various worldwide standards.

A radiated immunity test is designed to verify the immunity of equipment to electromagnetic fields generated by radio transmitters, transceivers, cellular phones, and various industrial electromagnetic sources. You cannot logically expect any radio receiver to be immune to the very signals it is expected to receive, so EN55020 requires broadcast receivers to demonstrate test immunity up to 150MHz. RF immunity tests for associated equipment may extend up to 1000MHz (1GHz).

The Immunity to Conducted Electrical Transients test is designed to verify immunity to bursts of short-duration, fast-rise-time transients that may be generated by the switching of inductive loads or electrical-system contactors. The transients are applied directly to the power mains, and capacitively coupled to signal lines. Many manufacturers also include surge-suppression devices in their designs.

AC-LINE HARMONIC TEST DATA

EMI testing requires specialized equipment and certified test facilities. Since manufacturers do an excellent job of keeping RF energy out of their equipment, I have limited my testing to 60Hz AC line harmonic measurements.

Figure 6 shows the 60Hz AC-line-voltage waveform (left trace) compared

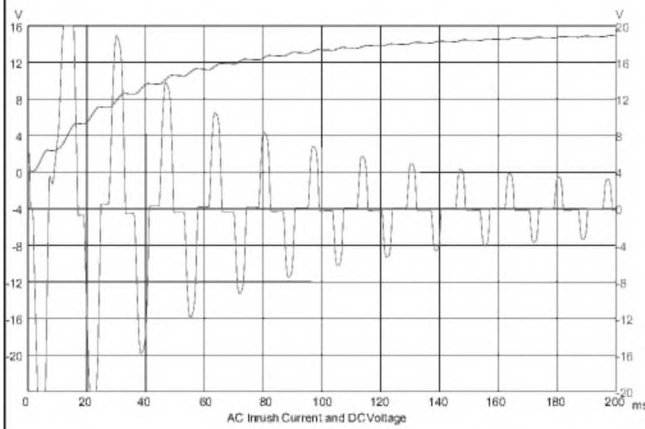

FIGURE 9: DC supply voltage versus AC inrush current.

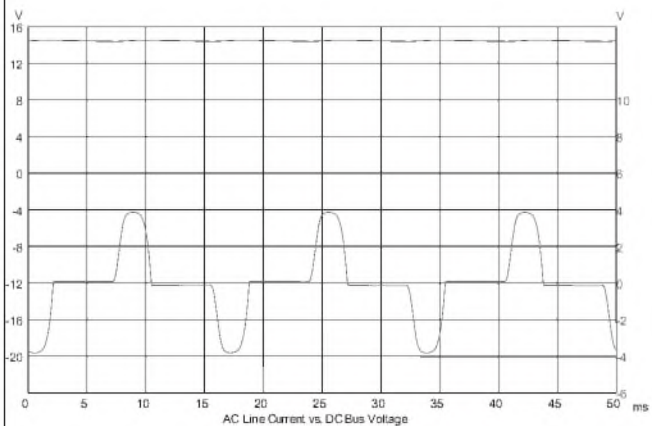

FIGURE 10: DC supply voltage versus steady-state AC current.

with the pure 60Hz sine wave from my distortion test set. I adjusted both waveforms in amplitude so their RMS values at the scope inputs are equal. You can see how harmonics have flattened the top of the AC-line-voltage waveform. The third, seventh, and eleventh harmonics flatten the peak of the fundamental 60Hz. The fifth, ninth, and thirteenth harmonics add to the peak, but not enough to offset the large effect of the third.

The distortion residual waveform for the AC-line voltage is shown in *Fig. 7*. The upper waveform is the AC voltage, and the lower one is the monitor output (after the THD test set notch filter) (not to scale). This distortion residual signal shows a series of odd 60Hz harmonics, with additional noise. The THD on this particular Sunday morning is 3.15%. During the week, the THD usually increases to about 3.5% as industry goes to work.

The spectrum of the 60Hz line voltage is shown in *Fig. 8*, from zero to 2.6kHz. The second, third, fourth, and fifth harmonics relative to the 60Hz fundamental measure −70dB, −33dB, −75dB, and −33dB, respectively.

Next, I'll investigate the line currents in a typical full-wave rectified capacitor-input DC power supply. The test setup I used looks at the DC voltage across a 2A DC load, and the AC current at the secondary of the power-supply step-down transformer. This particular 12A supply has about 60,000µF of filter capacitors, and is used to power car stereos and other 12V DC devices for bench testing.

HIGH INRUSH CURRENT

Figure 9 shows the DC voltage and transformer secondary AC current when the power supply is first energized into the 2A DC load. I used a current transformer with a response range of 25Hz to 5kHz to measure the AC current. The AC inrush current is very high for the first few cycles while the filter capacitors charge, then it decreases exponentially.

Figure 10 shows the steady-state condition, with the capacitors fully charged and supplying the 2A load. The 4A AC current peaks into the full-wave rectifier are not continuous over each AC cycle. The diodes conduct only when the voltage across the filter capacitor falls below the instantaneous transformer-secondary voltage, in a series of opposite-polarity pulses spaced 8.33ms apart. The height and width of these pulses increase in proportion to the load being supplied.

In *Fig. 11*, I used AC coupling on the DC-voltage input (top trace) to view the AC ripple on the DC voltage. The bottom trace is still the steady-state AC line current. You can see where diode conduction begins at each valley in the ripple voltage.

Compare the AC-current spectrum in *Fig. 12* with the AC line voltage spectrum in *Fig. 8*. While the line-voltage THD is 3.15%, the transformer-secondary AC-current THD is a whopping 72%. The second harmonic is very prominent at –41dB below the fundamental 60Hz, with the third almost as high (–2dB) as the fundamental. The fourth and fifth measure –47dB and –13dB, respectively. There is a lot more noise and hash in the spectrum as well. These high harmonic currents are impressed back through the power transformer onto the AC line, where they can adversely affect other connected equipment.

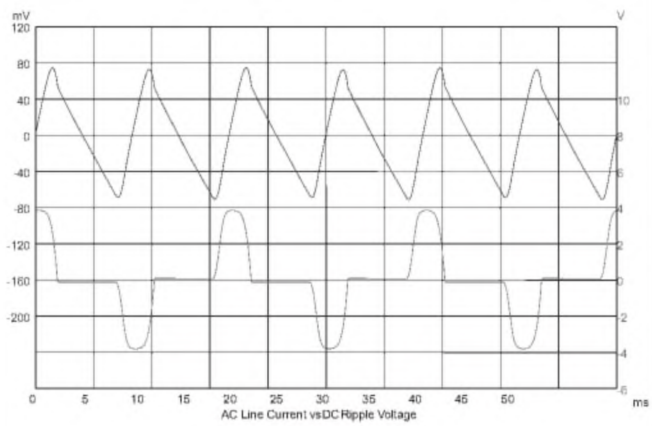

FIGURE 11: DC ripple voltage versus steady-state AC current.

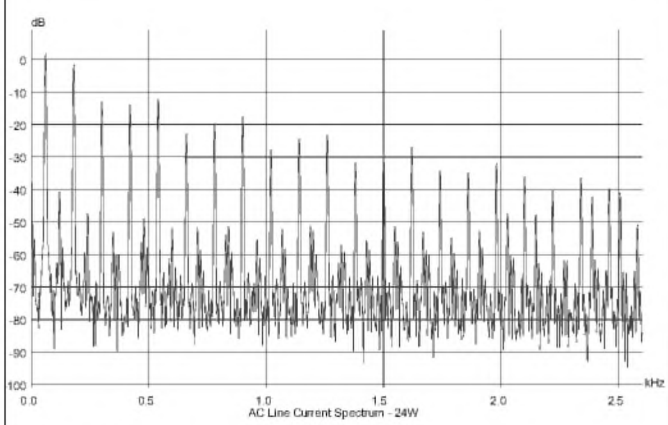

FIGURE 12: Spectrum of 60Hz AC line current.

FILTERS AND FIXES

A dedicated line from your service panel to your audio equipment (I call it "star sourcing," similar to star grounding) can do wonders to keep noise and harmonic currents out of your audio equipment. With a dedicated supply, your amplification need only endure the noise from its system mates, rather than the hash from the washer, garage-door opener, and furnace.

My own electrical supply consists of a dedicated 20A line from a dedicated circuit breaker, to a single heavy-duty outlet pair near my audio setup. I made an outlet extender from a four-gang steel electrical box, a heavy-duty Hubbell three-prong plug and a short length of ten-gauge three-conductor power cord. The extender box contains three 20A Hubbell hospital-grade (orange) duplex outlets, star-wired from a 20A switch using 12-gauge copper wire. The integral switch allows me to operate all the audio components at once. The extender-box plug allows me to disconnect the audio system completely in case of a thunderstorm, protecting it from lightning-induced surges.

If your outlets have spring clip "push-in" connections, do not use them. Terminal screws provide more surface area and a higher compressive force, for minimum resistance. I connected an individual green ground conductor to each outlet ground screw, then connected the first outlet ground to the metal box as the required safety ground. It was not easy to find a conventional four-outlet cover plate for the box. I found four-gang Decora™ plates, but none for four traditional receptacles. My solution was to make a functional cover by cutting one edge of two double outlet cover plates so they fit tightly over the steel box.

CONNECTION ORDER

I plug my main and subwoofer amplifiers into the outlet closest to the line-cord switch. Next come my rear channel amplifier and preamp. The CD player and cassette deck go into the third outlet. With all six outlets taken, I had to plug my turntable into a switched outlet on my preamp. The idea is to have the equipment with the highest current draw closest to the line-cord side of the box.

Some components have a detachable three-conductor IEC cable/receptacle, while others have an integral two-conductor power cord. In the latter case, if the plug is not polarized, you can experiment with the plug polarity to determine which orientation produces the lowest hum. Work your way back from the power amplifier, selecting the orientation that results in the lowest hum and noise. Do this without your interconnects plugged in, because you are first trying to eliminate the hum caused by AC-line ground loops.

Do not defeat polarized plugs or three-conductor line cords with a ground-lifting "cheater" plug. If you do this and a fault occurs, the chassis of your equipment could end up at 120V line potential, and your audio interconnects could be forced to carry huge 60Hz fault currents. At worst, it could cause a fire and damage some of your audio components. At best, the fault current on the shields of unbalanced interconnects will induce a high level of 60Hz hum into preamp or amplifier inputs, resulting in damage to your speakers, not to mention your hearing.

Once you have optimized the AC power, you should repeat the process with each interconnect in place in turn, starting from the power amplifier and working back to each audio source. Finally, find the quietest connection for your turntable grounding wire.

CROSS-COUPLING PREVENTION

In order to prevent cross-coupling of hum and noise between your components, make sure that the signal interconnects, the speaker cables, and the power cords are all kept away from each other. Power cords, speaker cables, and interconnects should only cross at right angles, to minimize pickup. I have seen articles suggesting that interconnects and cables should not even lie on the floor, so you can try some of those recommendations as well. Many audio racks have wiring channels that provide wire separation and also make for a neat installation.

There are many power conditioners and local filters available for audio systems[8], some of them quite expensive. Their purpose is to reduce the emission of EMI (suppression) and to prevent EMI from entering the electronics (noise immunity). The proper selection of an EMI filter encompasses mechanical packaging, leakage current, insertion loss, rated voltage and current, conformity testing, and compliance approvals. As I mentioned earlier, manufacturers are responsible for the conformity of their products to all relevant EMC specifications.

