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The DR280

By Bill Fitzmaurice

This latest addition to the DR lineup lies between the DR250 (Jan. '04 *aX*) and DR300 in both size and performance, with more power and lower frequency extension than the 250, but small and light enough for one person to carry.

DR280 is a two-way system composed of a woofer, loaded by a folded horn, and a vertical tweeter array (**Photo 1**). The folding of the woofer horn allows a much longer pathway, and thus lower cutoff frequency (fc) than otherwise would be possible from the same cabinet depth. Its vertically arrayed multiple tweeters give higher sensitivity and better directional pattern control than a single tweeter. The cutaway side view (**Figure 1**) shows the positioning of the drivers.

You can use a wide variety of tweeter horns—piezo or dynamic. For simplicity you may use standard tweeters right out of the box, but multiple elements on multiple throats terminating in a single mouth give the utmost in performance. Variations on this theme are used in top-of-the-line $3k plus line array cabs from EAW.EV, JBL, Meyer, and many others. There is simply no better way to get high-power high-frequency output, and while commercial multi-throat single-mouth tweeters are very rare and quite expensive, you can make one for about ten dollars.

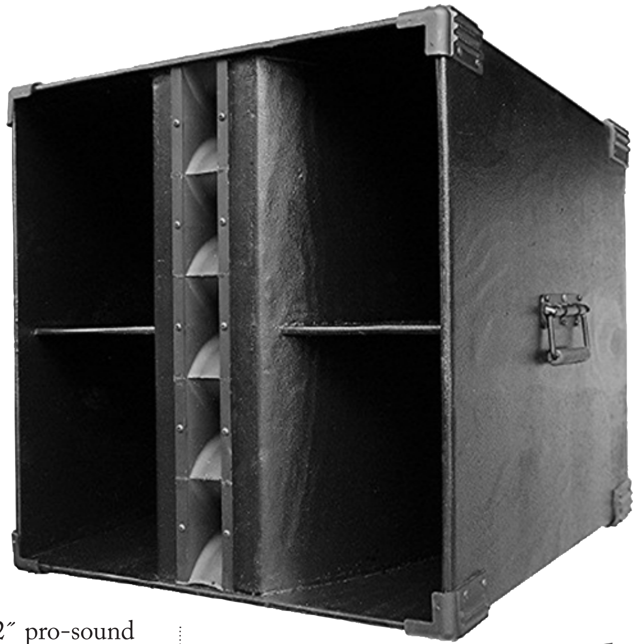
FEATURES

Smaller and lighter (70 lbs) than any commercial cabinet of similar capabilities, the DR280 has an 8° vertical splay angle built in. The lower boxes in a flown array may be fired downward, while a single box sitting onstage in backline duty will aim upward, so that the player can hear the highs. This is a critical feature, because vertical tweeter arrays have a very narrow vertical dispersion angle, while the splay of the box eliminates parallel walls in the woofer horn that would cause rough response. Rounded pathways through the horn bend allow it to work at least two octaves higher than folded horns built with flat panels, and these curved panels resist vibration better than flat panels triple their weight.

The recommended woofer is the Eminence Delta Pro 12. Pertinent T/S specs are fs 51Hz, SPL 98dB, Qts .35, Vas 82L, Xmax 3.2mm, and Pe 400W. You may substitute other 10 or 12″ pro-sound drivers with similar specs.

When shopping for drivers, consider seriously those with neodymium magnets, which are less than half the weight of ferrite magnet drivers, and also have more compact frames. The tweeters of the prototype are generic piezo horns sold under a variety of brand names, all using the 1025 model number. You can get drivers, as well as the rest of the cabinet hardware, from a variety of sources, including www.partsexpress.com and www.bltsound.com.

The SPL chart (**Fig. 2**) shows the response of the prototype cabinet, with tweeters and woofer only. Compared to the same driver in a bass reflex cabinet, the DR280 delivers 8dB higher average sensitivity from 100Hz to 1kHz. The 105dB/1W/1m



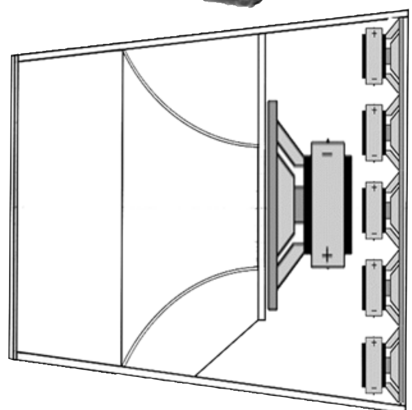


PHOTO 1 (top): The DR280.
FIGURE 1: (middle) Cutaway side view.

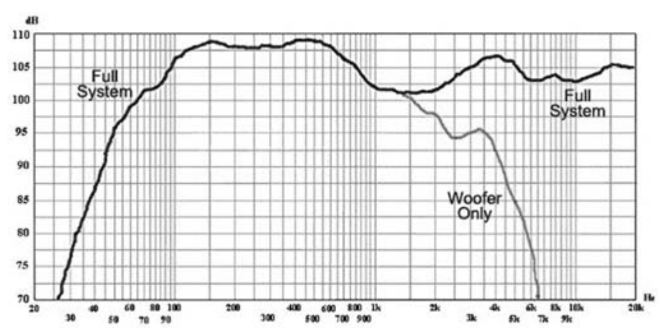


FIGURE 2: SPL 1W/1m half-space.

average SPL from 90 to 16kHz (±4dB) is a spec that very few cabinets at any price can match. The −6dB horizontal dispersion angle is 90°, suitable for most applications. With vertical stacking of more cabinets the horizontal dispersion remains unchanged, while the vertical dispersion is further reduced, for control of early reflections in difficult acoustic environments.

The cabinet is constructed primarily from ½″ plywood. You may use softwood if it has at least five plies. Baltic birch is a popular option, but be sure your birch is actually Baltic, i.e., Russian, Finnish, and so on. Northern birch from the US and Canada is also good. [Some large retailers sell "birch" plywood sourced from Asia that isn't actually birch. This product has a thin veneer outer layer that easily chips, and uses inferior glue that leads to delaminating. Quality Baltic birch has equal ply thicknesses and uses marine-grade glue. Baltic is usually found in 5 × 5′ sheets; you'll need one sheet per DR280.]

The curved throat horn sheaths and back halves are made from ⅛″ birch, while the curved mouth horn sheaths are ¼″ spruce or pine. Use plywood with three equal thickness plies, because plywood with two thin plies over one thick ply (commonly seen in lauan), or more than three plies as found in birch, doesn't bend well. A single sheet of each size will be more than sufficient for a pair of DR280s.

Bond all joints with adhesive and drywall screws, piloted and countersunk. Be sure to deeply countersink screws along the cabinet edges so that you won't hit the heads when rounding the edges over. Use 1.25″ screws jointing ½″ parts, 1″ screws with the thinner materials. Because the joints must be absolutely airtight, I highly recommend using caulking-gun-applied polyurethane base construction adhesive, such as PL Pro, which expands as it cures to fill gaps and makes caulking joints unnecessary. You also will need a hot-melt glue gun and a couple of feet of 4″ schedule 40 PVC water pipe, which measures close to 4.5″ outside diameter.

If you're going to build more than one of these, make tracing patterns of all the parts, because laying parts out is one of the more time-consuming aspects of the project. However, you need to figure out how to build it with the first copy before going into an "assembly line" mode to crank out more. I suggest that you read the instructions and look at the photos three or four times before starting the job. If you have any qualms at all, log onto my forum at www.audioroundtable.com/BillFitzmaurice/, where you can chat with me and other builders who'll guide you through the process.

CONSTRUCTION

Note that all the measurements provided are somewhat approximate. The actual thickness of the materials used and the accuracy of your construction influence sizes of the finished parts. This isn't a problem, because the design is very forgiving of minor errors. However, you do not want to attempt cutting out all the parts ahead of time, because a very small size deviation in the early stages of construction can be magnified into a fairly large error by the end.

Cut out each part as you need it, trim-

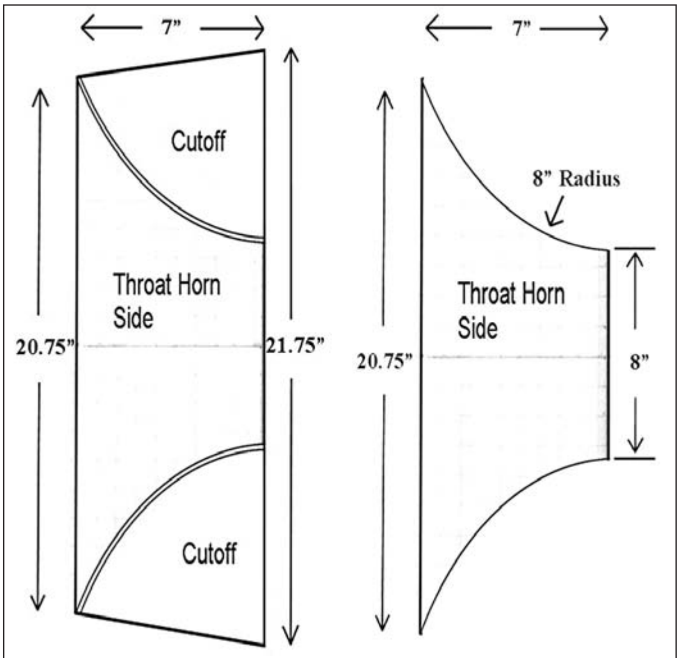


FIGURE 3: Throat horn side template.

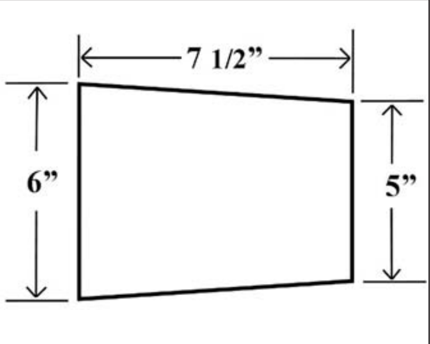


FIGURE 4: Throat horn divider.

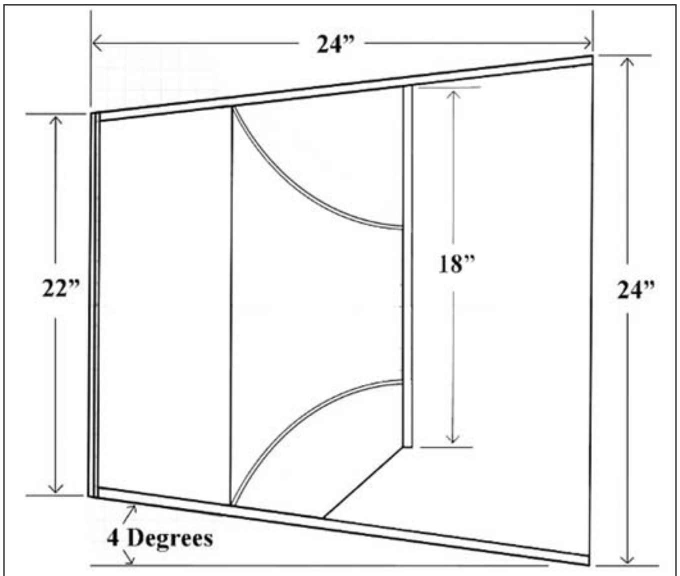


FIGURE 5: Side dimensions.

ming as required to actual finished size. When it fits properly, save it as a pattern to duplicate another. Always measure directly on the cabinet where each piece is to be installed to verify the true size required. When cutting parts from a full sheet of plywood, make them oversize and then trim them to final size on a tablesaw, preferably using a panel cutting jig for accuracy and safety.

The angle at which parts edges are cut, where applicable, will determine the finished cab size. The nominal cut angle for most of the angled parts is 4°, but a tolerance of ±1° is OK. While the nominal finished exterior height of the front of the cab is 24″, yours could end up a half inch shorter or higher. The same applies for the rear height. No worries, response won't be affected.

Start by cutting out the two throat horn sides and the throat horn divider. **Figures 3** and **4** show the blank that the throat horn side is cut from and the throat horn divider. **Figure 5** shows the side view dimensions of the entire cabinet.