Power conditioners or power-line filters may offer further sonic improvements, but they may also insert additional series impedance in your AC supply lines that adversely interacts with filtering designed into your equipment in the first place. In any event, your first step should be to make sure the AC power to your audio setup is as good as possible. ❖

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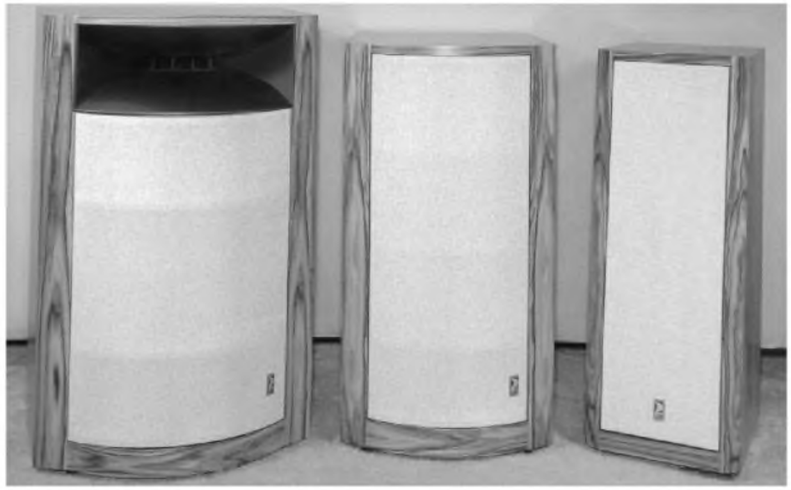
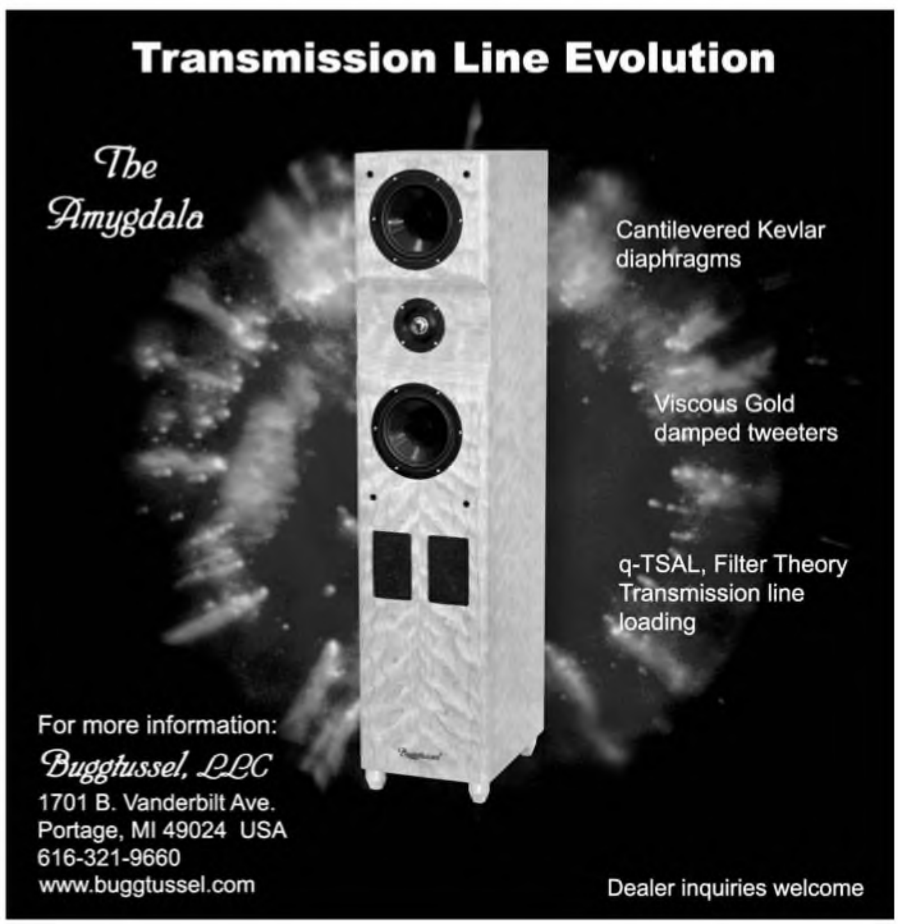




Product Review

Sony MDS-JE440 MiniDisc Deck

Reviewed by Gary Galo

Sony MDS-JE440 MiniDisc Deck. Sony Electronics, Inc., Consumer Audio Division, 1 Sony Drive, Park Ridge, NJ 07656. 1-800-222-7669 or (941) 768-7669, FAX (941) 768-7790, www.sel.sony.com. Suggested retail price: $250.

PHOTO 1: The Sony MDS-JE440 MiniDisc Deck and supplied remote control. This entry-level component MiniDisc recorder is intended for use with a complete audio system, and matches the styling of Sony's CDP-XE400 and XE500 CD players.

The MDS-JE440 is Sony's bottom-of-the-line MiniDisc deck (*Photo 1*). It is intended to be used as a component within a complete home audio system, and not for portable use (for portable field recording, Sony manufactures a complete line of portable MiniDisc recorders with microphone inputs). The MDS-JE440 is styled to match Sony's CDP-XE400 and XE500 Compact Disc players (reviewed in *audioXpress* August 2001).

MINIDISC BASICS

The MiniDisc has been around for nearly ten years. Sony introduced it in 1992 as a means of making digital recording affordable to the consumer. The MiniDisc has the same sampling rate and bit rate as the Compact Disc. But, the MiniDisc is only 2½″ in diameter. In order to achieve the same recording and playing time as the CD, a severe amount of data compression—about 5:1—is used during the recording process.

Sony's compression system is known as ATRAC, for Adaptive TRansform Acoustic Coding. In developing the ATRAC system, Sony employed psychoacoustic principles to determine what types of information loss are the least readily discernible to the human ear. In the early years of the MiniDisc, the format's sound quality left a great deal to be desired. The MiniDisc seemed like a giant step backwards—at a time when the CD was beginning to sound like music, the MiniDisc was a return to the harsh, edgy, dry sound of early digital audio.

Much has changed since the introduction of the format. Sony has continued to improve the ATRAC system, and their compression algorithms have evolved to the point where the MiniDisc is now gaining some respect, especially among those who need a portable recording system of reasonably high quality. The MDS-JE440 uses Sony's DSP Type R algorithms, which Sony calls ATRAC 3.

PHOTO 2: Sony Premium Gold MiniDisc. The media is only 2½″ in diameter, and the case measures only 2 13⁄16 × 2 11⁄16″.

RECORDING

The MiniDisc is a magneto-optical recording format using the Sony Magnetic Field Modulation system, which uses both heat and magnetism. During recording, the laser beam heats the recording medium, while a magnetic field applied to the other side of the disc leaves a permanent impression in the particles in the disc's recording layer. This system is a significant departure from conventional magneto-optical (MO) recorders, as explained in a link on Sony's website:

PHOTO 3: **Inside the MDS-JE440. A handful of proprietary Sony chips make up most of the circuitry.**

http://www.sel. sony.com/SEL/rmeg/mediatech/techspec/techMD.html.

Conventional MO recorders employ a fixed magnetic field, switching the laser beam on and off. With this system, a portion of the disc must be completely erased before you can record on it. This requires separate erase and record passes, doubling the time it takes to make a recording, or separate erase and record lasers, which significantly increase the complexity and the cost. The MiniDisc overcomes this problem by keeping the laser beam constant and varying the magnetic field.

PLAYBACK

In playback, a low-level laser beam is used, one that does not heat the disc. The changes in magnetic polarity cause the polarization angle of the reflected light to change. Pre-recorded MiniDiscs are manufactured using a molding process, but the reflections from the molded surface are compatible with the MiniDisc's laser pickup. All MiniDisc players are designed to play both pre-recorded and recordable MiniDiscs.

HHB, manufacturer of some of the finest professional CD and MiniDisc recorders and media available, has published an excellent guide to the various current digital recording formats. This document is available for download in .PDF format. Go to http://www.hhb.co.uk/usa.htm, and click on "Brochures, Manuals and Ads," then click on "Brochures." Look for "A Guide to Choosing and Using Digital Audio Recording Media."

Photo 2 shows a Sony Premium Gold MiniDisc, which is housed in a small plastic cartridge measuring only 2 15⁄16 × 2 11⁄16″. These premium MiniDiscs are manufactured with a shock-absorbing mechanism that reduces the transmission of vibrations made by the recorder or player to the disc itself. They are available five to a pack for around $15 in retail stores. Sony and several other manufacturers make cheaper media for less critical applications.

INSIDE THE DECK

Photo 3 shows the inside of the MDS-JE440. Nearly all of the circuitry is contained in proprietary Sony surface-mount integrated circuits. The A/D and D/A converters use Sony's Wide Bit Stream technology. The A/D converters are 24-bit, and the Hybrid Pulse D/A converter operates in conjunction with an 8× oversampling digital filter.

Surface-mount IC op amps are used for the analog amplification circuitry. My unit was supplied with 4570 types—these are not listed on the NJR website, but it is probably safe to assume that they are superior to the usual 5532 or 4558 types still found in some consumer gear. The MDS-JE440 contains a conventional (i.e., non-switching) power supply.

Photo 4 is a close-up of the mechanism. The laser assembly faces upward, toward the bottom of the MiniDisc. The magnet assembly is located above the laser assembly.

Photo 5 shows the same close-up with a MiniDisc inserted, and the play-

PHOTO 4: **Close-up of the mechanism, showing the laser and magnet assemblies.**

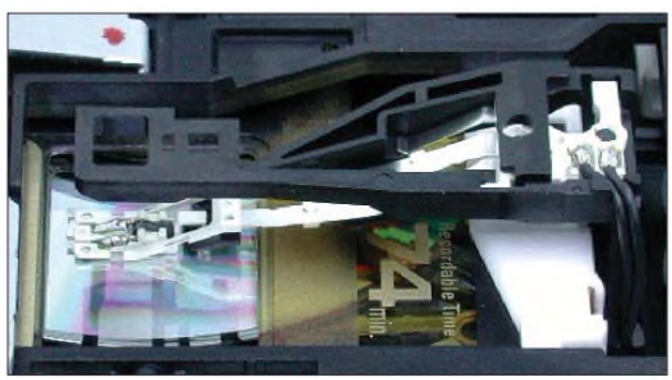


PHOTO 5: **Close-up of the mechanism with disc inserted and the deck in the record mode. The magnet assembly makes physical contact with the top side of the MiniDisc.**

TABLE 1
MANUFACTURER'S SPECIFICATIONS

System: MiniDisc digital audio system
Media speed: 400 to 900 rpm (CLV)
Error correction: Advanced Cross Interleave Reed Solomon Code
Sampling frequency: 44.1kHz
Coding: ATRAC-3
Modulation system: EFM (Eight-to-Fourteen)
Number of channels: 2 (stereo)
Frequency response: 5Hz–20kHz, ±0.3dB
Signal-to-noise ratio: 96dB
Wow and flutter: Below measurable limit
Analog input: 500mV RMS; 47kΩ
Headphone output: 28mW; 32Ω
Analog out: 2V RMS; 50kΩ
Power consumption: 15W
Dimensions: 17 × 3¾ × 11¼″
Weight: 6 lbs 10 oz

er operating in the record mode. During recording, the magnet assembly actually makes physical contact with the top side of the disc. Like Sony's low-end CD players, the MDS-JE440 is quite solidly built.

FEATURE-LOADED DECK

The MDS-JE440 comes with stereo analog inputs and outputs, plus a Toslink optical digital input. (A more expensive model, the MDS-JE640, also has a Toslink digital output, plus Toslink and S/PDIF coax digital inputs.) The digital input is intended for making digital copies of CDs, DATs, or any other digital format from a player with a Toslink output.

Making digital copies of commercial CDs that you already own is perfectly legal as long as the copies are for your personal use. Making copies of your friend's CDs is not legal. The MDS-JE440 uses the Serial Copy Management System, which allows you to make a first-generation digital copy of a commercial CD, but you will not be able to make subsequent copies from the first-generation copy. The MDS-JE440 has a built-in sampling rate converter, allowing you to make digital copies from 32kHz or 48kHz sources, as well as the standard 44.1kHz.

In the conventional stereo mode, you can record up to 74 minutes on a MiniDisc. But, the MDS-JE440 has three additional modes that allow longer recording time. The LP2 and LP4 stereo modes allow two or four times the normal recording time, respectively. There is a corresponding loss of fidelity, however, since the level of data reduction increases at a rate proportional to the increase in recording time.

There is also a mono mode, which gives double the recording time, but with the same fidelity as the normal stereo mode. Recordings made using the LP2 or LP4 mode can only be played on MiniDisc players supporting the LP format (not all do).

Using the basic recording and playback functions on the MDS-JE440 is as simple as operating a cassette deck. The bar-graph recording level indicator is not active until you press the LEVEL/DISPLAY/CHAR button. The deck allows you to adjust the recording level for both analog and digital inputs.

If you have a CD that was recorded at a low level, the MDS-JE440 allows you to change the level while making a direct digital copy. I know of few professional CD recorders or DAT recorders that allow you to change recording levels while recording via the digital inputs (HHB, Marantz, and Sony have recently introduced professional CD recorders with this feature). This can be an extremely useful feature, and I'm amazed to find it in a low-cost consumer product (and I wish that I had it in my professional CD recorder).

When the level indicator is not active, time and track information is displayed. Unlike most DAT or CD recorders, the level indicator will not function during playback.

The MDS-JE440 is loaded with features. The Synchro-recording feature automatically copies track numbers from a CD when digital copying is used. You can manually add index points during the recording process by simply pressing the RECORD button.

During playback, you can select specific tracks either with the remote control or by turning the AMS (Automatic Music Sensor) button. This is the same AMS system used by Sony in their low-cost CD players. The deck allows you to repeat a specific track, and also has shuffle play. You can even produce your own program by selecting specific tracks for playback, in whatever order you choose.

Editing functions are also included on the MDS-JE440. The deck allows you to erase a specific track or even a portion within a track. You can also divide a track already recorded by adding a new track number within an existing track. You can also combine tracks by eliminating a previous track number. Whenever you add or remove track numbers, the remaining tracks are automatically renumbered. You can also move tracks around, changing the order in which the various tracks appear, and you can add names to previously recorded tracks. The MDS-JE440 even allows you to change the recorded level of any track, after that track has been recorded! You can reverse any editing operation with the UNDO function. A full-featured remote control is supplied with the deck, and runs on a pair of AA batteries (supplied by Sony).

HOW DOES IT SOUND?

I evaluated the sonic performance of the MDS-JE440 by making copies of some of my reference CD tracks onto MiniDisc via the Toslink digital interconnect and listening to the results using the player's analog outputs connected to my main system. The MiniDisc was not intended to compete sonically with the Compact Disc, and indeed it does not. But, I was surprised that the results were as good as they were.

The MiniDisc dubs show a dryness and a loss of ambience. The soundstage is also reduced in size, and there is a slight edge in the high frequencies and a graininess added to the texture. Note, however, that I auditioned this unit with my reference system. A system such as mine is an unlikely place for a MiniDisc deck to reside. When matched with components of comparable price and quality, the degradation caused by the digital compression will be subtle, or may go unnoticed altogether.

I remember how distressed I was with the sound of CD players back in the late-1980s. The MDS-JE440 sounds considerably better than a stock Philips/Magnavox player from that vintage (a CDB460, for example; we still have dozens of these in service at The Crane School of Music, so finding one for comparison was no problem).

I also copied several CDs using the deck's analog inputs, fed from my preamp's tape outputs. The sound quality was not quite as good as the direct digital copies, but quite respectable nonetheless. Used via its digital or analog inputs, the MDS-JE440 is sonically superior to any analog cassette deck I've used.

One obvious place for the MiniDisc is in applications where portability is important. Many people who jog, walk, or otherwise like to spend time on the go with their music find portable CD players just a bit too large to be convenient. I often see people jogging while carrying a portable CD player in hand—a nuisance to be sure. Portable MiniDisc players are now available that are barely larger than the disc itself—they are truly "pocket-size."

The MDS-JE440 is an ideal recorder for making copies of CDs for portable use. At our music school, a number of students and faculty have found the MiniDisc format ideal for making recordings of rehearsals, and even audition tapes. Again, the MiniDisc sounds considerably better than an analog cassette, and is well-suited to this purpose.

Mated with a couple of affordable microphones and one of the small Behringer mixers (July 2001 review), the MDS-JE440 can form the core of a respectable low-cost digital recording system. Behringer's bottom-of-the-line MX602A, with a street price of less than $100, will be ideal where only two microphones are needed. The MiniDisc deck won't equal the quality of a DAT recorder, but unless you're pressing commercial CDs from your material, you may find the format to be quite satisfactory.

CONCLUSIONS

The MiniDisc has come a long way in the last ten years. It is not, and was never intended to be, a high-end, audiophile quality recording format. But, the sound quality of the MDS-JE440 is nothing short of remarkable for the price. The street price of the MDS-JE440 is now less than $200, and I find it quite amazing that this level of sonic performance, plus such an abundance of features, can be built into such an inexpensive consumer product.

There are many applications where the sonic performance of the MiniDisc will more than fit the bill. For those applications, the MDS-JE440 deserves a solid recommendation. ❖

Product Review

Circuit Specialists Digital MultiMeters

Reviewed by Gary Galo

Web-Tronics 9300GP and CSI9903 Digital MultiMeters. Circuit Specialists Inc., 220 South Country Club Drive #2, Mesa, AZ 85210, 1-800-528-1417 or (480) 464-2485; FAX (480) 464-5824, e-mail: info@cir.com, www.web-tronics.com. Price: $19 (9300GP); $29.95 (CSI9903).

PHOTO 1: Circuit Specialists' 9300GP and CSI9903 Digital Multimeters offer solid performance at bargain prices. The feature-loaded CSI9903 (right) may be the best buy in test equipment from any vendor.

I have often received queries about projects I've published in this magazine (and its predecessors) only to find that the person having problems lacks even basic test equipment. The most basic—and essential—piece of test gear for anyone building or working with electronic equipment is a digital multimeter, or DMM for short. The cost of quality test equipment is lower than ever, thanks to manufacturing in the Far East.

Until recently, Taiwan and South Korea have been the most prolific suppliers of test equipment, much of it marketed by familiar American firms. Over the past couple of years, mainland China has become the source for a growing amount of test equipment. The result has been even lower prices and, in many cases, very good quality.

There are a number of US-based firms importing test equipment from China and placing their own names on it. Circuit Specialists is a company that may not be a household name to readers of this magazine, and their test equipment selection isn't as large as some of the better known suppliers. But, they do have some of the best prices around, on brands that include their own Chinese-manufactured Web-Tronics line, plus more familiar names such as Fluke, Goldstar, Wavetek, and Protek.

DMM FEATURES

The two Web-Tronics DMMs I reviewed offer excellent performance at unbelievably low prices (*Photo 1*). The 9300GP sells for only $19, yet is a full-featured, accurate DMM. The 9300GP tests AC and DC voltage, resistance, DC current, and bipolar transistor HFE.

Six resistance ranges are included, plus five ranges for DC voltage, two for AC voltage, and five for DC current. Only one range is needed for transistor HFE. The manufacturer doesn't specify accuracy on all ranges, but the meter generally reads within about 1% of my bench reference (a Tenma meter from MCM Electronics). The 3½ digit display features large 21mm numeral height for easy reading.

The CSI9903, which sells for $29.95, is auto-ranging and tests AC and DC voltage, resistance, AC and DC current, and transistor HFE. This meter also measures frequency and temperature in degrees Fahrenheit (a temperature probe is included). The temperature feature can be extremely useful in determining whether you've used sufficient heatsinking on transistors and ICs.

A range hold button overrides the auto-range feature, allowing manual range selection by toggling this button. The meter also includes data hold and an AC/DC button that switches the resistance function between the continuity and diode tests. The CSI9903 reads within about 0.5% of my bench reference. The large 3¾ digit display also includes a bar-graph, which is handy when you need analog-type metering to monitor a peak indication (doing an alignment, as an example).

OPERATION

Both of these meters are well-built and include a rubber boot to protect the meter (at no extra charge). I was especially impressed with the firm, solid mechanical feel of the selector switches, a refreshing change from the flimsy switches I've encountered on some inexpensive test equipment.

These meters may not have much in the way of static protection. Last winter I walked across the carpeted floor in my listening room to check the DC offset of a power amp I was reviewing. We have forced-air heat, so it becomes quite dry in the winter. I failed to discharge myself, and when I connected the negative lead of the 9300GP to the negative speaker terminal, I drew a spark. That was the end of the meter. This was my own fault, and I'll be more careful with my CSI9903 (my 9300GP was out of warranty by then).

The 9300GP takes a 9V battery; two AAA batteries are required for the CSI9903. On both meters battery replacement requires the removal of screws on the rear panel—there's no "snap-in" cover for the battery compartment. Both meters have a built-in stand, consisting of a hinged plastic support on the rear of the meter. On the CSI9903, in particular, the plastic support is quite flimsy. If you push on the meter hard enough to turn the selector dial, you may break the plastic support.

The minor inconvenience of having to remove screws to change batteries and the less-than-robust support are two obvious places where corners have been cut to reduce costs. At these prices, something has to give, and Circuit Specialists has made the right compromises, I believe.

COMPANY OFFERINGS

Circuit Specialists offers a one-year warranty on both meters, plus a 30-day approval period. If you are not happy with your DMM, return it within 30 days for a full refund, excluding the shipping charges. How can you lose? The 9300GP is so cheap that there's no excuse for audio hobbyists not to own a DMM. But, the additional features on the CSI9903 make this one an even better buy—this meter has to be the best value in test equipment from any supplier.

Circuit Specialists offers a complete line of electronics parts, and their website is sure to be of interest to electronics builders. One item unique to Circuit Specialists is their RF Prototyping Board (#IF-RFB), which used to be sold as Ivan Board, and which I've recommended for a number of projects I've authored in *Audio Amateur* and *Audio Electronics*. One side has 0.1″ solder pads, and the other side is a solid ground plane. This is one of the best prototyping boards available. A 10″ square piece is $29.

Circuit Specialists does most of their business on the internet. Once they get your e-mail address, they'll alert you to special offers, which are usually available only if you order on their website. Recently, they offered the CSI9903 DMM absolutely free with any order of $30 or more (that's how I got mine)! *audioXpress* readers should do themselves a favor and pay a visit to the Circuit Specialists website. ❖

TABLE 1
MANUFACTURER'S SPECIFICATIONS—9300GP

RANGE	RESOLUTION	ACCURACY
DC VOLTAGE		
200mV	100µV	Not specified
2000mV	1mV	Not specified
20V	10mV	±1% +2 dgts.
200V	100mV	Not specified
1000V	1V	Not specified
DC CURRENT		
200mA	100nA	Not specified
2000mA	1mA	Not specified
20mA	10mA	±1.2% +2 dgts.
200mA	100mA	Not specified
10A	10mA	±1.2% +2 dgts.
AC VOLTAGE		
200V	100mV	±1.2% +10 dgts.
750V	1V	Not specified
Frequency response: 45Hz–450Hz		
Maximum allowable input: 750V RMS		
Response: Average responding. Calibrated in RMS of a sine wave.		
RESISTANCE		
200Ω	100mΩ	Not specified
2000Ω	1Ω	Not specified
20kΩ	10Ω	±1.2% +2 dgts.
200kΩ	100Ω	Not specified
2000kΩ	1kΩ	Not specified
20MΩ	10kΩ	±2% +10 dgts.

TABLE 2
MANUFACTURER'S SPECIFICATIONS—CSI9903

RANGE	RESOLUTION	ACCURACY
DC VOLTAGE		
300mV	0.1mV	±1% +5 dgts.
3V	1mV	±1% +5 dgts.
30V	10mV	±1% +5 dgts.
300V	100mV	±1% +5 dgts.
1000V	1V	±1% +5 dgts.
AC VOLTAGE		
3V	1mV	±1.5% +8 dgts.
30V	10mV	±1.5% +8 dgts.
300V	100mV	±1.5% +8 dgts.
750V	1V	±1.5% +8 dgts.
DC CURRENT		
300µA	100nA	±1.2% +5 dgts.
3mA	1µA	±1.2% +5 dgts.
30mA	10µA	±1.2% +5 dgts.
300mA	100µA	±2% +8 dgts.
20A	10mA	±2% +8 dgts.
AC CURRENT		
300µA	100nA	±1.5% +5 dgts.
3mA	1µA	±1.5% +5 dgts.
30mA	10µA	±1.5% +5 dgts.
300mA	100µA	±3% +5 dgts.
20A	10mA	±3% +5 dgts.
RESISTANCE		
300Ω	0.1Ω	±1.5% +5 dgts.
3kΩ	1Ω	±1.5% +5 dgts.
30kΩ	10Ω	±1.5% +5 dgts.
300kΩ	100Ω	±1.5% +5 dgts.
3MΩ	1kΩ	±5% +10 dgts.
30MΩ	10kΩ	±5% +10 dgts.
FREQUENCY		
300kHz	100kHz	±3% +5 dgts.
TEMPERATURE °F		
0–1832°F	1°F	±1% +6 dgts.

HYBRID AMP

In Fig. 2 of the article "A Hybrid Tube/MOSFET SE Amp" (May '01 *aX*), I believe the drain and source of Q4 are reversed. Also, the value of R14 may require some tweaking, as Vgs is not terribly consistent from unit to unit and varies with output current, as well.