To trace radii make a compass out of a strip of plywood, with a screw point or nail at one end, and a hole at the radius distance from that through which a pencil is inserted to draw the line. Trace a radius from each end of the part—where the two radii overlap is where you place the compass point to trace the final curve. That junction may fall off the piece of wood you're cutting, so have another piece of plywood handy for the compass point to rest upon (**Photo 2**).

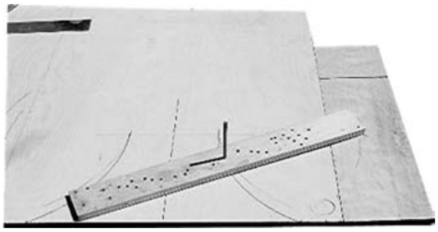


PHOTO 2: Drawing radii.

When you cut out the throat horn sides, you'll end up with four semi-triangular cutoffs; save two of them. Attach the throat horn sides to the divider, with the extra ½″ of material protruding on the driver end. The narrow end of the divider goes toward the driver. Offset the divider ¼″, so that one edge is on the centerline of the throat horn sides. This way the two halves of the horn are not quite exactly the same size, making for better response.

Temporarily screw the assembly to a piece of scrap plywood, squaring the sides, to stabilize the assembly (**Photo 3**). Cut the throat horn sheaths from ⅛″ plywood. Flex the plywood first to determine the easier-bending axis. The sheaths are cut oversize at 8″ wide by 11″ long, to be trimmed to final size later.

PHOTO 3: Throat horn on assembly jig.

Attach one of the triangular cutoffs to one of the sheaths, down the middle, with the driver end of the sheath flush to the end of the cutoff (**Photo 4**). Attach this to the throat horn assembly, with the sheath flush at the driver end, hanging over on the other. Repeat the process for the other side. To avoid splitting the wood, don’t put screws within 2″ of the end of the parts. There will be gaps at the ends of the joints; use long clamps to hold those closed until the adhesive has set (**Photo 5**), after which you can remove the assembly from the jig.

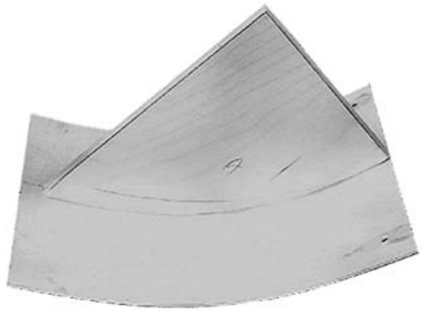


PHOTO 4: Throat horn sheath/brace.

PHOTO 5: The throat horn, clamped.

Trim and sand the sheaths flush to the sides, leaving the extra sheathing hanging over the ends. Cut out the throat horn supports, oversize at 22″ by 4″, with one long side at a 4° angle. Saw or drill holes in the supports, 3″ or so diameter. These holes allow the passage of air within the cabinet, lighten the assembly, and offer a handy spot to use a C clamp for assembly if you don’t have any pipe clamps (**Photo 6**). Always clamp parts together when possible to hold them in alignment while you drive the screws.

Attach the horn supports to the throat horn, the 4° angled side to the horn. Make sure the angles face the proper direction so that the resulting assembly is flat. You need to drive the screws from the inside of the horn, which is easy if you have a right angle or short shaft screwdriver.

While the adhesive is setting up, cut out the top and bottom, according to **Fig. 6**. When you cut the excess from the rear, save the semi-circle scraps, which you’ll need later. Draw all parts intersections on the top and bottom (**Fig. 7**). Onto the bottom draw and then cut out the driver access porthole per **Fig. 8**.

When the adhesive has set, trim the throat horn assembly to final size using a tablesaw, preferably with a panel-

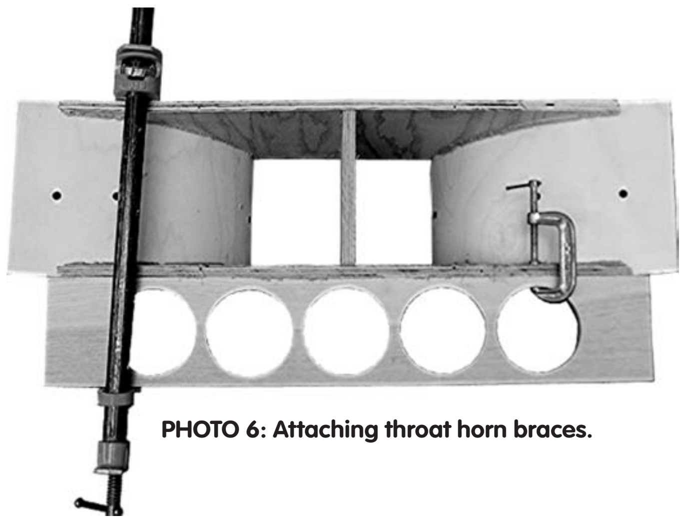


PHOTO 6: Attaching throat horn braces.

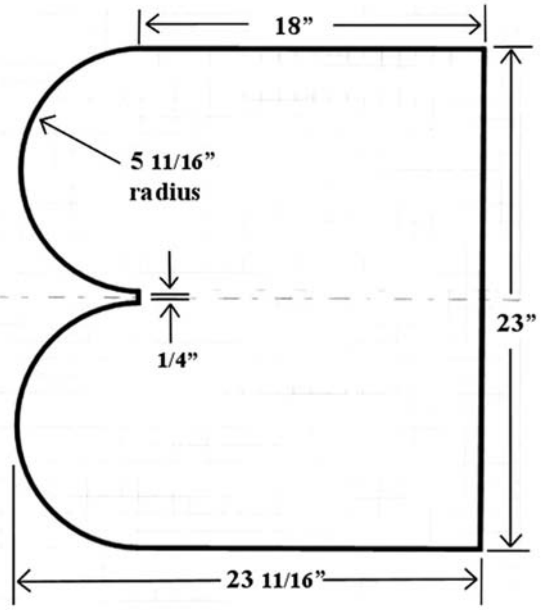


FIGURE 6: Top/bottom template.

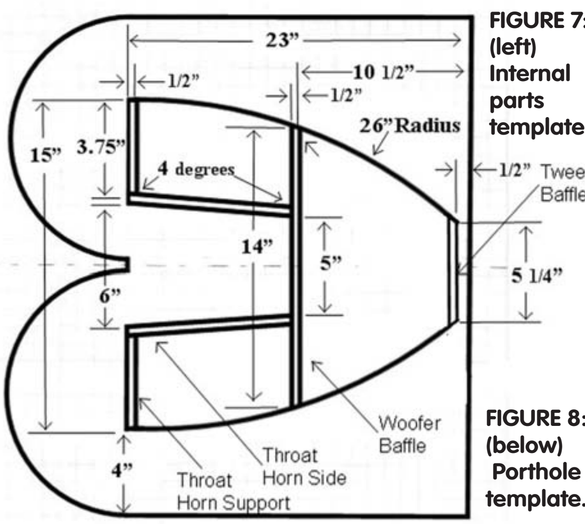


FIGURE 7: (left) Internal parts template.

FIGURE 8: (below) Porthole template.

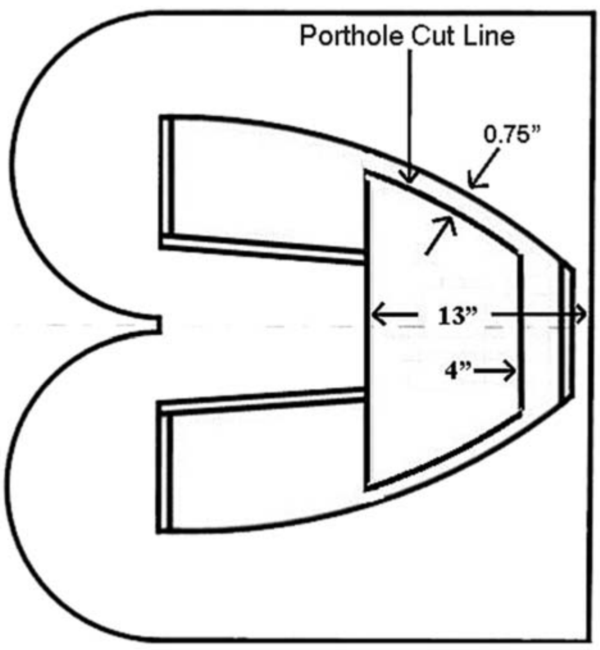


PHOTO 7: Throat horn before and after trimming.

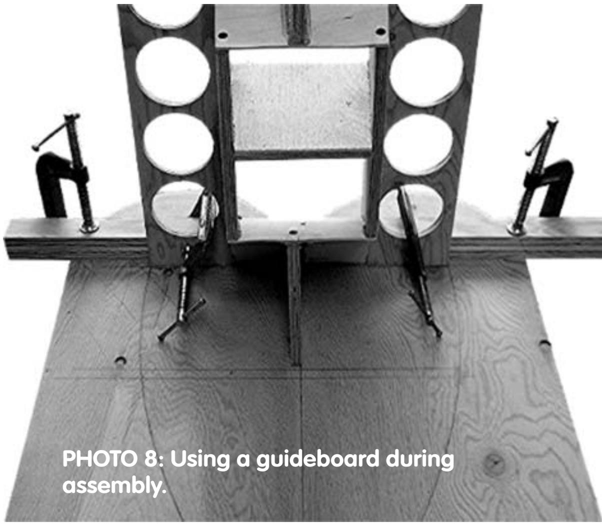


PHOTO 8: Using a guideboard during assembly.

cutting jig, which will square the assembly as you trim it. The edges of the supports are cut square, the top and bottom of the assembly at a 4° angle to match the taper of the box. Be sure the angle faces the right way.

Don't try to cut the height in one pass; ease it down cutting equally from both ends until you get it to about 20.75″. Trim/sand the cutoff braces as required to match the 4° box taper; cut about a 3″ triangle off the rear of one. **Photo 7** shows the trimming process atop a panel-cutting jig, and the before and after results.

Place a straightedge across the throat and braces where the baffle will be going, sanding the braces as required to true the assembly to flat. Attach the assembly to the cabinet top. This, and all jointing, is best done by pre-drilling pilot holes for the screws, clamping a straight 2 × 2″ guide board (made from a few layers of plywood) to the joint line, applying adhesive to the joint line, and then clamping the mating part to the guideboard (**Photo 8**).

Finish the job by drilling a pilot/countersink through the pre-drilled holes and drive screws, at least two per joint. To account for where the parts are at a 4° angle, cut one 4° edge on your guideboard. Also be sure to wipe any adhesive from the guideboard as soon as you remove it. Note that the untrimmed cutoff brace end of the assembly attaches to the top, the trimmed end to the bottom.

The woofer baffle (**Fig. 9**) is very tricky, so pay close attention to what you're doing. The rough dimensions are 18″ high by 14″ wide. The top edge is cut at a 4° angle, the sides at a 12° angle, the angles facing opposite directions. Refer to the diagrams to be sure of getting the angles right.

Cut a 5″ wide × 8″ high throat hole. Its center is about 11″ from the cabinet top; measure the assembly to confirm the

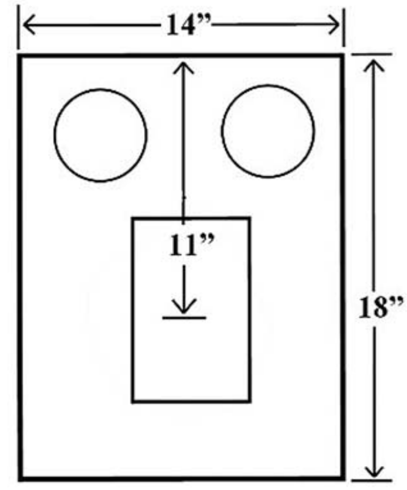


FIGURE 9: Woofer baffle.

exact distance. You also will need to add a ⅜″ or ½″ plywood spacer to prevent cone slap in long excursions. The hole in it is 11″ in diameter, while the outer edge is cut into a 12.5″ hexagon. For a 10″ driver make the hole 9″ in diameter. Using the woofer to determine their locations, drill holes through both the spacer and the baffle for T-nuts.