I enjoyed the opening salvo of "The Virtual Crossover (p. 8)." I'd encourage you to keep publishing articles that contain significant math. As we get deeper into the digital age, I suspect that a copy of Mathematica or Matlab and a compiler for the DSP of choice will become as much a part of our arsenal as a good oscilloscope and soldering iron!

Rudy Chalupa
Pleiades Audio + Electronics

I read the article on the Hybrid Tube/MOSFET SE Amp, by Generoso Cozza, with great interest. This seems to be a very practical melding of tubes and solid-state electronics.

I think the schematic on page 20 has an error in it. Q1 and Q2 are shown as NPN transistors but they should be PNPs; the emitters should go up towards the resistors with the collectors going to the tube plates. Also, the base of Q2 goes to its own collector and the base of Q1.

After breadboarding the amp, I have a question for Mr. Cozza on the constant-current source (Q3, D1, P1, and R5). I noticed that the DC offset of the amps output seems to drift with output amplitude and also somewhat with swings in the supply rails. Could this be improved by using an LM334 adjustable current source to feed the tube while keeping the zener for the FET current source?

The noise figure for the 334 is higher, but I wonder whether it would be audible?

Also, does anyone know where you can buy small quantities of heatsinks without being taken to the cleaners?

Bob Cleary
Quincy, Mass.

Generoso Cozza responds:

Bob and Rudy, I'm happy that you like the Hybrid Tube/MOSFET Amp. You are right, the initial schematics published in previous issues of audioXpress *includes some print errors.*

For the benefit of all readers I'm including an updated schematic (Fig. 1) *of the amp. For those that like the Internet, I suggest you take a tour around my web pages (http://digilander.iol.it/essentialaudio) for additional information.*

The split of the constant-current source into two separates, one for the input stage and the second for the output stage is a good idea that I tested with positive results but with limited audible differences.

Since I'm in Italy, I consider of limited interest to inform you about a local heatsink provider.

FIGURE CORRECTIONS

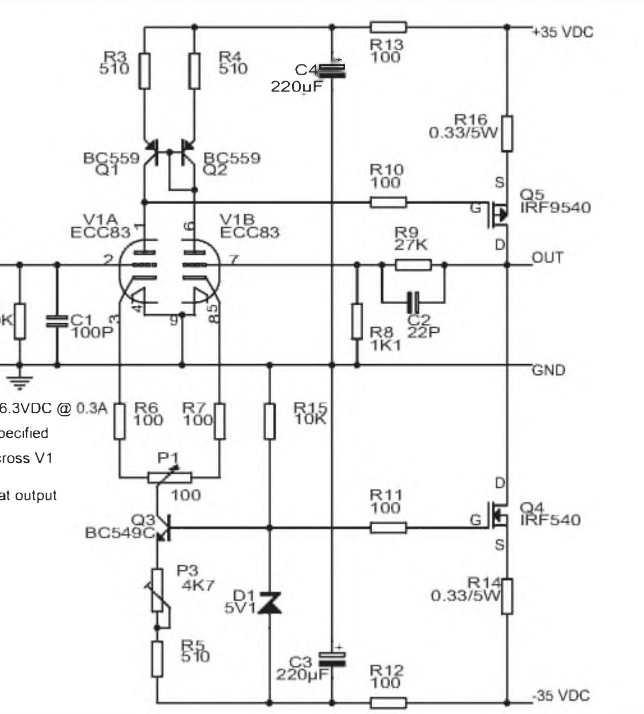
I found a few errors in my articles in Aug. '01 *aX*. In "A High-Quality Two-Channel Chip Amp" (pp. 12–14), I used the wrong component designations in how it works. Figure 1 shows the R channel, but the description is for the L channel (*Fig. 2*). I'm sure most readers will figure out that substituting 2 instead of 1 for the lead digit will correct it (C101 becomes C201, and so on).

I am including a revised Fig. 3 for my Roksan Caspian review, p. 59 (*Fig. 3*). The text changes made it into the final article, but not the revised Fig. 3.

Figure 1 in the review of the Music Hall MMF-2.1 is actually Fig. 2, and the Fig. 2 shown is actually Fig. 3, which was reprinted correctly as Fig. 3. The real Fig. 1 (*Fig. 4*) became lost somehow, since it was correct in the edited manuscript I reviewed.

Chuck Hansen
cmhj@concentric.net

FIGURE 1: Hybrid circuit.



100W AMP WIRING

Regarding your 100W triode amplifier (*GA* 3/00), I am unsure of the connections at the heater of the 5687 tube (V1). I looked at my data sheet for this tube, and pins 4 and 5 do indeed connect to either side of the heater. In your schematic, pin 9 is connected to

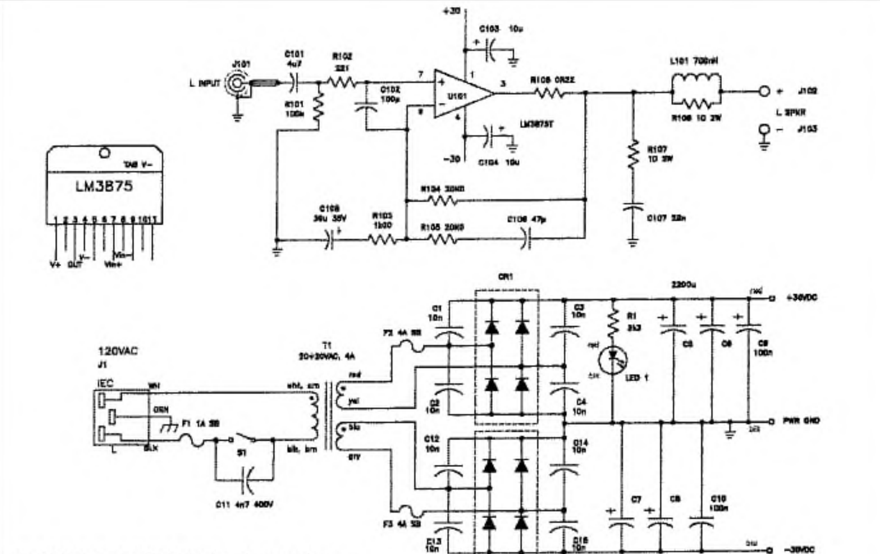


FIGURE 2: LM3875 R channel amp.

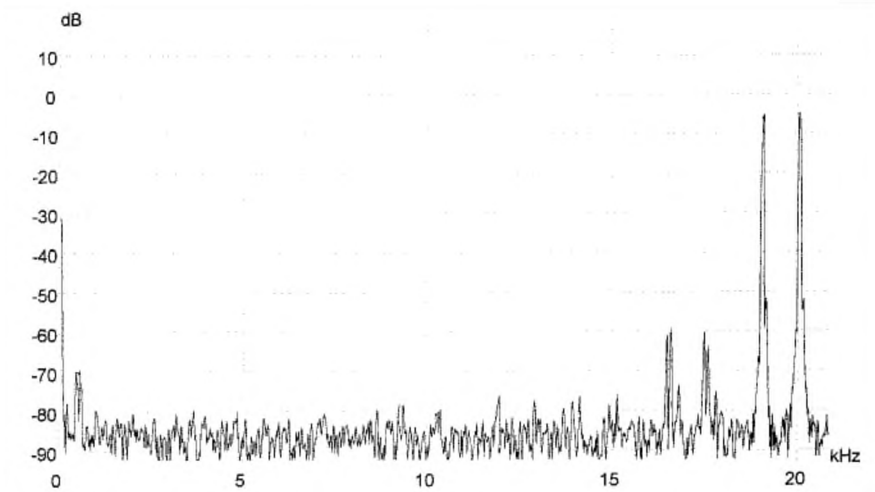


FIGURE 3: Roksan Caspian revised intermodulation.

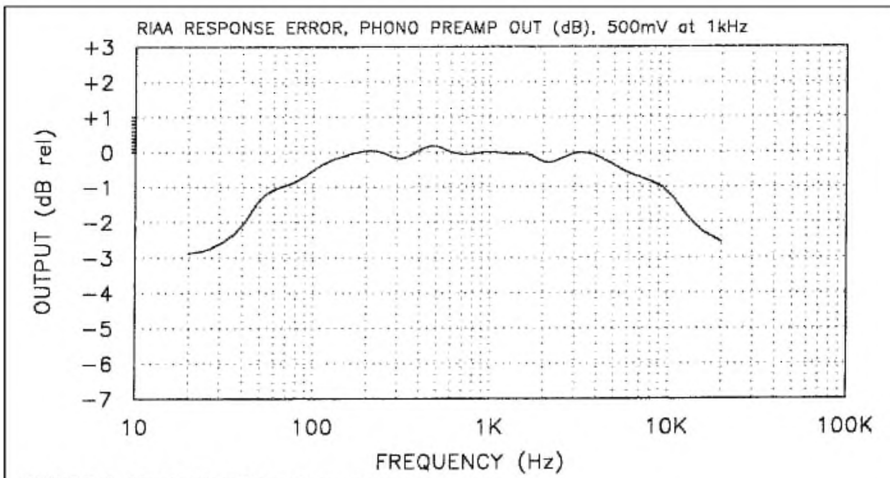


FIGURE 4: Music Hall frequency response.

the center of the heater, and in my book this connection is made at pin 8. I just wanted to clarify this connection with you before I start to build the amp.

How does the heater receive a flow of electricity if its supply pins are tied together? I assume it has something to do with the connections at the center of the heater, which is something I'm not familiar with.

I appreciate your time, and let me say that I look forward to constructing what promises to be a very rewarding amplifier. I have been waiting for a few years for just the right project to come along, and this is it!

Blaine Konkle
Ontario, Canada

Joseph Norwood Still responds:

You indeed have a very awesome project ahead of you and I wish you the best on your efforts to build a truly great amplifier. The amplifier has great sound and provides textbook perfect square waves at 100Hz and 10kHz (100W).

You are right about the 5687 wiring diagram being incorrect. Pins 4 & 5 are used only for 12.6V AC operation and pin 8 is only used for 6.3V AC operation. In the 6.3V AC mode, pins 4 and 5 are paralleled and the circuit is completed at pin 8. As explained in the text, two 1Ω resistors are used at these heater connection points to provide 5V AC operation of the heaters. Also, note that pins 6 and 9 are incorrectly wired and pin 6 should be shown tied to the cathode of V1B and pin 9 tied to the plate of V1B.

IMPORTANT: To avoid instant catastrophic damage to the output tubes, observe these precautions. Make certain the control grid circuit of the 6550s is wired as shown on the schematic diagram and none of the six control grids of the 6550s measures infinity resistance or shows evidence of an open circuit. The 2.7kΩ resistor connects to pin 5 of V4/V7. The 330Ω resistor is connected between pin 5 of V4/V7 and pin 5 of V5/V8, and also the 330Ω resistor is connected between pin 5 of V5/V8 and pin 5 of V6/V9.

NOTE: Before applying plate voltage to the 6550s make certain that –65V DC (approx.) is present on pins 5 of V4 through V9. After you determine that –65V DC is present on the control grids of all six 6550s, you may apply the +500V DC to the plate circuit of the 6550s via the output transformer.

I recommend that you order the 220Ω, 2W (R38–R43) and 50Ω, 4W (R32–R37) from RadioShack.com, 1-800-442-7221. The R32–R37 (50Ω, 4W) resistors are formed by paralleling two 100Ω, 2W resistors. These resistors must be metal-oxide film types and must be ordered from Radio Shack or NTE Electronics, as these are the only suppliers that have resistors suitable for high power/high voltage circuits. The Radio Shack part numbers for these resistors are 900-0794 (qty-24) and 900-0802 (qty-12).

The 220Ω, 2W grid no. 2 resistors of the 6550s must be tied directly to the plate side of the 50Ω, 4W resistor as shown on the schematic to prevent tube burn-out in case of failure of the 50Ω, 4W plate resistor.

I believe I have covered items that may not be readily obvious and will enable you to construct these amplifiers with few problems. If you have any more questions please do not hesitate to contact me and I will do everything I can to help you. I am very aware of the awesome project you are undertaking and I'm sure you are going to be the only one in the country (universe) with a 100W Triode Stereo System.

Please remember, at all times, you are working with lethal voltages and never leave the amplifier upside down when children are present. Very good luck to you and wishing you a successful outcome.

TRANSFORMERS

I do agree with a lot of Mr. Millett's conclusions in "Power Transformers for Audio Equipment" (June '01), but there are some points I would like to make. First, you can easily design EI and UI core transformers so you do not have any problems with stray magnetic fields; it's just a question about how close to (or rather far from) the saturation of the core you're calculating and making your transformer. You can easily check this yourself with a vario transformer and an AC milliamp meter.

Without any load on the secondary, measure the primary idle current as you go from 0 to 110% mains voltage in 5% steps. The first part of the curve is almost a straight line, then it begins to curve upwards and rises steeper and steeper until it becomes almost horizontal (only dependent on the DC resistance of the copper). Most commercial transformers are close to the steep end of the curve. I've made some myself that are below the steep part of the curve and located very close to the extreme high gain end of guitar amps without any problems.

Second, most commercial transformers are wound close to the magnetic limit in order to save copper (money) and in most applications this is absolutely no problem.

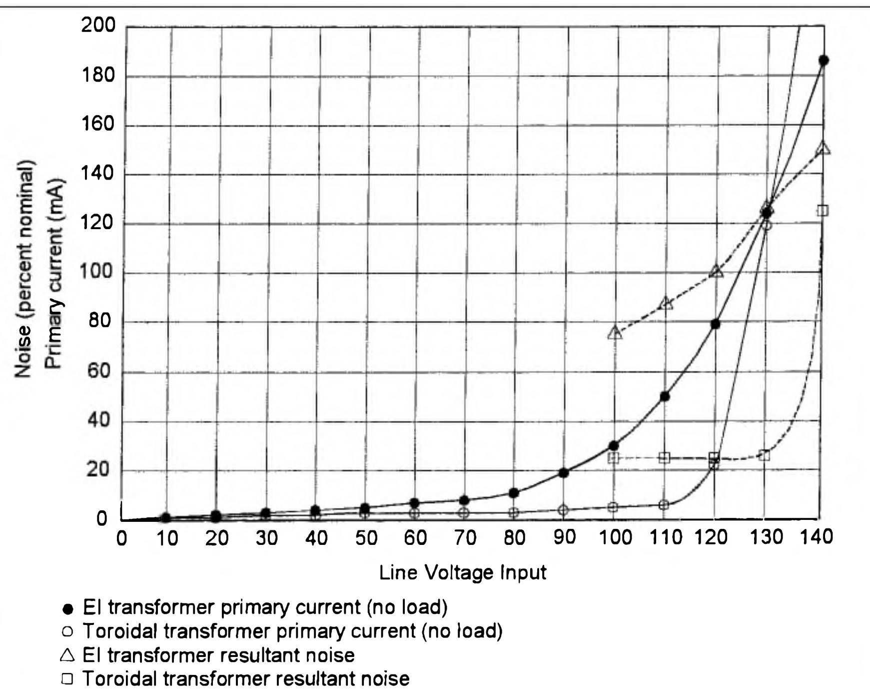


FIGURE 5: EI/toroidal transformer measurements.

Third, I've seen a good number of toroidal transformers behaving badly after the mains here in Europe was raised from 220V to 230V; they can stray, too. It's just that in most cases they behave much better than EI or UI transformers and therefore got the reputation that they are much better, but there's no guarantee that this is the case every time.

Per Byrgren
Denmark

Pete Millett's well-done article reminded me of something that may be useful to readers. C-core transformers can over time become mechanically noisy, with a kind of buzz that can be really annoying. This occurs because the cut ends of the core begin to oxidize. I had this problem with European equipment (Studer and others), where, as the author said, this kind of transformer is quite common, particularly with high power transformers used in big multitrack recorders or power amplifiers.

The remedy can be quite simple: cut the metallic band joining the two halves of the core. On a flat surface and with very fine sandpaper (400 or finer), polish the ends of the core, while keeping their flatness. Put a drop of oil on the ends to protect them against future oxidation and remount the transformer. Because the C-core should be in intimate contact, use the kind of plumbing bridge that can be screwed to hold the two halves. (Those made by Tridon in the USA can be found in any hardware store.)

Try the transformer before reinstalling it; from time to time, you may need to polish more. But keeping the flatness is really very important. Working on marble is not essential, but it helps. And it works as well when you want relap magnetic heads for a tape recorder. And a flat piece of marble is not really hard to find.

Philippe Trolliet
Montréal

Pete Millett responds:

Per Byrgren's comments about how the stray flux emanating from a transformer may be related to core saturation raises a good point that I hadn't thought much about. Although I disagree with the blanket statement that EI and UI-core transformers can easily *be designed so you do not have* any *problems, it does indeed make sense that as the core approaches saturation, the amount of stray flux increases dramatically. A saturated core presents much more resistance to magnetic flux, so more flux would tend to take paths through the air outside the core.*

*Always a skeptic, I performed an experiment to try and quantify this. As Per suggested, I measured the primary current (with no load on the secondary) of both the EI transformer and the toroidal transformer (*Fig. 5*) to try and see where the transformer cores saturate.*

With the EI transformer, the current does rise nonlinearly as the core starts to saturate, though there is not a very sharp "knee" to the curve. The toroidal transformer is similar, but has a very pronounced corner to the curve when saturation begins.

Next, I measured the noise coupled into the amplifier to see whether it correlated to the saturation point; my simplistic reasoning being that, if the noise was due to core saturation alone, the noise should pretty much track the no-load *primary current, as opposed to the* loaded *primary current, which is nearly linear (over the operating range) with input voltage.*

My results were not completely conclusive. It appears to me (again, referring to Fig. 5*) that there is a correlation between where the core saturates and stray-flux induced noise with the EI transformer. The problem is that (at least in this transformer) the onset of saturation is so slow that over the operating range of the amplifier, it's almost linear.*

The toroidal transformer is certainly another matter. When it enters saturation (abruptly), it obviously starts to radiate stray flux. This is easy to see on the graph. This confirms Per's third point very dramatically...if this transformer is run on a line voltage of 140V instead of the design-center 115V, there would be a 500% increase in stray-flux related noise in this amplifier! This would explain why a toroidal transformer designed for 220V with little margin would saturate, and radiate lots of stray flux, when run on 230V.

I'm not a transformer engineer, but I suspect that the different slopes of the saturation curves between the EI and toroidal transformer tells most of this story. Although *you could design an EI transformer to have such low flux density that it would emit as little stray flux as a comparable toroid (in other words, operate at the far low end of the curve), it would need to be bigger...and more expensive than a typical EI design. On the other hand, owing to the sharp knee in the toroidal transformer's saturation curve, you can achieve low stray flux with a toroid, right up to the point where the core saturates.*

The problem for the hobbyist (and the manufacturer who can't justify a custom transformer) is that you don't necessarily get to specify, or know, how close to saturation a given transformer is designed. Had I understood all this when I first specified the transformers, I would have achieved better results, but may also have wound up with an EI transformer that was too big, and more expensive than the toroid.

I think the bottom line is that a poorly designed transformer will perform poorly, be it toroidal or an EI transformer. I still think that in many cases a toroidal transformer is a better choice for use in audio equipment. To some extent, I'm sure, you get what you pay for, regardless of the construction (EI or toroidal) of the transformer.

Philippe Trolliet's advice on reconditioning a noisy C-core transformer sounds good to me. Though I've never had to address this issue myself, it does make sense that if the core ends become rusty, the magnetic path will develop more resistance, and the transformer will become less efficient, and noisy—not unlike dirty electrical contacts introduce resistance into an electrical contact.

MIXERS

Thank you for the review of the Behringer mixers and the article on building a tube compressor (July '01 *aX*)–I would like to see more articles geared towards performing musicians and recording. Mr. Galo's review of the Behringer MX802A and MX2004A does not reflect the fact that the Mackie 1202VLZ Pro (and predecessor 1202VLZ) has four crucial features that are not available on the competing Behringer MX802.

First, the Mackie permits you to set input trim levels for each channel by pressing the SOLO button, and monitoring the LED meters that are switched to PFL (prefader level) to check the incoming level and thereby avoid overload of the input stage. The Behringer has no corresponding method, as far as I can ascertain by reviewing the MX802A manual at the Behringer website.

Second, the Mackie has a SOLO function, which the Behringer does not. The SOLO function is invaluable in locating a non-functioning or distorting input.

Third, the Mackie includes individual channel mute buttons, which allows you to mute channels when not needed. Muting eliminates the noise contribution of unneeded channels and reduces the possibility of feedback in sound reinforcement without needing to disturb the level pots, which were carefully set during the sound check.

Finally, the Mackie meters are calibrated to read specific decibel levels tied to clipping and standard 0 VU, and the level set pots show unity gain. By contrast, the Behringer meters do not appear to be calibrated and instead represent arbitrary levels. The lack of these features, in my view, disqualify the MX802 from serious consideration.

Mr. Galo states that the Mackie uses the 4560 IC op amp. However, in the June 2001 issue of *Electronic Musician*, the Mackie advertisement indicates that the VLZ-PRO uses a 2068 IC op amp—which is a much better op amp than the 4580 used on the Behringer (although not as good as perhaps some of the Analog Devices or Burr-Brown IC op amps currently in vogue with audiophiles). I wonder whether Mr. Galo is confusing the earlier VLZ model with the current VLZ-PRO (something that I notice sellers do all the time on eBay).

Finally, with respect to value, I recall reading on Audio Asylum and several other pro audio forums on the Internet that Behringer was the subject of a product infringement suit filed by Mackie, which was settled in Mackie's favor. I think that we all need to be aware that good engineering costs money, and that American quality of life is threatened by foreign firms who do not respect intellectual property (I leave it to users to decide the matter of American assembly jobs versus Chinese assembly jobs).

Furthermore, the downloadable manual for the Behringer mixers indicates that warranty service is available only by returning the mixer to Germany. This clearly reduces the cost for the manufacturer, but at what cost to the user? Mr. Galo's review suggests that Behringer warranty service is available in the US; if this is true, it speaks poorly of the accuracy of information available on the Behringer website.

For all the reasons previously mentioned, I am sticking with Mackie.

Geary Mizuno
Bethesda, Md.

Gary Galo responds:

Mr. Mizuno is correct regarding the additional features provided on the Mackie 1202VLZ Pro. However, I don't consider it fair to compare the Mackie to Behringer MX802A, since it is considerably more expensive (street price around $330 compared to $130 for the Behringer).

The Mackie is really for users who need the additional features of the 1202VLZ Pro, and who need the additional line inputs it provides (four stereo line inputs vs two for the Behringer). Contrary to Mr. Mizuno's assertion, Behringer does offer a method of adjusting the input trim levels on the MX802A, one that does not require the use of a SOLO button. This procedure is explained in sections 5.3 and 5.4 of the MX802A manual. I strongly disagree that the lack of the features Mr. Mizuno mentions disqualifies the MX802A. The MX802A offers remarkable flexibility for the price, and will fit the bill in many applications. If it doesn't, then the extra money for the Mackie will certainly be justified.

The mixer I am using at The Crane School of Music is the 1402VLZ-Pro. A peek inside shows that the Pro-version definitely contains 4560 op amps, but I also noticed a 2068 situated on the mike preamp board. Looking at the data sheets, the 2068 appears to be somewhat better than the 4580 in certain aspects of performance. It is slightly faster than the 4580 (6V/μs vs 5V/μs for the 4580), and somewhat lower in noise (0.56μV voltage noise vs 0.8μV for the 4580).

On the other hand, the 4580 has the lower distortion (0.0005% vs 0.001% for the 2068). Mackie appears to have switched to the 2068 to extract the lowest possible noise from their mike preamp topology. Note *that the noise specifications for the microphone inputs on the Behringer MX2004A and the Mackie 1402VLZ-Pro are identical; both are –129.5dBu with a source impedance of 150Ω.*

I am somewhat mystified by Mr. Mizuno's comments on Behringer's metering system. On my Behringer MX2004, a "zero" reading on the meters corresponds to a balanced output level of +4dB, exactly like my Mackie 1402VLZ-Pro. A +10 indication is, indeed, 10dB higher, and "Clip" indicates that the maximum output level of +28dBu has been exceeded. "0" on the Behringer input and output faders set the mixer for unity, just like the Mackie when it is set to "U." Both mixers are, in fact, within 1dB of unity gain with the faders at this setting. Calibration of the MX802A mixer's metering and input/output level controls is very similar, except that the MX802 has a maximum output level of +22dBu.