As large as this cabinet is, there's still not much room in the driver chamber, so I highly recommend that you use 3/16″ × 1.5″ Allen socket-head mounting bolts that you can drive by feel. Don't locate bolts at the baffle edges. Check the baffle edge angles to be sure that the wider side faces the throat horn and the narrower side the driver before installing the T-nuts or spacer. Drill or saw a couple of 4″ holes in the baffle to minimize internal reflections.

Install the T-nuts on the side of the baffle opposite the woofer and glue the spacer in place, holding it there with attaching bolts (**Photo 9**). Attach the baffle to the assembly, using a T-square to check the alignment. No screws are possible at the joints with the sheaths, so use plenty of adhesive there, as well as where the sheaths joint the top and bottom. Add the bottom to the assembly (**Photo 10**).

TWEETER ASSEMBLY

You may use dynamic horn tweeters if you wish, but they will add greatly to the speaker cost, have the added complexity of a crossover, and, despite what you may have heard, don't work any better than piezos. The best result occurs with tweeters whose frames have been

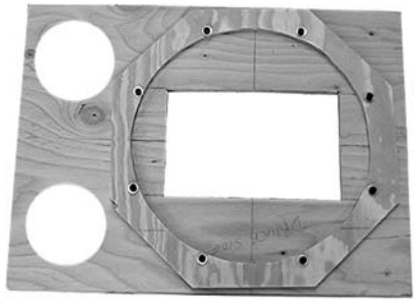


PHOTO 9: The woofer baffle.


PHOTO 10: Top, bottom, and throat horn.

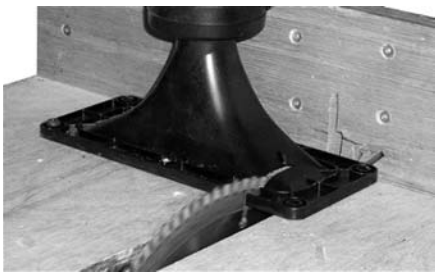


PHOTO 11: Trimming a tweeter.

trimmed and glued together, forming a large horn with one continuous mouth. My prototype used model 1025 tweeters, though there are other options.

Five of these will fit the DR280 baffle after you trim them to 4.5″ long. Cut them on a table saw, preferably using a panel-cutting jig (**Photo 11**). Use either a very fine tooth plywood/laminate blade or abrasive blade. Trim an equal amount from each end of three tweeters to arrive at the desired 4.5″ length, but on the last two cut only one end.

Trim the flash from the cuts with a knife. Glue them together one at a time with medium body ABS cement, which, used to glue black plastic water pipe, is black in color and fills any small voids. While the glue sets, place the array facedown on a flat surface, with waxed paper underneath so it isn't glued to the work surface, and with the edges aligned with a straightedge. Make sure all the + and – polarity markings on the drive elements are aligned in the same direction.

Let them set for a half-hour or so and then apply another layer of cement to the joints. Wait a day, then sand the faces, first with 200 grit sandpaper to even things out, then 400 grit. They end up with a nice even satin finish that looks as though they were cast as one unit (**Photo 12**).

The tweeter baffle is made in two halves, one edge cut at 35°, each about 1.56″ wide on the narrower face, so that when mounted there is a 2.12″ gap between them. Measure the box to be sure of the exact required length, which should be in the vicinity of 23″. Check the flatness of the top and bottom with a straightedge; if there's any warping you can cut the baffle to the required length to push or pull the warp out, as the case may be. Don't forget the 4° angle on each end of the baffle halves.

The drive elements of the tweeter assembly will not fit through a 2.12″ gap, so where the drive elements will go through the baffle assembly the equivalent of 2.75″ diameter holes are required (**Photo 13**). These cutouts are 4.5″ apart, on-center, and they are all shifted 1″ up so that when you install the tweeter unit it slides down an inch; otherwise, the mounting screws would fall on the holes. Install the tweeter baffles (**Photo 14**). Now you can cut the tweeter to its measured final size, both ends at 4°.

BRACING

Cut out four horn braces and two side braces (**Fig. 10**). When laying this out


PHOTO 12: The multi-throat single mouth tweeter array.


PHOTO 13: Tweeter baffle.

on the plywood, nest three more horn braces against the first to speed the cutting process. After cutting them, trace another side brace from the first and cut it out. The bottom horn braces span the throat horn supports and the tweeter baffle, acting as mounting flanges for the access porthole cover. Trim them at either end to fit.

The top horn braces are bisected to accommodate the woofer baffle. Cut four 24″ × 1.5″ half-inch plywood strips. Trim two to span the horn brace from the top to the bottom, about halfway between the baffle and the throat horn supports, with the narrow edge facing the horn sheath. Install the other two about halfway between the woofer and tweeter baffles, with the wide face parallel to the horn sheaths so that they won't interfere with accessing the driver mounting bolts.

PHOTO 14: Tweeter baffle installed.

Two more similar size strips with one edge at 35° back up the tweeter baffle so that it's a full inch thick. **Photo 15** shows a closeup of the bracing, looking at the cabinet top. Note the 4″ hole in the cutoff brace, to remove unnecessary weight from the box.

Use plywood scraps to complete the driver access port flanges. **Photo 16** shows how notches are cut into the flanges to allow the driver frame to fit

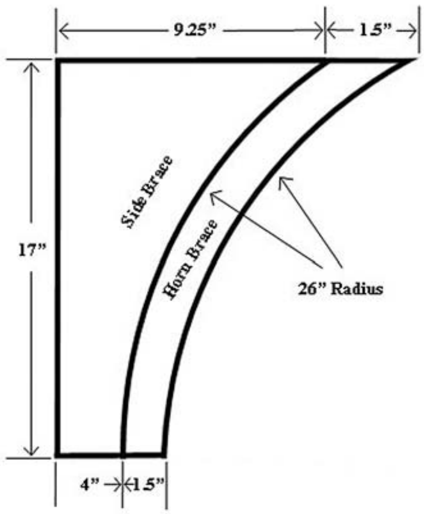


FIGURE 10: Horn/side brace template.

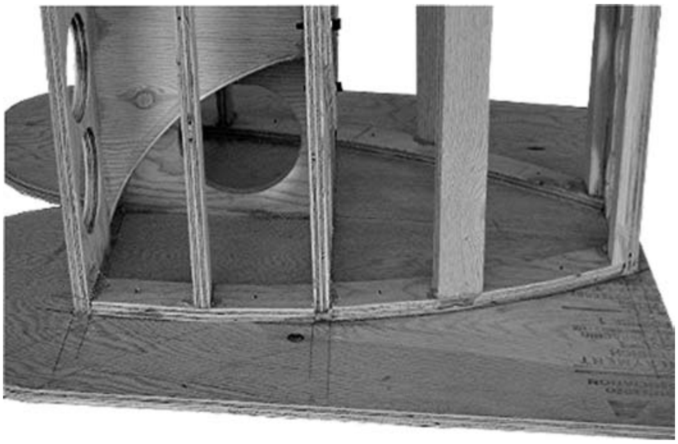


PHOTO 15: Bracing detail.

PHOTO 17: Measuring for the horn sheath.

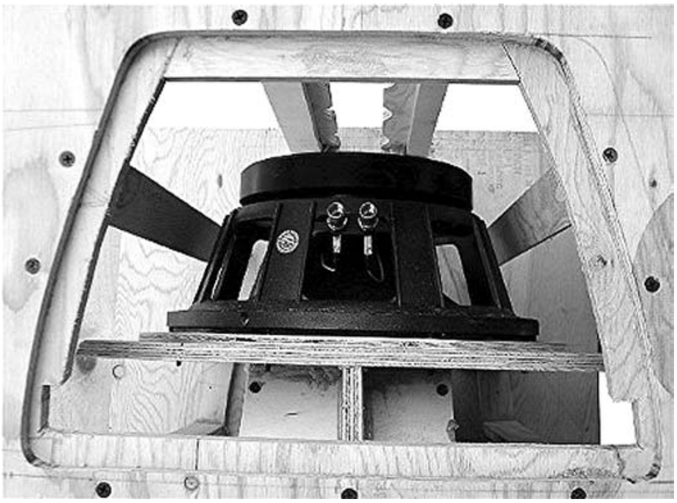


PHOTO 16: Driver access porthole.

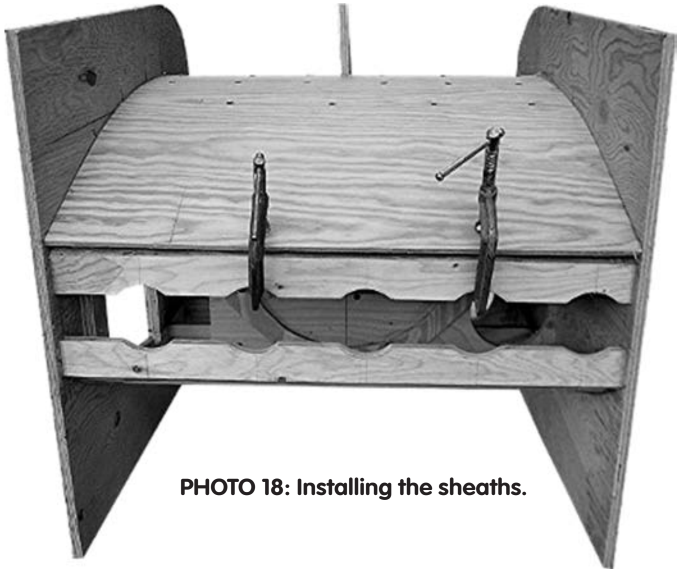


PHOTO 18: Installing the sheaths.

through. Make absolutely sure that your driver fits now, because you won't be able to make any necessary changes later.

Make two back braces—semi-circles the same size as the semi-circular sections on the top and bottom. Install these braces roughly midway on the throat-horn supports, with one end mating with the throat horn divider. Use a clamp and a couple of scrap plywood cauls to hold them in place while you screw them to the supports from the inside of the cabinet. Before attaching them, use a straightedge spanning the top to the bottom to be sure everything aligns properly, trimming the braces as required.

Cut the horn sheaths from ¼″ plywood. Before cutting, flex the plywood to determine the easier bending axis. The rough size of the sheath is 23″ high, 19″ wide. Measure the cabinet to be sure of the actual size before cutting.

Use a framing square to determine the exact shape of the sheaths. First measure the height at the horn supports. Clamp the square to the support (**Photo 17**). Measure at the horn mouth the distance from the edge of the square to the top to see how much higher the sheath must be at that point. Repeat the process with the measurement to the bottom.

To find the width of the sheath, use a flexible tape to measure along the curve of the horn. Cut the length a bit long, and sand off excess. Attach the sheaths starting at the throat horn supports, and screw them to the supports, all the braces, and both baffles. Mark on the top and bottom where the vertical braces and baffle are so that you'll be able to find them with the sheath in place. Use a clamp or two to help hold the sheaths in place while you screw them down (**Photo 18**).

FINISHING TOUCHES

After the adhesive has set, remove any screws penetrating into the driver chamber, lest they bite you later. Fill all of the screw heads and any other glitches in the plywood with polyester auto body filler, which grips tenaciously, doesn't shrink, is ready to sand in a half-hour, and is very inexpensive. Sand it all smooth, slightly rounding over the joint at the tweeter baffle.

Halve lengthwise a 20″ long piece of 4″ ID schedule 40 PVC pipe. To do this use an abrasive blade, securely clamping the pipe to a panel-cutting jig (**Photo 19**). Be careful when finishing the cut because the pipe may have a tendency to close on the blade. Cut the PVC sections to fit between the top and bottom and the back braces (**Photo 20**).

One end will be square, one will be at a 4° angle. Attach these to the cabinet using a hot-melt glue gun. Sand a bit of a chamfer on the edges and ends of the PVC, and on the cabinet where the PVC joints it, producing shallow troughs to hold the glue. Be sure it is totally airtight.

The side braces divide the horn into not quite equal sections; just like the back braces and throat divider, they are offset ¼″ from center, extending from the PVC reflectors to about 2″ back from the tweeter baffle. Use a clamp and wood blocks to temporarily clamp them in position and trace on them the contour of the sheath if the existing shape is more than ⅛″ off, trimming it if that's the case. Then run a straightedge from the top to the bottom at the front and rear to determine their final size.