I was aware of the lawsuit between Mackie and Behringer. A press release on Mackie's website, dated Nov. 8, 1999, notes: "Mackie Designs Inc. and Behringer Spezielle StudioTechnick GmbH announced today that the Companies have resolved all pending legal disputes between the parties. Under the agreement, the terms are to remain confidential and therefore, there will be no further or additional comment. The parties acknowledge that they are pleased with the outcome of this settlement and look forward to participating in friendly competition in the future with their respective brands."

There are a lot of rumors floating around on the Internet regarding who actually won, but it is all pure speculation. At the time I wrote the Behringer review, this dispute had been settled for over a year, to the satisfaction of both parties, and I did not deem it pertinent to my review. From what I have been able to gather, the lawsuit was over one of Behringer's larger Eurodesk mixing consoles, and not the products I reviewed.

I, too, am concerned when American jobs are lost to overseas manufacturing. But this problem did not start with China. It has been going on for decades, involving many countries in the Far East, including Japan, Taiwan, Singapore, South Korea, Indonesia, and Malaysia. China is merely the latest competitor.

If you go to Behringer's home page and click on "Contact," you will note Behringer's instructions to contact your local distributor, or the headquarters in Germany. If you click on "distributor," you will be taken to a country list, which will then take you to the phone number for Behringer USA. If you call this number, you will be referred to a factory-authorized service center nearest you (which is what I did).

SILICONE

Phillip Milks expressed concern about the corrosive effects of silicone sealer in his letter on pg. 81 of the August '01 *aX*. The acetic acid he mentions is released from one of the components, methyltriacetoxysilane, on contact with moisture in the air during curing (G.E. Silicone II releases ammonia during cure). While silicones are gas-permeable and cannot "trap" these vapors, they can cause some corrosion during the curing phase.

After 24 hours, cured RTV is 100% chemically inert solid silicone. Builders use it to seal aluminum siding and flashing, without adverse effects. However, the silicones widely available at home centers should not be used on electronic equipment.

Special industrial sealants are available for application around electrical and electronic equipment. We used them extensively in contact with aluminum and even magnesium in the aerospace industry, and to encapsulate electronic circuit modules.

Chuck Hansen
cmhj@concentric.net

20W AMP

I've been a subscriber for a couple of years. Nice work! The new format is great as well.

I have a question about the 20W amp project in *GA* 5/00 ("A 20W $260 Amplifier," p. 20). I just finished the project, or so I thought. I can't seem to get rid of the hum. It's not window-breaking, but is quite noticeable. I tried moving grounds, adding grounds, removing grounds, different locations for the filter caps, and so on. What am I missing?

Has anyone else had this problem? The connection error on the 6L6s has been addressed.

Mark Fahler
Gaylord, Mich.

Joseph Norwood Still responds:

I'm very pleased that you constructed the 20W U.L. Integrated Amplifier, and when you eliminate your hum problems, I'm sure you'll be very pleased with this amplifier. I am very sorry you have a hum problem and will endeavor to provide a solution to your problem.

Since you don't mention that the hum is in both channels, the input circuit to the 6L6s on the schematic in the article is incorrect. The resistors R12 and R13 should tie directly to pin 5 of the 6L6 tube sockets—no other connections are required. Grid #3 of the 6L6 is tied internally to the cathode within the tube (Fig. 6).

As to possible remedies of the hum problem, I suggest you try the following:

1. Make sure the input of the 6L6 is wired as previously described.
2. Make sure the J1 to J4 RCA jacks are connected to rotary switch S1 via a four-conductor shielded cable, which must be grounded at both ends. Make sure the output of S1 is connected to R1 and R14 via a shielded cable (grounded at both ends). You should use shielded cable for all long leads going to the input of the 6SN7.
3. In the high-wattage rectifier, circuit D1 or D2 may not be functioning. I would replace D1 and D2 if you do not have a voltmeter that is capable of reading the AC ripple voltage. If you do, the AC ripple as stated in the article is approximately 0.6V RMS at 396V DC output when the DC supply is functioning properly.
4. Check the power-supply schematic and make sure it is wired correctly, with A, B, and C connected to the proper resistors. Make sure wires from A, B, and C of the power supply are connected to pointer A, B, and C of the amplifier.
5. Make sure the center-tap of the 6.3V heater winding is returned to ground.
6. Check coupling capacitors C2 and C3 for leakage, as well as bypass capacitors C6 and C7.
7. Interchange the input (center) 6SN7 with inverter tube (6SN7) to see whether hum disappears and eliminates the possibility that you may have a bad tube.
8. If you used a steel chassis instead of an aluminum chassis, make sure you scrape the paint down to bare metal wherever grounds are made in the amplifier, terminal strips, and so on.
9. The noise of the amplifier measured at the 8W taps is 3mV RMS and somewhat less at the 4W taps with a properly operating amplifier.

CAUTION: When making the checks, make certain the grid circuit and cathode circuit components are in place at all times; otherwise, 6L6 tube failure would occur.

Also note: The parts list on p. 22 of the article (Table 1) is corrected by transposing 4Ω and 8Ω for resistors R18A/B and R19A/B.

BAD LINK

Great article on Digital Impedance Measurement in the July '01 issue. The references on p. 47 may contain a bad link. Can you please check to see if the link for reference #8 (frequency-counter and sine-generator) is correct? I get no response from www.espanet.com/caribu/software.html. This looks like something really useful.

Dennis Yantz
Malabar, Fla.

Oliver H. McDaniel responds:

I would like to thank Mr. Yantz for his letter. He is correct concerning Reference 8. The link that I referenced no longer exists.

The signal generator contained in Wavetools (Ref. 3) works well as a sine-wave generator and can be substituted for the one in Reference 8. I have recently discovered that the frequency counter from Refer-

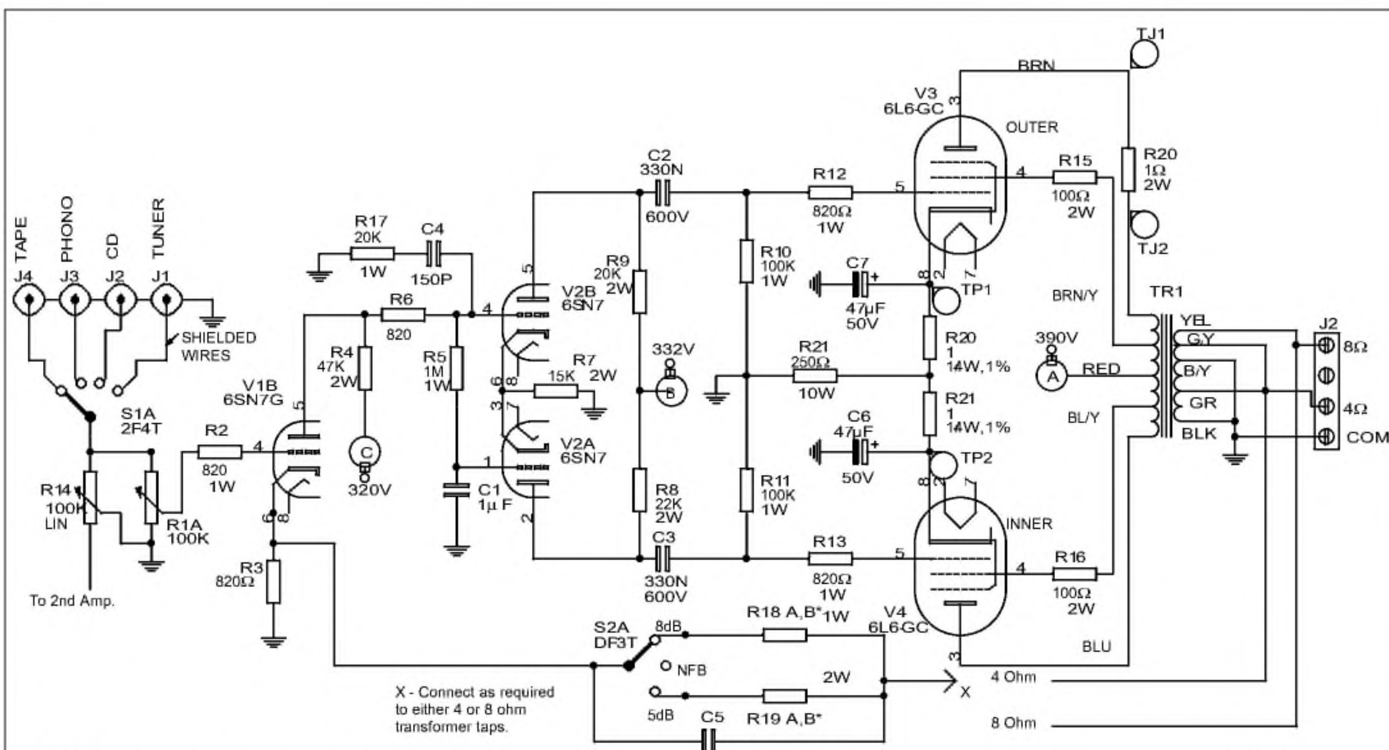


FIGURE 6: Corrected schematic from Still article.

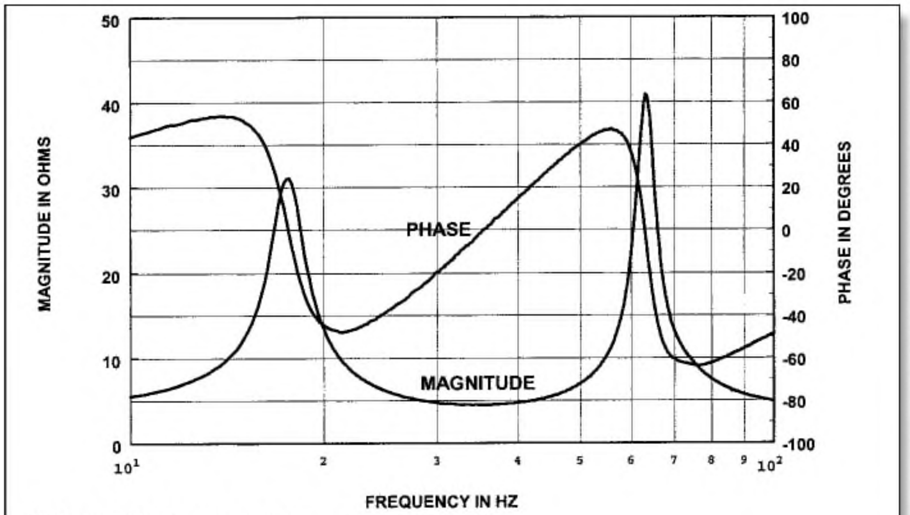


FIGURE 7: Woofer vented box measurement.

ence 8 will not work with PCI bus soundcards. As a substitute for that, the FFT analyzer in Wavetools has a built-in frequency counter accurate to 1Hz. The frequency is displayed in the lower-right corner of the analyzer window.

I have found two errata in the published article that should be obvious to most readers. Figure 5 appears again as Figure 8. The correct Figure 8 is included (Fig. 7).

The other error is in Equation 7, which should read:

$$F_B = (F_L^2 + F_H^2 - F_C^2)^{½}.$$

I apologize for not picking up these errors when I reviewed the final manuscript.

CLUB CONTACTS

I have read *Glass Audio* since 1997 and continue to enjoy *audioXpress*. A few years ago, my friends and I founded a club—Sofia Tube Audio Club (STAC).

Unfortunately, the economic situation in Bulgaria has worsened, so we disbanded. However, we would like to connect with other clubs, especially those interested in tubes.

Nikolay Panayotov
Sofia Tube Audio Club
Sofia, Bulgaria
npanayotov@hotmail.com

From time to time, audioXpress *publishes a list of audio clubs who have contacted us. See the Sept '01 issue for our latest list.—Eds.*

HELP WANTED

I would like to get as much possible bass as I can from my home stereo system. I own all Sony components. I have two older Hitachi speakers with two 12″ subs. They sound real good, but I want more. I bought two 15″ subs and want to build a box. What type of dimensions would be best for the result I would like?

Darcy and Janie DesRoches
Darcy.desroches@sympatico.ca

I'm designing a home that will have recessed wall speaker enclosures in drywall construction. Does anyone make a hinged aluminum frame covered with stretched, acoustically transparent, translucent fabric? I want to be able to place the rather large speakers into the openings in the wall close to the framed fabric doors behind them and turn on the backlights, causing the fabric to illuminate.

Mike McGee
mmike78@qwest.net

I am looking for plans to build a set of Klipsch horns.

Bruce Bishop
bbishop@wsipc.org

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

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New Chips on the Block

AKM AK5370 ADC

By Charles Hansen

AKM Semiconductor has introduced the AK5370 16-bit single-channel A/D converter (ADC) with built-in USB audio controller and microphone preamp. In addition to the high-performance ADC, all of the functions essential for a USB microphone—USB transceiver and programmable gain amplifier, controllable-from-the-host USB audio command processor, two endpoint FIFO for isochronous transfer, microphone bias control circuit—are realized on one chip. There is no need for an external ROM or a microprocessor, reducing the number of external components, lowering system cost, and reducing circuit area.

Other features include:
- Mute/volume control
- AGC pin for external control
- On-chip PLL
- Sampling frequencies support 8kHz, 11.025kHz, 22.05kHz, 44.1kHz, 48kHz
- Programmable gain amplifier: +20dB to −58dB
- ADC S/N 84dB at 44.1kHz
- Single power supply, low power: +3.3V ±10%

The AK5370 is packaged in a 24-pin VSOP, priced at $5.51 in 10k quantities.

An evaluation board is also available, featuring a USB B-type receptacle and microphone jack. Contact Richard Kulavik toll-free at 888-256-7364.

AKM Semiconductor, 2001 Gateway Pl., Suite 650W, San Jose, CA 95110, www.akm.com. ❖

Burr-Brown PCM1801 ADC

By Charles Hansen

The PCM1801 is a low cost, single chip stereo analog-to-digital converter with single-ended analog voltage inputs. It uses a delta-sigma modulator with 64× oversampling, a digital decimation filter, and a serial interface which supports slave mode operation and two data formats. The PCM1801 is suitable for a wide variety of cost-sensitive consumer applications where good performance is required.

The PCM1801 features a dual 16-bit monolithic 64× oversampling delta-sigma ADC and internal high-pass filter. Passband ripple is ±0.05dB and stopband attenuation is −65dB. THD+N performance is −88dB (typ), with 93dB SNR (typ) and 93dB (typ) dynamic range.

The PCM audio interface is left justified, I^2S. Sampling rates can be selected from 4kHz to 48kHz, using system clocks of 256fS, 384fS, or 512fS.

The PCM1801 uses a single +5V power supply in a small 14-lead SOIC package. Pricing for 1,000 pieces is $2.25 US.

Texas Instruments Inc., Burr-Brown Products, PO Box 11400, Tucson, AZ 85734, 800-548-6132, www.burr-brown.com ❖

ERRANT 300B BEHAVIOR

I have built Nobu Shishido's DHT-SE (*Glass Audio* 3/97) and Jack Elliano's Ultrapath, but with 76 tubes. I'm really content with the musical results, but I'm not only a music lover. I continually try to measure all the parameters (THD, harmonic analysis, and so on) to improve the performance of my DIY projects.

For the DHT-SE, I used Sylvania's 6SN7W, 6L6GA, and SV300B because I believe they provide better results than the 12AU7 and 6F6.

Now to my problem: because I have only two 300Bs, I cannot discover why one of the amps has a totally different behavior from the other. One of my 300Bs produced harmonic distortion under 0.1% (1W/8Ω), and this is not usual. This behavior is independent of the amplifier, so I think one of the two 300Bs is not behaving normally. I think it is perhaps nearly a perfect harmonic cancellation.

I have another question: Can I drive an SV811/10 from my Tango NC20 IT (A2-class)?

I have also built horn speakers with a Lowther DX-4 driver. To avoid the hum from AC-heating, I use DC for the 300B. But this isn't good (emission nonlinearity due to the different potential over the filament). I thought about ultrasonic heating (150–180kHz) for the 300B. What's your opinion?

Vaclav Skricka
Burgthann, Germany

Eric Barbour responds:

Your "problem" with 300Bs is more subtle than you realize. It is most likely that one tube is producing harmonic cancellation when used with its driver tube, with the plate characteristic of each tube being nearly opposite to that of the other, thus producing even-order harmonic cancellation (with, albeit, a probable increase in odd-order harmonics).

These effects are extremely difficult to model mathematically. I know researchers who have attempted to use SPICE modeling to find a solid example of harmonic cancellation—and been stymied by inadequate accuracy in existing, and even improved, SPICE models.

Electronic circuits in general can have some real-world variation, due to the variations in individual components. Tubes can vary ±20% or more in their plate characteristics, an effect that can "swamp out" minute variations in distortion (especially in no-feedback circuits, such as the typical 300B SE amplifier). Add this to variations in resistors, capacitors, transformers, and load impedances, and it becomes obvious that cancellation is difficult and elusive to achieve.

So far as I know, the only way to reliably achieve harmonic cancellation in an SE tube amp is to use very linear triodes to begin with, and then reduce the second harmonic by experimental trial and error. I have managed to achieve this with an SV572-10 amp I built some years ago, although I would not guarantee such effect with any given set of tubes. This is why I haven't published the circuit yet; perhaps in a future issue.

Yes, you can use your Tango interstage transformer to drive an SV811-10. It would be a good idea to buy some SV811-10s or SV572-10s as soon as you can, because Svetlana Electron Devices has closed down all U.S. marketing and distribution. The fate of Svetlana audio tubes is in limbo as I write this. Furthermore, the SV811s and SV572s were not made at Svetlana proper, but at Electronpribor Ryazan, a factory that is currently in bankruptcy.

There were thousands of those triodes in the Alabama warehouse at last word, although their eventual fate is unknown. They never became popular with OEMs, in spite of years of determined marketing. I am told that PM Components in the UK will become the distributor for Svetlana products, although this cannot be verified yet.

RF filament heating is a very obscure and arcane subject—rarely done in real-world equipment. John Atwood wrote an article about his experiments in RF heating of triode filaments for issue 15 of Vacuum Tube Valley. *This is the only such recent article that I know of. The great complexity of the resulting circuit might put you off the notion. DC heating is a good compromise, and is much more reliable and less costly, so I would stick with it.*

GUITAR AMPS

Could you tell me how a low-power-rated guitar amplifier works? I know that it has a preamplifier, power amplifier, then speakers, but I don't really know how they each work with one another, and how the signal from the guitar pickups is changed to suit the guitarists.

Thanks very much.

Scott Thomson
thomson@jwpt.freeserve.co.uk

Charles Hansen responds:

All guitar amplifiers are similar, with power rating being only one of the variables. There are three main types of pickups: magnetic, piezoelectric, and microphone, each with distinct signal characteristics.

A magnetic guitar pickup has coils of fine wire wrapped around a permanent magnet, and has an inductive impedance. The output of this type of pickup decreases at high frequencies, so the preamp portion of the amplifier is designed to boost the highs. The power-amplifier section is not unlike a music amplifier, except the frequency range need not extend to 20kHz, and higher distortion is desirable in some applications.

The loudspeaker in a guitar amplifier has a limited range (about 5kHz max) and must be very rugged. A complete guitar amplifier can have two or more inputs, tone controls (bass, treble, sometimes midrange), and often has additional "effects" such as reverb, tremolo, and distortion controls. Amplifiers for rock music tend to have different features than those used for jazz or acoustic music.

Piezoelectric and microphone pickups tend to be used in acoustic guitars. Piezo pickups are capacitive in nature and their output favors the high frequencies, so a guitar preamplifier is designed accordingly. Microphones usually have balanced outputs, and require an additional gain stage due to their lower output.

Since acoustic guitars are more prone to feedback, the preamp usually has a tunable notch filter in addition to tone controls, and lower overall distortion. Acoustic power amplifiers have wider frequency range and lower distortion, and may have a tweeter in addition to the main driver.

As far as power output, this is up to the individual guitarist, and is determined by the size of the venue in which he or she plays, and the volume desired. Tube amplifiers are very popular, and are preferred for some guitar styles. Some of the low power tube amplifiers (such as the VOX AC-4, and the Fender Champ, Harvard and Princeton) are single-ended Class-A, and are highly prized.

PUZZLE SOLUTION

I've just purchased a pair of Enigma minimonitors. After listening to them, and on the advice of a friend, I think they are very well suited to my Audio Innovations 25W tube amp. At the end of the review in your March 2001 issue ("Testing Enigma Acoustique's Oremus Loudspeaker," p. 57), mention is made that this speaker is well-suited to someone who likes to tweak his or her system, because of its lack of midrange. Do you have any ideas on how to bring out or boost the midrange so it's more pleasing for vocals? I'd appreciate any ideas you have on the matter, since I'm fond of the speakers and I purchased them used at a decent price.

John Alvarez
jdalvarez2@home.com

Joe D'Appolito responds:

The remark Mr. Alvarez attributes to me was actually made by Dennis Colin. That said, the Enigma has a broad 5dB response dip centered on 3kHz. The Enigma drivers are relatively high quality. My approach would be to redesign the crossover to fill in the dip. Lacking this, you could try boosting this area with a parametric equalizer.

The contact information for the company is 716 St. Rose Ave., Saint Rose, LA 70087. ❖

Book Review

5.1 Surround Sound Up and Running

Reviewed by Mark Florian

5.1 Surround Sound, Up and Running by Tomlinson Holman; $29.95; Old Colony Sound Laboratory, PO Box 876, Peterborough, NH 03458-0876, 603-924-9464, Fax 603-924-9467, custserv@ audioXpress.com.

Tomlinson Holman has had quite an aural effect on all of us who enjoy the movies, both at the local theater and at home. His THX (Tomlinson Holman's eXperiment) standard has drastically changed the sound that you hear at the movies for the better. Dolby Digital or AC-3 or 5.1 has become the new standard for home viewing with DVD players.

Mr. Holman covers a lot of ground within this book's 273 pages. Its purpose, he says, "is to inform recording engineers, producers, and others interested in the topic about the specifics of multichannel sound." Note that multichannel sound covers mixes for both film soundtracks and audio-only formats, as in DVD-Audio.

Main chapter headings are Introduction, Monitoring, Multichannel Microphone Technique, Multichannel Mixing, Delivery Formats, and Psychoacoustics. Each of these headings then discusses in detail a large number of sub-topics. For instance, the chapter on monitoring covers 33 sub-topics! And that's just one chapter. Space limitations prevent me from going into much detail in describing all of the topics covered.

These chapters are followed by three appendices at the end of the book: Sample Rate, Word Length and Surround Resources, and an index. Fortunately, each chapter begins with a "Tips from this chapter" summary, so I'll cover the high points and then offer a summary at the end.

CHAPTER BREAKDOWN

The 12-page introduction begins with a brief history of multichannel sound. From 1938–41, Disney studios developed Fantasound for its new movie *Fantasia*. The format used a total of five channels: three in the front and two surrounds located in the back corners. Sound familiar? The Disney system, though, used only three source channels on the film that were steered between the five loudspeaker channels.

Chapter Two on monitoring discusses the needs of standardization for level, frequency response, and the amount of direct to reflected sound energy; the importance of bass management in the 0.1 channel; control room acoustics and why live-end/dead-end arrangements are not useful; key factors in choosing monitor loudspeakers; pros and cons of direct radiator surrounds and surround arrays or dipole radiators; the ITU-775 standard for monitor placement around the console; how you can use electronic time delay devices to synchronize loudspeakers; the purpose of the low-frequency enhancement channel and how to properly mix it; and the importance of calibrating all monitor systems for level.

Chapter Three, Multichannel Microphone Technique, starts off with an analysis of the four basic stereo microphone techniques: pan pot stereo, spaced omnis, coincident or near-coincident (X-Y, M-S, ORTF, Faulkner, Ambisonics, and so on), and dummy head binaural. It then examines how these techniques are modified for multichannel recording use; the two different points of view in multichannel stereo: direct/ambient versus sources-all-around; how to enhance spot miking with digital time delay; how to record for both 2 and 5.1 channel releases on one 8-track machine; surround microphone recording techniques; and an evaluation of special 5.1 channel microphone systems from Brauner, Holophone, Sanken, Schoeps, and SoundField.