Attach the braces, driving a couple of screws from the inside of the box. Also clamp the joints with the back braces, using wax paper or polyethylene to prevent gluing the wood blocks as well. Use adhesive to fill gaps in the joints with the sheaths and let the adhesive set before proceeding further.

Cut the back halves from ⅛″ Baltic birch, again flexing it first to determine the easy bending axis. The nominal size of these is 21.5″ high by 23″ wide, which should allow them to extend at least 2″ onto the sides of the cabinet. Attach first one and then the other to the cabinet. Clamping is not required, as the plywood bends easily on this fairly large radius, but a web-style clamp or two makes the job much easier.

After the adhesive has set, trim and sand them flush to the top and bottom. Rough-trimming them is a snap using a saber saw or reciprocating saw equipped with a fine tooth blade at least 6″ long.

This is a good time to put a finish on the inside of the horn. I recommend DuraTex (available from Acry-Tech at 1-800-771-6001), which is a water-base urethane acrylic that goes on with a standard paint roller, drying quickly with a lightly textured finish that's very durable and easily recoated if you ding it. It costs not much more than paint, covers quite well, and you can finish a cabinet of this size in about 20 minutes exclusive of drying time. Use a small trim roller to reach tight spots. Before coating, sand well with 100 grit to minimize wood grain.

Drill a hole through one horn sheath, large enough to run a 14- or 16-gauge speaker wire through to the driver

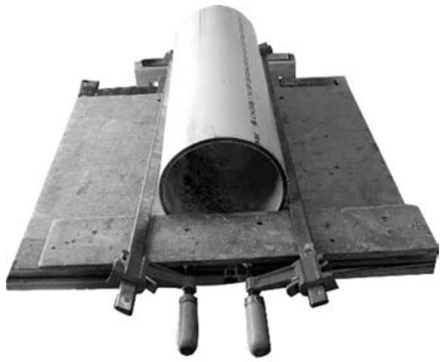


PHOTO 19: Slicing PVC on a jig.

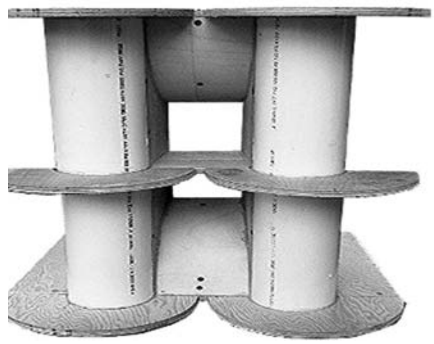


PHOTO 20: PVC reflectors in place.

chamber. Drill another hole in one of the back pieces, as close as you can to the center of the box. Run the speaker wire through, sealing both holes airtight with hot-melt.

Cut out the sides, roughly 24″ high by 24.5″ wide. To mark the final cut lines, attach the rough-cut side to the assembly with a couple of screws or long clamps. Lay a straightedge across the back at both the top and bottom, marking the junction with the sides (**Photo 21**). Add ¼″ to account for the thickness of the back cover, which is overlapped by the sides, and you have your cut marks for the width.

PHOTO 21: Finalizing the side dimensions.

Trace the actual joints with the top and bottom. When cutting to size, leave the panels about ⅛″ oversize. Using a router or dado blade, cut a ⅛″ relief roughly 2.5″ wide where the side overlaps the back. You won't be able to sand the inner surfaces of the sides easily later, so do it now.

Attach the sides. Use a long clamp to square up the side braces until you screw them. If you want to use flush-mounted handles, drill holes for them now, making sure not to hit any braces. Locate the handles so that you can drive screws through the lower mounting holes right through them into the braces, with the remaining fastening done with short bolts and T-nuts.

Trim the ¼″ plywood back cover, rough dimensions 21.5″ by 23″, to the exact height of the back (those edges at a 4° angle to match the taper of the top and bottom) and to the exact width of the cabinet, less 1″ to account for where the side extensions will overlap it. Cut a rectangular hole, about 3″ high by 5″ wide, centered in the back sheath. Install the back sheath, being sure it is centered and true to the cabinet, with the speaker wires reachable through the hole. There is room to clamp a guide-board in place at the rear corners to assist in the attaching process, which is best done with a few small nails. Make sure the nails are set deep so that they won't be hit later when you round off the edges.

Trim the cutoffs saved from the top and bottom to fit the gaps at the top and bottom rear of the box. You can tell by the wood grain where each piece came from. These pieces are very small, so use a panel jig and an attached wood caul for safe cutting (**Photo 22**). To install, slather the joints with plenty of adhesive and hold the filler piece in place by screwing the caul to both it and the cabinet until the adhesive cures, again using wax paper or polyethylene to keep the cauls from being glued (**Photo 23**).

When cured, remove the cauls and fill any remaining joint gaps with adhesive. Sand the whole shebang down, fill any remaining imperfections, and then sand it all again. Use a round-over bit and router and/or sander to round all the exterior edges. Complete applying the finish. Install protective corners and feet.

PHOTO 22: Rear corner filler with caul.

PHOTO 23: Gluing up the rear fillers.

Put the porthole cover in place, drill and countersink for the attaching screws, remove it, and put a finish on it. Fully line the driver chamber and porthole cover with an inch thick layer of either acoustic foam or polyester batting. Install the woofer. To reach the woofer bolts use a ¼″ ratchet wrench outfitted with a 10″ extension shaft and a U-joint adaptor with an Allen head bit, working through the tweeter baffle hole.

WIRING

Cut a piece of ⅛″ plywood an inch or so larger than the hole in the cabinet back for your jack(s). Install the jacks to the plate, wire them, insert the plate, rotate it 90°, and screw it in. Wire the tweeters, connecting all the like polarity terminals, plus two long leads to connect to the jack wire.

Terminals are OK, but a soldered connection is more secure. Use a color coded speaker wire. The same wire you used for the lead to the jacks is OK, but because the current load going to the tweeters is quite small, you can use wire as light as 20 gauge.

Install the tweeter assembly. An airtight fit is critical, so use a silicone sealant to caulk all the joints between them and the cabinet. Connect the long leads from the tweeters to the wire from the jack and then connect that to the woofer terminals, the + coded wire to the red terminal. Gasket the access cover flanges with weather stripping foam and attach the cover. Rock on.

TWEETER OPTIONS

You can use unaltered model 1005 piezos, which measure about 3.375″ square; six of them will fit on the tweeter baffle (**Photo 24**). Instead of the two-piece baffle of the prototype, use a one-piece baffle 5.25″ wide on the front face, with 3″ diameter mounting holes evenly spaced. Wire all six tweeters in parallel. This option, and the five 1025 tweeter version, will handle average input levels of up to 200W. That's plenty for backline bass and keyboard use, or for small to medium club PA.

For higher power levels you need at least eight tweeters. The easiest option is to use model 1016 piezos, which measure about 2.62″ by 5.62″. You can fit eight of them horizontally aligned on the baffle (**Photo 25**). Use a single baffle 5.25″ wide (the sheaths add another half inch, so the tweeters won't overhang) with 3.75″ wide by 2.12″ high mounting holes, evenly spaced. The tweeters are series/parallel wired for 400W power handling. Two banks of four tweeters each are parallel wired, with the two banks series wired.

Other tweeter scenarios achieve both superior wave-front integration and high power handling simultaneously. The first is to use eight 1005 tweeters, cut to a height of 2.875″ (**Photo 26**). Start with a full width baffle, drilled with 3″ diameter holes on 2.875″ centers, which will result in a dual baffle. The wiring scheme is the same as with eight 1016s, for 400W power handling.

Gluing together twelve 1016s (**Photo 27**) offers the ultimate in integration, dispersion, power handling, and looks cool. They are horizontally mounted, with height trimmed to 1.9″. This requires a two-piece baffle with an opening 3.75″ wide, resulting in a face width of only .75″, too narrow for adequate strength. However, the backing reinforcement strips on the baffle need only a 2.25″ gap, so you can cut them wider to afford the baffle adequate strength, as per **Fig. 11**.

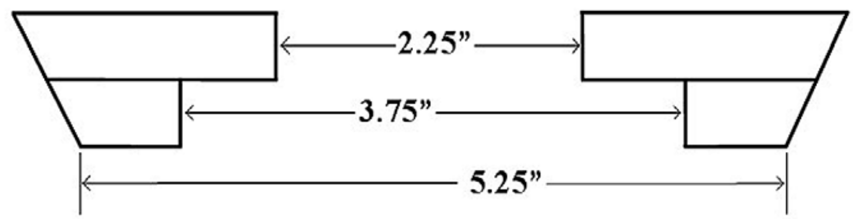


FIGURE 11: Tweeter baffle template/twelve 1016 option.

Wire this super array of tweeters for the highest possible sensitivity and 400W power handling with two banks of six tweeters each in parallel, the two banks in series. For 700W power handling, with a bit less sensitivity, wire three banks of four tweeters each in parallel, the three banks in series.

You also can add a professional touch to your DRs with the addition of this logo (**Figure 12**). Use this link (www.billfitzmaurice.com/logo2.gif) to copy the original .gif image, size it as desired, print it at high resolution on photo quality paper, and use a piece of Plexiglas to protect it. aX

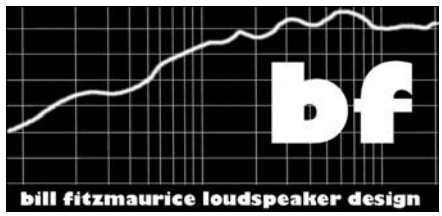


FIGURE 12: BF Loudspeakers logo.

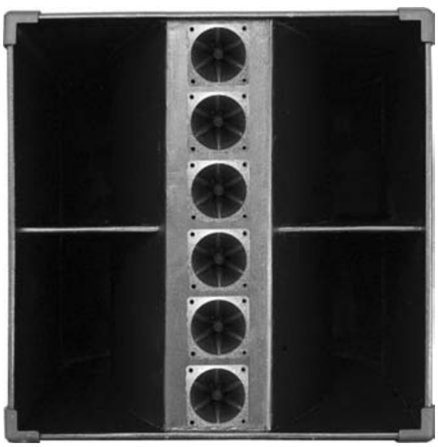


PHOTO 24: Six 1005 tweeters.

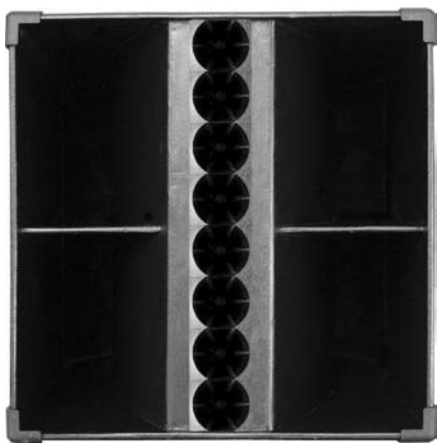


PHOTO 26: Eight 1005 tweeters.

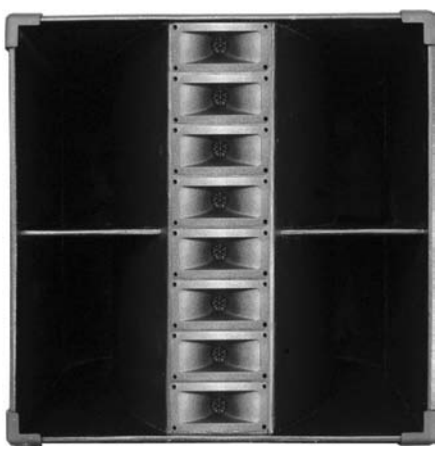


PHOTO 25: Eight 1016 tweeters.


PHOTO 27: Twelve 1016 tweeters.

Tuono: PP PL504 Amplifier

This author's amp design unleashes the thunderous sound of PL504 tubes in a push-pull configuration.