Chapter Four, Multichannel Mixing, discusses how mixing for 5.1 is actually easier than for stereo and why; differences between spaciousness and envelopment in the sound field; assigning microphone channels to loudspeaker channels; fixed panning versus dynamic panning; the three types of panners in common use and how they work; panning laws and the use of divergence and focus controls; standard versus nonstandard panning; and using complementary comb filters, decorrelation, and reverberation to change the size of a source. It then examines equalizing multichannel recordings and how it varies from stereo; routing multichannel in the studio to minimize the effects of jitter; multichannel track layouts with

ABOUT THE AUTHOR
Mark Florian received a BS in Electrical Engineering from the University of Texas in 1983 and has been involved in electronics since disassembling a TV set at the age of seven. He grew up singing in choirs and has played various musical instruments, such as the trombone, guitar, banjo, and hand bells, and is currently learning the piano. Mark is owner of West End Audio, a residential A/V design and installation company.

ITU and SMPTE standards; fitting multichannel mixes on digital video recorders; and multichannel monitoring electronics and outboard gear.

Chapter Five, Delivery Formats, begins with definitions of new terms: data essence, metadata, and wrappers, followed by thorough discussions of Linear Pulse Code Modulation (LPCM), LPCM with MLP (Meridian Lossless Packing), and 1-bit delta-sigma conversion; bit-rate reduction ratios and low bit-rate perceptual coders (Dolby AC-3, DTS, and MPEG AAC); word length and sample rate considerations for recorders and workstations; dynamic range versus effective number of bits; and ITU/SMPTE standardized track layouts. It also discusses the multiple audio stream capability of DTV and DVD-V; differences in metadata content between DTV and DVD-V; the three level-setting mechanisms: Dialogue Normalization (Dialnorm), Dynamic Range Compression (DRC), and Mixlevel; and debugging sync problems between audio and video for DVD-V and DTV systems. The chapter includes a thorough discussion for various media types: laser disc, DTS CD, DTV, DVD-V, DVD-A, and Super Audio CD (SACD); and intellectual property protection schemes, including watermarking. Tables presented in this chapter include: capacity of various release media, typical audio bit rates for AC-3, DVD types and their capacity, comparison of LPCM, AC-3, MPEG-2, and DTS audio coding methods for DVD-V; and playing time and bit rate for LPCM versus LPCM with MLP at various sample rates.

Chapter Six covers psychoacoustics and the areas of principal localization mechanisms: Interaural Time Difference (ITD), Interaural Level Difference (ILD), pinna effects and Head-Related Transfer Function (HRTF); Minimum Audible Angle (MAA); bass management and LFE psychoacoustics; effects of the localization mechanisms on 5.1 channel sound; the law of the first wavefront; Phantom Image Stereo; Phantom Imaging in Quad; localization, spaciousness, and envelopment; lessons from concert hall acoustics; rendering five channels over two; auralization and auditory virtual reality; and beyond 5.1, looking at 10.2.

APPENDICES

Appendix 1 is a thorough discussion of sample rates and how they affect audible quality. This is followed by an analysis of aliasing, and multi-bit and one-bit conversion processes, including converter tests.

Appendix 2 addresses word length, aka bit depth, resolution, and many other aliases and the trade-offs between them. Additional areas covered are: quantization distortion, dither, dynamic range, actual converter performance (real bits versus "marketing" bits), what level of performance is really needed, noise shaping, and analog reference levels related to digital recording. Tables in this appendix include: LPCM dynamic range versus number of bits, LPCM s/n ratio versus number of bits, and dynamic ranges for a number of professional DACs and ADCs.

Appendix 3, Surround Resources, contains contact information for setup and test materials, microphone manufacturers, small and large format console manufacturers, systems for live sound, hardware and software for processing multichannel information on digital audio workstations, 3D sound production tool manufacturers, monitoring switching systems, volume controls and controllers manufacturers, outboard equipment, multichannel meters, format encoders and decoders, hardware and software, headphones for 5.1, multichannel power amplifiers, upmixing facilities, and professional organizations.

As you can see, this book contains a lot of information. I found it very practical, in-depth, and exhaustive in its coverage. I highly recommend it to anyone wishing to learn the details of mixing multichannel sound.

This book would also be of interest to those in the business of installing home-theater systems in residential houses as a reference for their employees. By better understanding the details of a multichannel mix, you can better understand how to set up the reproduction system for your clients. ❖

Classifieds

VENDORS

The **BassZilla** has landed! Finally, a high-efficiency and affordable loudspeaker kit by Dick Olsher. Kit plans at: **http://www.blackdahlia.com**.

BILLINGTON
EXPORT LIMITED

All types of audio tubes. 300B 6DJ8 ECC81 ECC83 KT88 Mullard GEC Sylvania. Discount for large quantity. Minimum order £150 Export (£100UK) Billington UK. Tel (0)1403 784961. Fax (0)1403 783519. Email/website www.bel-tubes.co.uk

The LP track carries both stereo channels vs. alternating pits on the CD. The LP remains superior with "tight coupling for audio," $15; using an "OTL amplifier circuit," $35; driving efficient "Bass Horn" plans, $50. Visa, MC. ***STEEN AUDIO LAB***, PO Box 2185, Vancouver, BC V6B 3V7, CANADA.

Electrostatic Loudspeaker Components & Information

Parts for the DIY ESL Enthusiast
From: Barry Waldron's
ESL Information eXchange
1847 Country Club Drive
Placerville, CA *95667*
(530) 622-1539
Website: www.jps.net/eslinfo.

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Michael Percy, Box 526, Inverness, CA 94937

AD INDEX

WANTED

TUBE HIFI, HORN SPEAKERS, and HIGH END. Altec, Marantz, JBL, Mcintosh, Electrovoice, Fisher, Scott, Dynaco, Leak, Eico, Audio Research, CJ, Quad, Bedini, etc. Sonny, Phone: 850-314-0321 Fax: 850-314-0284 **sonnysound@aol.com**.

Yard Sale

FOR SALE

Scan-Speak matched pair drivers—18W/8545 6½″ mid-woof, $160/pr; D2905/9000 Revelator tweeter, $200/pr. Audax-Vance Dickason A652 MTM kit drivers, x-over, and plans, $235. All with low hours. John (503) 842-5160.

Dozens of back issue audio magazines including *Glass Audio*, *Amateur Audio*, *Speaker Builder*, *Stereophile*, and *FI*. Also have hundreds of misc. parts for tube amps and other projects. E-mail for complete list roymottram@aol.com.

JBL titanium tweeters 044TI, $50 ea; JBL 104H midrange (3), $35 ea; JBL 108H midbass (1), $50; Audax tweeters AWO25S3 (3), $15 ea; Audax midrange HM100X2-2 (2), $25; Adcom GFA535 60/60, $165. All items in perfect condition. You pay shipping. Ernie Holmes, 7616 Away Way, Citrus Hts., CA 95610, 916-726-7039, ernieh@lanset.com.

(4) Scan-Speak D3806, $80 pair or $150 for all four. Vifa MW18WN-19-04 cast frame car midwoofers, $70 pair. Morel MW114-S 4Ω, $95 pair. Morel MW166 4Ω, $90 pair. Jerry, (601) 271-8193—leave message if no answer.

Dynaudio 17w75x1 (2), $70 ea; Dynaudio D28/2 (2) $50 ea. All are 8Ω and in excellent condition, with less than 50 hours on each. Nick 559-583-6511 or jet12@pacbell.net.

Complete set of *Speaker Builder*, 1980-2000, $250 plus shipping. If you are interested, e-mail me at: giorgiocasini@globecast.it.

Altec: 290 Alnico drivers, $375/pair; 808 Alnico drivers, $250/pair; 811 horns, $175/pair; Stephens 12″ woofer, $75; Gauss 1″ driver, $100; JBL 2416H drivers, $225/pair; Harman Kardon CD91 cassette, $75; Marantz 3020 cassette, $75; AR10(Pi) speakers, 60 lbs each, $475/pair, extra mid $60. David (845) 688-5024.

WANTED

Weathers Modernaire bass module and crossover or complete with satellites. Sprague, PO Box 422, Allendale, NJ 07401-0422. (This is a late '50s three-way speaker system.)

Output transformer for an Eico HF-81 or similar. Eico part #32013. Pete Rogosky PCBHM@epix.net.

Two Focal 7N 402 DB drivers. Preferably new, but used fine. Ruud—Phone/FAX 415-383-4958, e-mail Ruud66@aol.com.

Searching for unique tube 2-way audio crossover used ahead of preamps for electric guitar—need a schematic. Need tube output transformer for Sansui 1000 A receiver. N. O'Shana, 101 Treble Rd., Bristol, CT 06010.

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Vintage Glass

Bell Sound 2145

I make no claim to understand how this unit, which is from a well-known manufacturer, works. An interstage transformer feeding cathode-follower 6SN7s? The amp uses push-pull 6B4Gs, which should please the directly heated triode crowd. ❖

Lance Cochrane
San Francisco, Calif.

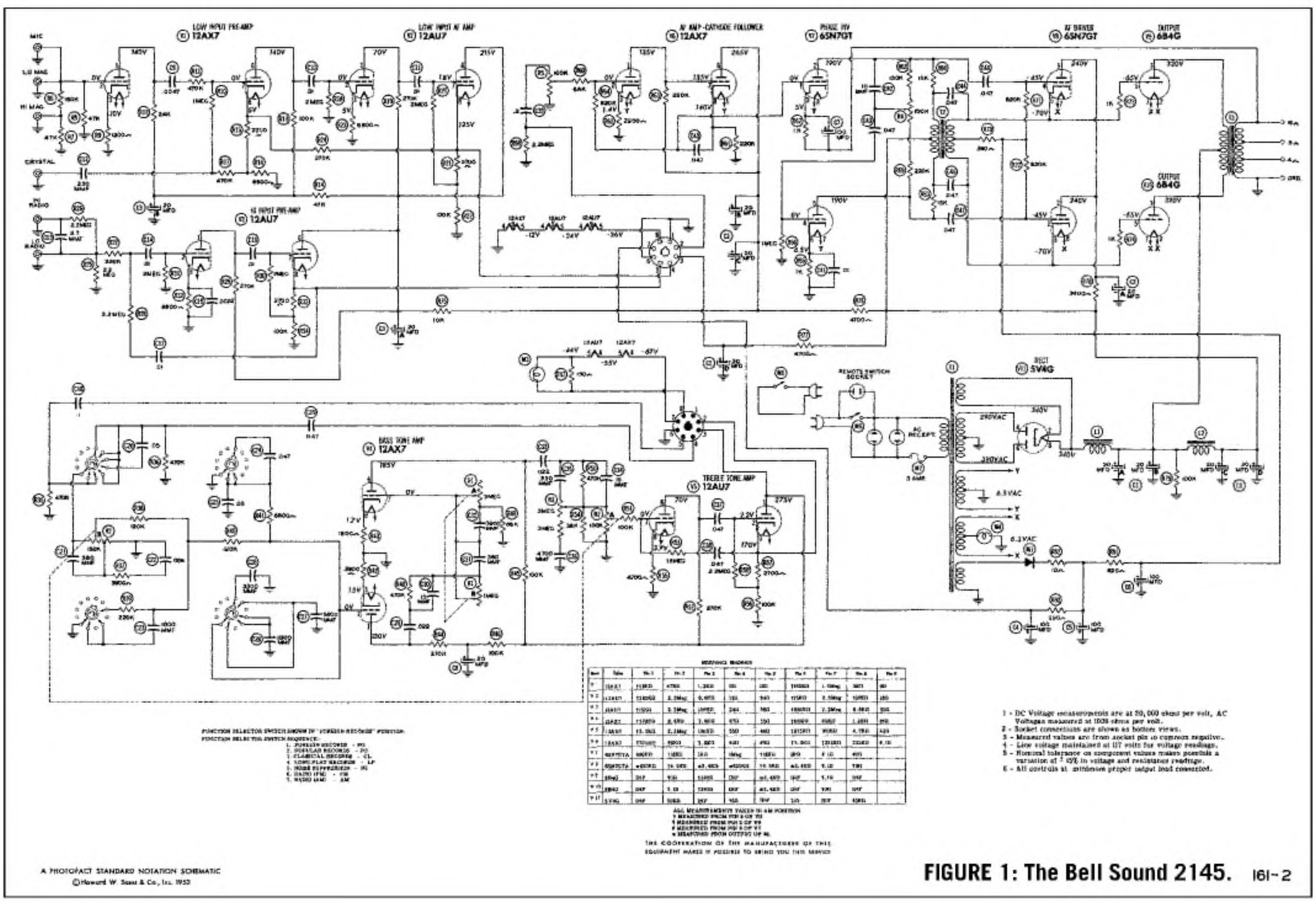

FIGURE 1: The Bell Sound 2145. 161-2