By Claudio Rosada

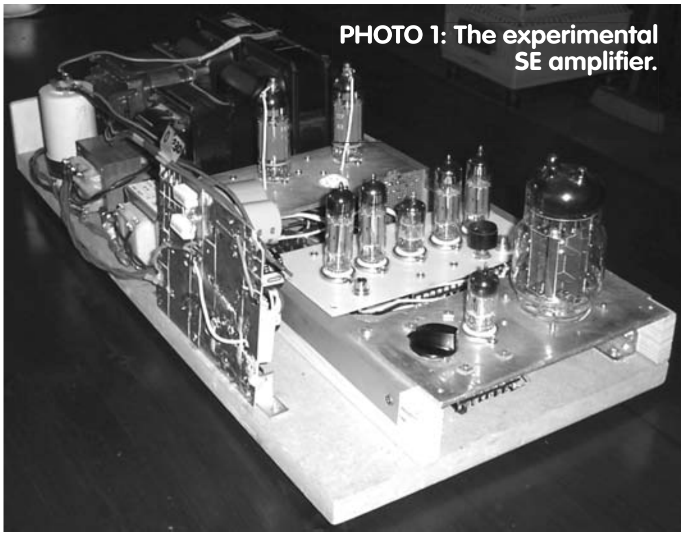


PHOTO 1: The experimental SE amplifier.

There are, in my opinion, some basic ingredients that you must combine to produce a good amplifier. First of all, you must look for a circuit with inherent linearity, avoiding as many "tricks" as possible to improve instrumental performances. Furthermore, an amplifier must be capable of driving "normal" speakers.

It is possible to derive from these generic principles some important technical recommendations. First of all, the device type: tubes and especially triodes have a good intrinsic linearity. Even transistors could be linear, but the very high parameter dispersion of these devices normally doesn't permit the use of simple circuit topology. So feedback is mandatory to produce stable and repetitive performances. I believe that the famous "tube sound" is highly due to the circuit topology.

Thus triode tubes, or triode-connected pentode tubes, are preferable, and global feedback must be avoided. I prefer to design amplifiers capable of driving "normal" speakers. From the electrical point of view, it means that the amplifier must feed the appropriate sound level speaker with 88–90dB efficiency, providing the current required at the minimum impedance point of the speaker.

For these reasons, I try to have at least 10W of output power and to limit the output impedance. The combination of these requirements is a good starting point for a design, but, unfortunately, they are in conflict with themselves! In fact, it is not easy to have low impedance and high power using just triodes without global feedback.

In the last few years I designed two amplifiers following these requirements. The first one was single-ended (SE) using the 6C33 tube with a high load. The second one was push-pull (PP) featuring the PL504 beam pentode in triode configuration. This article focuses on the latter.

Because I built this amplifier for a friend, I briefly discussed his requirements with him. He intended to have just one device replacing his existing transistor-based amplifier with three line inputs and one phono input (for just MM pickups). My basic idea was to design a modular unit to be used as either the final amplifier or as an integrated amplifier adding a preamplifier stage and an optional phono section. The volume control and three line level unbalanced inputs complete the system. The power supply is designed in modular format as well; only minor adaptations are necessary to move from one configuration to the other.

In the following paragraphs I'll describe the design starting from the final stage, because in a logic flow, each stage defines the requirements of the previous one.

THE OUTPUT STAGE

In the last few years I bought some PL504s for around $1 each. This tube was widely used in the past as a "beam" amplifier in TV receivers, so you can still find it at a reasonable price. The PL504 has a mechanical structure similar to the well-known EL34, apart from the anode

TABLE 1: PL504 PERFORMANCE IN SE CONFIGURATION.

					Load 600Ω		Load 1200Ω	
N	Va	Ia	Vg	Pd	P_{max}	THD@ P_{max}	P_{max}	THD@ P_{max}
1	128	115	–14	14.7	1.08	4.86	0.71	3.9
2	140	125	–15	17.5	1.15	4.6	0.74	3.86
3	172	100	–23	17.2	2.46	5.72	1.56	4.24
4	185	100	–25	18.5	3.1	6.0	2.0	4.41
5	212	100	–30	21.2	4.3	5.2	2.9	3.4
6	230	90	–35	20.7	5.5	5.8	3.6	3.6
7	230	85	–36	19.5	5.5	6.1	3.7	3.6
8	248	80	–40	19.8	6.6	7.0	4.5	4.0
9	242	85	–38	20.6	6.0	7.0	4.2	4.0
10	240	80	–38	19.2				

on top of the glass bulb and the unusual heater voltage of 27V. The dissipation is only 16W typical (22.5W absolute maximum), but the big advantage of this tube is the low (compared to EL34 or similar) internal resistance in triode connection (ranging from 200Ω to 500Ω). Also, linearity in triode connection is appreciable.

Unfortunately, there are not many designs based on this tube, so I decided to start evaluation in SE configuration, in order to better define a suitable working point and dynamic load. I built a complete amplifier (**Photo 1**) to evaluate both electrical and acoustical performance. I used a tunable fixed-bias circuit to change the grid bias, and assembled a tube based (using 6C33 triode as series regulator and a 12AX7 as error amplifier) variable voltage-regulator circuit (front of photo) to change the anode voltage.

The driver stage was the first prototype of the one described in the following. I varied the effective load simply by changing the resistive load on the output transformer. You can find the result of such experiments in **Table 1**.

As you can see, output power increases with the anode voltage, but the more interesting point is in the range of 240–250V, which is much higher than the one I was expecting from just looking at the curves. What is not evident in the table is that a further increase in anode voltage is not convenient, because the interdiction area of the tube will increase the distortion, providing no further benefit in terms of maximum output power. The conclusions of this experiment are that point numbers 8/9 are good candidates for the working point, while as far as the dynamic load is concerned, increasing the load produces limited effects on the distortion, while highly limiting the output power.

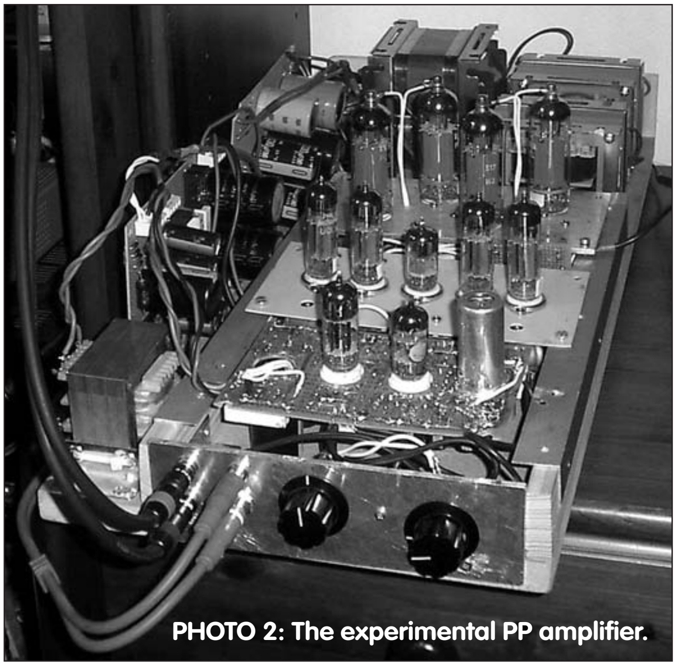


PHOTO 2: The experimental PP amplifier.

So I decided to use point number 10 with a dynamic load of 1200Ω. Someone may note that the dissipation in this point is 19.2W, much more than the typical 16W. To be honest, I don't know the real effect on the tube life of this condition. But considering that the absolute maximum value is 22.5W, I believe that this stress will not affect the tube life much.

Another important result of my SE experiment was the evaluation of driving capability required for the PL504. In fact, this tube, when dynamically driven, is sinking current, even at negative grid point. I was not able to characterize such current, so I cannot provide any figure about it, but I concluded that a high current driver was highly recommended for such a tube. **Table 2** shows the THD+N performance versus output power of the SE configuration for point n.5.

In order to fulfill the requirement of power in excess of 10W, one tube is not enough, so you must look for a parallel SE or PP configuration. After some trial and error with PSE configuration, I turned to PP, in order to investigate the performance and acoustic results that this topology can provide. All the experimentation done on the SE configuration was very helpful in defining the requirements of the output transformer for the PP configuration and of the driver stage.

THE OUTPUT TRANSFORMER

The requirements for this component are: 1) push-pull configuration; 2) dynamic impedance in the range of 1200Ω for each tube (resulting in 2400Ω rail to rail); 3) maximum output power at least 15W at 30Hz; 4) 20kHz bandwidth at full power. Even if this type of device is available on the market, I decided on a custom transformer. So I provided my requirements to a friend who has good experience in designing audio transformers. A few weeks after the first two samples were available, I started experimenting with the push-pull configuration. I measured a total primary winding around 64Ω.

The transformer was a device using oriented grain iron 0.3mm thin with an E/I shape and secondary winding implemented in double wiring. I measured a primary winding resistance around 32Ω, so if you use a different transformer, you might need to rearrange the power supply. The winding ratio for the whole primary winding is 18.54, resulting in an effective rail to rail load of 2750Ω on an 8Ω load.

THE INVERTER/DRIVER STAGE

The following requirements are necessary to feed the output stage: a) output dynamics in the range of 40V pick corresponding to 28.4V RMS; b) high driving capability to feed grid current; c) low distortion (you can put a limit of 1% at maximum modulation); d) full audio bandwidth (obvious); e) good dynamic balance over the full spectrum. I spent some time analyzing the existing inverter stage, and concluded that all these requirements cannot be fulfilled with only one stage at a reasonable cost. In fact, even if transformer-coupled or LC-coupled stages can probably provide the required current, they increase the complexity and cost of the design too much. So I determined that a cathode-follower stage was a good candidate for the grid feeding. Considering the variety of possible inverter stages, I prefer the Smith model for its symmetrical look.

Unfortunately, the Smith inverter can provide a good balance only when the cathode load of the two tubes approaches the infinite value. Mark Kelly's article in *Glass Audio*[1] offers the right idea. In

TABLE 2: PL504 PERFORMANCE IN SE CONFIGURATION AND POINT 5.

Power	THD+N at 600Ω	THD+N at 1200Ω	THD+N at 2400Ω
0.1	1.1	1.1	0.75
0.2	1.3	1.1	0.97
0.5	2.0	1.6	1.6
1	2.9	2.5	2.6
2	4.4	3.7	
4	6.8		

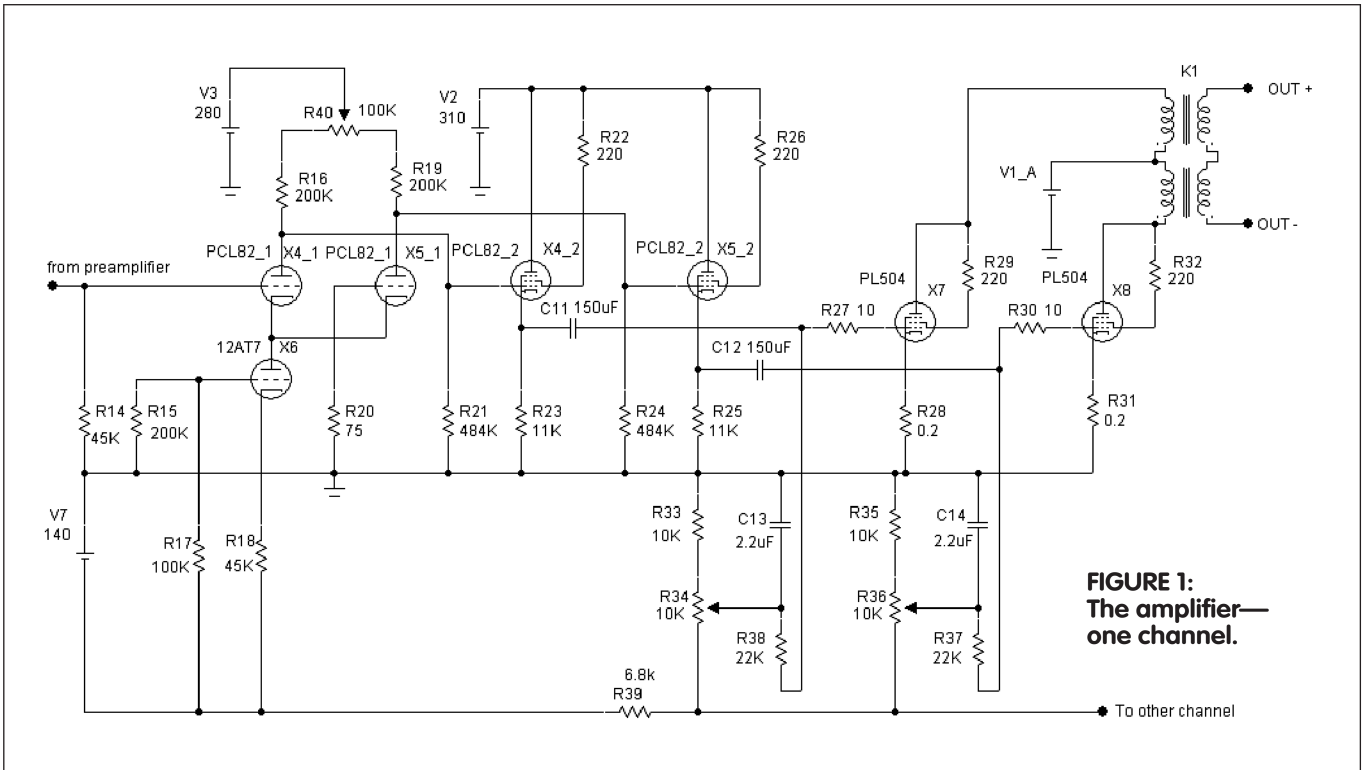


FIGURE 1: The amplifier—one channel.

fact, if you replace the resistive load with an active load, the balance of the Smith inverter can be highly improved. Such an active load can be implemented very simply with a triode, adding just a double triode component to the parts list. The combination of Smith inverter and cathode-follower current buffer make possible a DC coupling between the two stages, thus reducing the number of capacitors in the signal path.

This solution complicates the design, because now the two stages are strongly coupled and any deviation on the bias of the first stage impacts the second one. To implement the circuit topology previously defined, you have basically two different possibilities: a) use one high-mu double triode (e.g., 6SL7 or 12AX7) for the Smith inverter and a medium-mu low-impedance double triode (e.g., 6SN7 or 12AU7) for the buffer; b) use two asymmetric double triodes. Because I recently bought some UCL82 triode/pentode tubes, I decided on the second alternative.

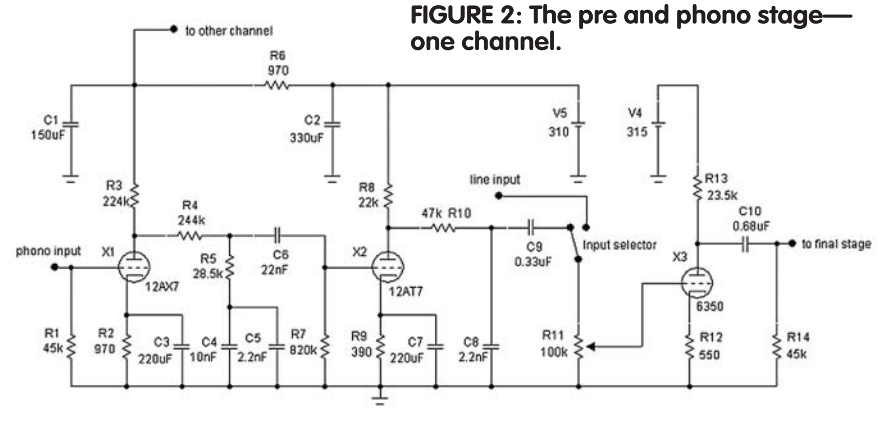


FIGURE 2: The pre and phono stage—one channel.

The xCL82 family, which is widely known in Europe, is available with three different heater voltages: 6.3V = ECL82; 16V = PCL82; 50V = UCL82. The equivalent US versions are 6BM8, 16A8, and 50BM8. Due to the wide dispersion of these tubes in the past, they are still widely available as NOS at reasonable cost. The tube was originally designed for audio applications and mainly used in TV and radio receivers.

The pentode section of such a tube, when connected in triode configuration, provides a very low impedance drive, in the range of 250Ω. After fine-tuning, I chose the following working points for the two sections: Vg = –1.5V, Va = 300V, Ia = 0.6mA for the triode; Vg = –27V, Va = 230, Ia = 12mA for the pentode. The dissipated powers are 0.180W and 2.76W, well beyond the limits (1W and 7W, respectively).

The polarization current for the Smith inverter is obtained with a 12AT7; it is polarized in order to provide the 1.2mA required. Because the xCL82 tubes were used only in the triode/pentode configuration, no standard documentation is available for the triode connection of the pentode section. But this tube is widely used by Italian DIYers, and such documentation was published some years ago in an Italian magazine[2]. Because the Smith inverter is doing even harmonic cancellation, in order to obtain the best performances, a dynamic balance of the stage is helpful.

While static balance is good—because the xCL82 tubes were produced with high quality, which also means low parametric dispersion—it works on dynamic balance in order to improve harmonic distortion. But in order not to complicate the stage too much and to reduce the tuning operations, I suggest you simply buy a few more tubes and try to exchange them until you achieve a good combination of performances for the two channels (left/right). The distortion results of such a stage are presented in **Table 3**. I measured adding my distortion analyzer (with 100kΩ impedance) in parallel to the grid load of the final stage.

As you can see, because the instrument is doing the THD plus noise measure, the low-level signals are penalized by the residual power-supply ripple (refer to the power-supply section for more details). The driver/inverter stage has a total gain of 21.5 (and a maximum acceptance of 1.5V peak in class A1 corresponding to 1.06V RMS), so the required input signal to drive the final stage at full power in class A1 (corresponding to 38V peak) is approximately 1.76V peak. This means that at full power the triode section of xCL82 will be slightly driven in positive grid. This should not be a problem because the recommended bias in the data sheet is Vg = 0V.

To be honest, I tried without success to limit the working condition to a pure class A1 operation, but, unfortunately, any attempt to do so (due to DC coupling) impacts the bias of the pentode section, thus limiting the dynamics of the second stage. This operation in class A2 of the first stage limits the use of the high-impedance preamplifier section. Sensitivity should not be a problem for a CD player, but it is still far from the standard for an integrated amplifier (it should be more properly in the range of 0.5V RMS or so). In some cases, a preamplifier section might be necessary.

THE PREAMPLIFIER STAGE

My idea, as stated at the beginning of the article, was to build a modular integrated amplifier, so I added the pre stage just to have a better sensitivity. Most potential users don't really need this stage at all. The requirements for the preamplifier section are: a) low gain, in the range of some units; b) low impedance in order to drive in class A2 the further stage without degradation. For this simple stage I immediately chose the single-ended configuration with RC coupling. There are many different low-mu and low-impedance tubes that you can use in such an application—most of them designed for audio.

I decided to try a relatively unknown and inexpensive tube: the 6350 twin triode. As stated in the data sheet, this medium-mu twin triode was designed for use in electronic computers and other "on-off" control applications, involving long periods of operations under cutoff conditions. The tube is manufactured using high-quality materials and it is deeply tested. Unfortunately, according to my experience, this tube has two important drawbacks for audio applications: It is highly microphonic and sensible to heater noise injection. The tube has a relatively good linearity when used in low signal applications, such as preamplifier.

After analyzing the curves, I fixed the following working point: Vg = –4.35V, Va = 115V, Ia = 7.9mA. The resulting power is 0.9W per triode, far from the maximum value of 7.7W total. The tube has a mu of 18 and an internal resistance in the range of 4kΩ. When used in my SE configuration—without the bypass capacitor on the cathode resistor—the effective gain of the stage drops to the range of 10 (I measured 9.8 for both channels), which is more than needed, providing to the whole amplifier a sensitivity of 122mV RMS at full power.

The distortion performances under those conditions and with the effective circuit load are presented in **Table 4**. I measured adding my distortion analyzer in parallel to the grid load of the Smith inverter stage. As in the previous measurements the low-level signals are penalized by the residual power-supply noise. Note that distortion is very low (around 0.1%) as long as the further stage is in pure class A1 operation, while it increases when its grid starts to sink current.

You must use the tube with a DC heater to avoid excessive hum injection, and the tube is also very sensitive to mechanical vibration. In order to reduce the pickup effect to an acceptable level, install two dumping rings on the tube in case of vertical installation, while in horizontal installation they should not be necessary. If you adopt these precautions, the tube will provide a good performance for a reasonable cost.

THE PRE-PHONO STAGE

I added this stage at the end of the design to provide a direct turntable connection with pickup having sensitivity in the range of MM types. I think that designing a tube-based phono preamp is probably the most difficult challenge for a DIYer. In fact, the very low level of the incoming signal, its equalization, and all the potential problems related to ground loops—magnetic coupling, tube microphonicity, and so on—greatly limit the performance of such devices.

I tried to solve the problem by embedding the phono stage, including the preamplifier stages and their power filtering, in a very compact form. I chose the topology that is sometimes considered in the DIY community as the best

TABLE 3: UCL82 BASED DRIVER/INVERTER PERFORMANCES.

Output volt [V RMS]	THD+N left	THD+N right
1	0.97	0.82
2	0.48	0.39
5	0.21	0.24
10	0.22	0.39
20	0.58	0.79
30	1.28	1.36

TABLE 4: 6350 PREAMPLIFIER SECTION.

Output volt [V RMS]	THD+N left	THD+N right	Amplifier output power [W]
0.1	0.32	0.34	
0.2	0.18	0.19	
0.5	0.10	0.11	2.7
1	0.14	0.12	11.9
1.2	0.21	0.23	16.2
1.5	0.45	0.58	Clipping

one in terms of fidelity—the passive split RIAA. The real drawback of this topology is the S/N ratio, because the equalization is done in a totally passive way, so it is necessary to have a very high amplification factor in order to loose it with further attenuation. The RIAA equalization network requires three time constants: 1) low-frequency attenuation at 3.18ms; 2) low-frequency emphasis at 318µs; 3) high-frequency attenuation at 75µs.

In the split RIAA topology these time constants are separated between two gain stages and implemented using two equalization networks, one after the first active stage and the other after the second. In my implementation the first network implements the first two time constants, while the second network implements the third one. In addition, this circuit introduces a subsonic filtering by means of two RC decoupling networks that isolate each stage. This filtering—even if not mandatory in RIAA standard—is preferable to decouple low-frequency signal fluctuations coming from the pickup.

When the circuit topology is defined (fixing the position of the two filters), you just need to find the value of the RC components in order to have the proper equalization curve. A first definition can be done using simplified formulas, but a fine-tuning requires simulation tools. The main problem is to find the right tubes and topology for the active stages.

A proper overall gain for MM pickups at 1kHz is normally in the range of 40dB. Considering an insertion loss of 20–25dB for the passive equalization, you need a total gain of 60–65dB.

In order to improve S/N ratio, the first stage must have the highest possible gain, still maintaining the capability to feed the following passive network. To do that, high gain tubes are mandatory, so the straightforward solution is the 12AX7 tube (the highest gain tube for audio I know). Basically you still have the possibility of using it in SE or SRPP configuration (as is well-known, the second one has the big advantage of dividing by two the output impedance of the stage, maintaining a very high gain), but in order to have a very compact stage, I used the first option (the gain is 36dB).

The situation for the second stage is even more critical: It must provide an acceptable total gain while driving a further stage with an input load in the range of 100kΩ (the highest potentiometer value normally acceptable for S/N reasons). Because these two requirements are in conflict, some designers are introducing a third stage for current buffer, solving the problem in this way. In order to avoid an additional stage, you must find a tube with a still high mu but a reasonable output impedance. There are some tubes suitable for this stage, but the only one available in my stock was the 12AT7, so my decision was very easy. Once again, you have the option of SE and SRPP configurations, but for the same reason above I chose the first one.

In this configuration the stage is providing a gain of 31dB. To have the highest gain, both the cathode resistors are bypassed with capacitors. Because the first equalization network has a loss of 22dB and the second one of 4dB, the overall gain at 1kHz is 36 + 31 - 22.5 –4 = 40.5dB, just as expected. The maximum acceptance of the preamplifier is limited by the second stage polarized at -2V, thus it is corresponding to 400mV at 1kHz, while the minimum acceptance is in the range of 50mV at very low frequency. To get a reasonable precision on the RIAA curve implementation, you must implement all the stages with 1% resistors and the best capacitors you can find.

I suggest you buy more capacitors than needed and then select them (the simple capacitor measure included in most testers is sufficient). Many words have been spent about the sound of the capacitors, and this stage is really one of the points where you can highlight the differences of each cap's technology. I suggest you start using standard MKT and then, if you prefer, move to paper/oil and so on. To be honest, my amplifier is still running with the original MKT of the first shot, because I think they are in line with the general quality level of the amplifier. Also for S/N improvements, I migrate immediately to DC heating (the results with AC heating were totally unacceptable).

I also tried to put the heater at a low positive voltage; in this way the leakage electrons leaving the heater are rejected from the cathode (that is at ground potential) improving the S/N ratio. To be honest, I didn't try the two solutions in order to verify the real effect on the noise level. You will find the Microcap model of the phono stage on my website, www.tube-friends.com.

Microcap is a simulation program available in evaluation version for students. You can download it at www.spectrum-soft.com. Even with the limitations (in terms of maximum components) of the evaluation version, this program is suitable for most of the simulation that could be necessary in the design of tube-based amplifiers. The real limitation of this program is the number of existing models for tubes. In fact, while it is quite easy to build a simplified tube model using the "3/2 power low," it is more difficult to get SPICE models affordable in the saturation region or in positive grid driving.

On my website you will find the SPICE library of the simplified models for all the tubes I used in these designs (but for the pentode I just produce the triode model). For additional information on tube SPICE modeling, please refer to the very interesting article at www.birotechnology.com/articles/Vtspice.

Even with its limitations (considering that in phono stage the working point of the tubes is in full class A area), the simplified model is quite good and the simulation provides results perfectly comparable with the real circuit.

THE POWER SUPPLY

In order to contain system costs and dimensions, I used silicon diodes and RC cell filtering for the power supply design. The power supply features two main sections, with a modular approach. In the first section there are the anode voltages for the two final stages (V1A and V1B), the anode voltages for the driver invert stage (V2 and V3), and the negative voltage for the final stage polarization and active load (V7). I implemented this section using two secondary windings of a custom 250VA main transformer. One winding derives both Va1 and Va2 (with two independent outputs of the winding). Another secondary winding is used for Vd.

We can also include in this section the heater supplies for the PL504 and 12AU7 for the active load. They are provided by two 12.6V sections of the main transformer connected in series with the center tap grounded (**Fig. 4**).

For the UCL82 I added a dedicated transformer (TR3). Because 54V is not a usual value, I used a 48V transformer with more power, in order to have a higher secondary voltage with a more limited load (because normally the nominal voltage value is granted at full power). If you replace the UCL82 with PCL82 or ECL82, you must also change the transformer. All these heaters are powered in AC.

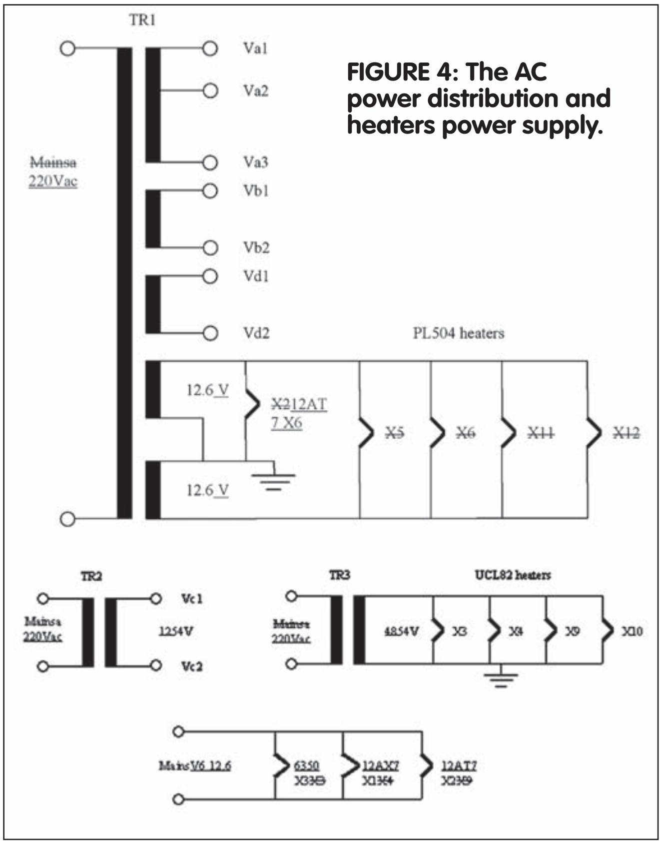


FIGURE 4: The AC power distribution and heaters power supply.

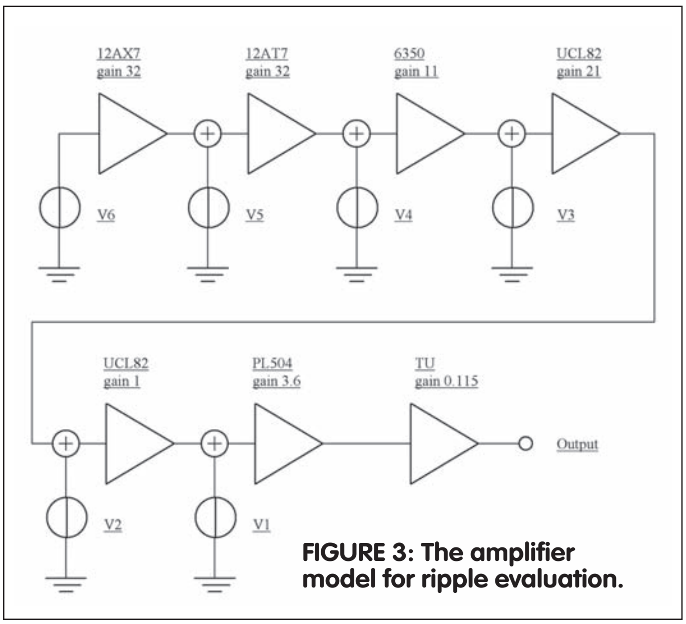


FIGURE 3: The amplifier model for ripple evaluation.

The second section of the power supply includes all the anode voltages for the preamplifier and the phono stage. They are derived from a single section of the custom transformer (Vb). The preamplifier heater power supply is provided by a dedicated transformer (TR2).

In this case I used a standard 12V transformer with a little higher power in order to get 12.6 DC voltage after the filtering. I recommend you use a dedicated transformer anyway, in order to reduce parasitic coupling between the heaters of the pre section and the other sections of the main transformer. With a different transformer, you can arrange the resistors R12 and R13 in order to get the proper voltage value after filtering.

My design, based on a 220V 50Hz mains, is optimized for the 230V level, since this is the normal value available in an Italian house. For the design of the RC filters I used the PSU program that you can download at www.duncanamps.com/psud2/. This program provides good results, but a fine-tuning with the real components is necessary.

The simulated ripple voltages are quite precise, assuming that you know the ESR of the capacitor you are going to use. But I did not, because normally I buy them from surplus and cannot get their complete documentation easily. The schematics include the winding resistance, so you can easily adapt the power supply to different transformers.

To design the power filtering I used a simplified model of the amplifier illustrated in **Fig. 3**. In the model each power-supply ripple is considered a voltage generator in the amplification chain. The noise contribution on the load for one ripple generator depends on the number of following amplification stages. To simplify, I considered the total noise as a sum of each contribution. So if you fix an acceptable ripple level on the load, you can define the maximum ripple level for each stage by the model.

Furthermore, for the push-pull section of the amplifier (from Smith inverter to final PL504 push-pull), you must consider the harmonic cancellation effect in the stages that increase the CMRR, so the ripple requirements for this section are more relaxed. I measured a ripple attenuation factor around 100 in these two stages. For the previous stages in single-ended configuration, this discount is no more applicable and the ripple requirements become stronger at each stage.

The measured results compare well with the estimated ones, as far as the PP section is concerned. The actual noise of the SE section is more influenced by the layout and parasitic coupling effects, so the result can be very bad in an inappropriate layout implementation. I can say that the PP section is quite easy to be realized while, considering the noise problems I encountered with the 6350 tube and the pre-phono stage, I can recommend the implementation of these two sections only to expert DIYers. I fixed as design target a ripple level of 1mV RMS, corresponding to 81dB of S/N, which is normally an acceptable value.

For the power supply 5% resistors are acceptable. Using different transformers, you would need to fine-tune some resistors in order to obtain the indicated voltage values with a reasonable error (1–2%).

THE ASSEMBLY

I followed a modular approach in assembling the amplifier. On one PC board with a metal plane (4″ × 8″, more or less), I assembled the driver inverter section (five noval tubes), while, to increase mechanical strength, I directly assembled the final stage on an aluminum plate with the same dimensions (four magnonoval tubes). The four trimmers for plate current tuning are soldered on a small PC board—it must be easily accessible when the circuit is assembled.

The power-supply filters are installed in two PC boards (4″ × 6″, more or less), with the exception of the pre-phono filtering section, which is directly assembled on another PC board with the same dimensions, hosting also the active section (three noval tubes). I used gold-plated ceramic sockets, and I recommend the shielded socket for the 12AX7 of the pre-phono. Vibration dumpers are generally not necessary, apart from the 6350 tube, where they are really mandatory.

For all the active sections, even using PC boards, I realized a 3D assembly by means of contact strips directly screwed on the PCB. The metal plane of the PCB is used as diffused ground plane.

I first installed all these circuits on an experimental wood base (**Photo 2**), but later transferred them to a metal box, which I had originally designed for another amplifier (**Photo 3**). This box is a combination of an iron structure with black deposition on surface and wood panels for aesthetic purposes and is divided into two totally shielded sections. In the rear section the two output transformers are installed one on top of the other.

This box provides both electrical and magnetic shielding. In the front section of the box all the power transformers and circuits are installed in the lower part, while on top a metal plate sustains the active circuits. In this way the tubes (apart from the phono section that is mounted internally) are visible from the outside through a big window in the front plate (**Photo 4**).

A totally drilled top metal cover is


PHOTO 3: The mechanical kit.


PHOTO 4: The final look—tubes.

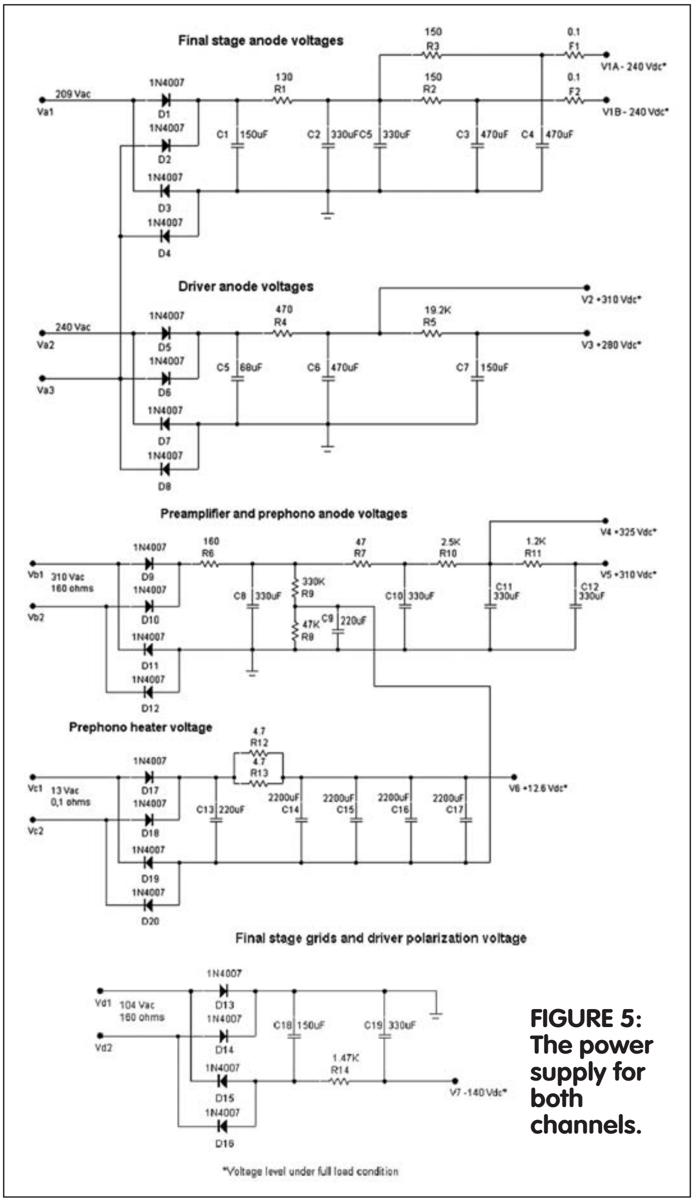


FIGURE 5: The power supply for both channels.

PHOTO 5: The final look—front view.

completely shielding the tubes section, providing a suitable air flow to refresh them. All the connections are on the rear, directly screwed on the metal structure (**Photo 6**). The system is completed with a wood base and a wood front panel. I also designed special knobs (one for volume and one for source selection) and feet (three of which are screwed on the wood base) in brass (**Photo 5**).

THE TUNING AND OVERALL PERFORMANCES

For the active circuit it is preferable to use 1% resistors. Sometimes the required value can be achieved by combining (series or parallel) two resistors.

The circuit requires three different types of tuning. Due to the DC coupling, the inverter driver section is sensitive to power-supply values (V2, V3, V8, in **Fig. 1**). In order to ensure the class A polarization for the first section of the PCL82 tubes, you must adjust R17 (one channel) for +1.5V at the cathodes of X4_1 and X5_1. With the voltage levels of the schematics, R17 of 100kΩ is providing exactly 1.5V in my circuit. Pay attention that a voltage value higher than 1.5V—even if it is interesting in order to avoid class A2 operation for the X4_1 and X5_1—will move the polarization of the following X4_2 and X5_2 tubes, resulting in an unacceptable reduction in the circuit dynamic.

The second tuning is necessary to fix the plate currents and cancel the DC components in the output stage. The cancellation of the DC component is mandatory to avoid premature saturation in the output transformer. Even if a normally good output transformer can tolerate a small DC component, it must be reduced to a minimum (because it is impossible to guarantee a perfect balance over time). You must use the trimmers R34 and R36 (one channel) to do this.

PHOTO 6: The final look—rear view.

I suggest you start switching on the amplifier without the PL504 tubes and tune the trimmers for around -42/-45V at the grids. Then insert the tubes and, finally, control the total plate + grid current by measuring the voltage drop across the resistors R28 and R31. Due to the high value of the capacitors C11 and C12, the voltage will change very slowly when you move the trimmers, so the procedure will require attention in order to avoid a long trial-and-error sequence. For an 80mA current you must measure 16mV on 0.2Ω resistors. As usual, do the tuning after at least 15 minutes from power on and repeat after 10–20 hours of work.

I also added a third tuning circuit for the AC balance and will explain why. In **Fig. 6** you can see the output spectrum measured on an 8Ω resistive load with a pure 1kHz tone generating an output power of 8W. Without any AC compensation the second and third harmonics are comparable. After compensation, you can reduce the second harmonic by at least 12dB (right channel), producing a benefit on the total harmonic distortion (**Fig. 7**).

But I must emphasize that while the AC compensation reduces the THD figure thanks to the even harmonics cancellation (0.4% at 5W is very good for an amplifier without global feedback), its effects are not totally predictable. What is really important is the effect on the sound.

With some music types the uncompensated version looks to me more natural than the other one. In order to apply AC compensation you need a spectrum analyzer (a PC audio board plus adequate SW is fine) or a selective voltmeter (I used a voltmeter tuned on the second harmonic). Trimmer R40 (one channel) must be rotated in order to minimize the second harmonic or THD (because odd harmonics are unaffected, it is almost the same).

The –3dB bandwidth at 10W is ranging from a few Hertz (–0.4dB at 20Hz) to 19kHz (–3.5dB at 20kHz). The noise level (with line input in cc) is around 700µV, corresponding to a S/N value around 85dB.

CIRCUIT DRAWBACKS

Due to the problems I encountered designing and testing Tuono, I must mention two possible improvements for the circuit. First of all, the high value capacitors in the coupling of the driver stage with the final stage require a relatively long period to be charged at power-on (some seconds). The result is an overcurrent pick on the final tubes that could affect their life, so a power-on delay circuit on V1 voltage of 30 seconds could be a good improvement.

Furthermore, the grid polarization of X3 through the volume potentiometer could pick up a lot of noise in a critical layout. In this case a second resistor of

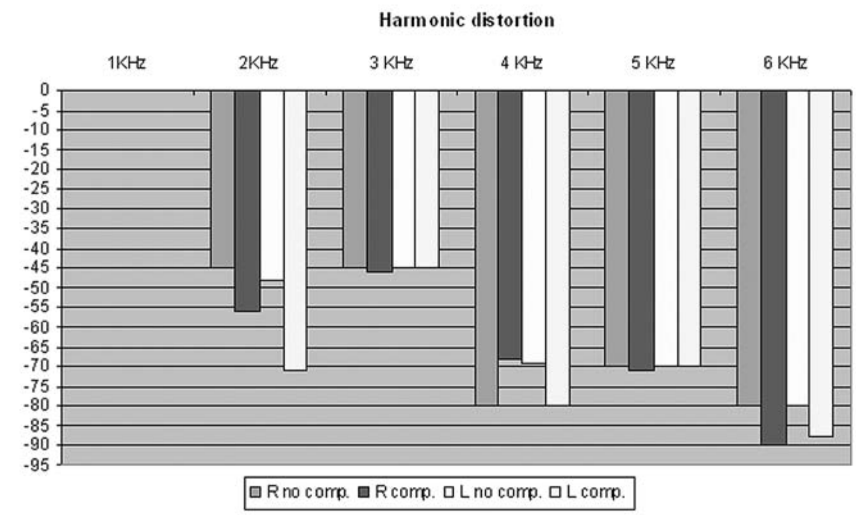


FIGURE 6: The output harmonics with and without AC compensation.

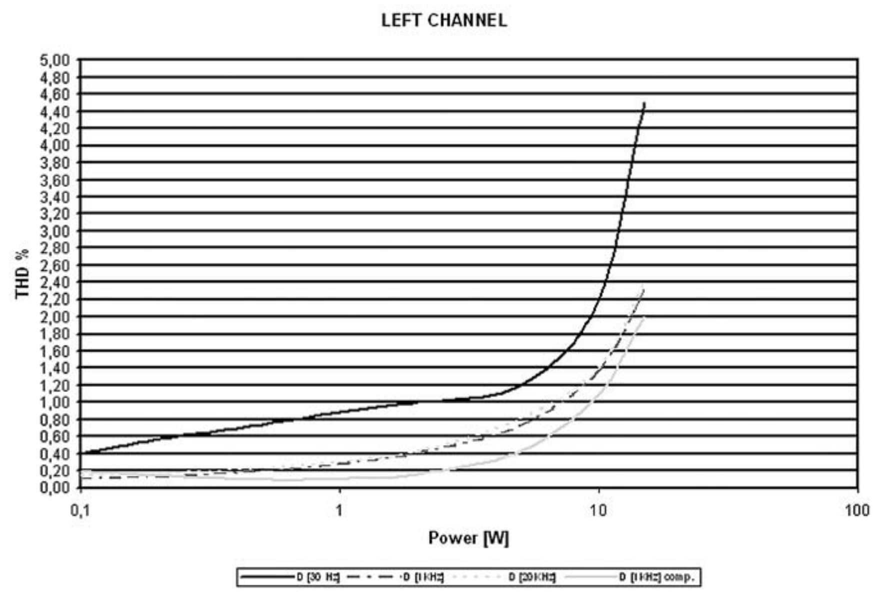


FIGURE 7: The output THD+N vs. output power.

AMPLIFIER PARTS LIST—ONE CHANNEL.

Reference	Value	Power	Voltage
R1, R14, R18	45k	½W	
R2	970	½W	
R3	224k	1W	
R4	244k	½W	
R5	28.5k	½W	
R6	970	5W	
R7	820k	½W	
R8	22k	5W	
R9	390	½W	
R10	47k	½W	
R11	Potentiometer 2x100k Alps Blu series		
R12	550	½W	
R13	23.5k	5W	
R15	200k	½W	
R16, R19	200k	1W	
R17	100k	½W	
R20	75	½W	
R21, R24	484k	½W	
R22, R26	220	½W	
R29, R32	220	2W	
R23, R25	11k	10W	
R27, R30	10	½W	
R28, R31	0.2	½W	
R33, R35	10k	1W	
R34, R36	Trimmer 10k	1W	
R37, R38	22k	½W	
R39	6.8k	2W	
R40	Trimmer 100k	1W	
C1	150µF		400V
C2	330µF		400V
C3, C7	220µF		35V
C4	10nF		800V
C5, C8	2.2nF		800V
C6	22nF		800V
C9	0.33µF		400V
C10	0.68µF		400V
C11, C12	150µF		250V
C13, C14	2.2µF		250V
X1	½ 12AX7		
X2, X6	½ 12AT7		
X3	½ 6350		
X4, X5	UCL82		
X7, X8	PL504		
K1	See text		
F1, F2	200mA fuses		

high value very close to the X3 grid will reduce the effect. Be careful that this doesn't change the phono stage gain and equalization; otherwise, a redesign of the network could be necessary.

POWER SUPPLY PART LIST—BOTH CHANNELS.

Reference	Value	Power	Voltage
R1	130	10W	
R2, R3	150	10W	
R4	470	5W	
R5	14K	2W	
R6	160	5W	
R7	47	2W	
R8	47K	½W	
R9	330K	2W	
R10	2.5K	5W	
R11	1.2K	2W	
R12, R13	4.7	5W	
R14	1.47K	2W	
D1-D20	1N4007		
C1, C7, C18	150µF		400V
C2, C5, C10, C11, C12, C19	330µF		400V
C8	330µF		450V
C3, C4, C6	470µF		400V
C9	220µF		100V
C13, C14, C15, C16, C17	2200µF		35V
TR1	220V	See *Fig. 5*	
TR2	220V/12V	See text	
TR3	220V/48V	See text	

HOW DOES IT SOUND?

As every DIYer knows, it is very difficult to be impartial when you spend more or less one year of your free time on a new design. I compared Tuono using my preferred music with my personal reference (a single-ended using 6C33 tubes). Due to the same technical approach, both amplifiers have the same correct extension in frequency (bass is deep and robust) and enough energy to follow all the transients of the classic orchestra. The sound stage is well-extended in width and height.

As in the major part of the tube-based amplifiers, you can have all the music details even at a very low sound pressure: In a normal room with medium sensitivity loudspeakers, you will never look for more power. Tuono is just losing a little definition compared to SE configuration. On the other hand, with guitar and voices, Tuono is providing its best, it sounds for sure different and, at a first listening, better than the SE. Furthermore, AC compensation can introduce a degree of freedom by better characterizing the "push-pull like" sound. aX

***Claudio Rosada** is an electronic engineer with major experience in digital circuits. He spent ten years designing telecommunication equipment and from 1998 has been working in the automotive electronics field. He "discovered" tube-based electronics a few years ago to improve his personal hi-fi equipment. Now he is active with a group of friends in designing and comparing different tube-based amplifiers and preamplifiers. His preferred music is pop/rock from the '70s and '80s and full-orchestra classical. He often uses songs with voice and acoustic guitar as well as symphonies with high dynamics for accurate comparative test sessions. Claudio can be contacted at* claudio@tube-friends.com.

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Greening of the Behringer Ultra-Drive Pro DCX2496

By Jan Didden

You'll enjoy this mod project, which adds a replacement analog output board to this useful Behringer crossover unit.

Face it, the world is analog in nature. Yet, digital technology has its advantages. Witness the Behringer Ultra-Drive Pro DCX2496 (I'll call it the DCX from now on), with its two analog balanced stereo inputs, an AES/EBU digital input, six ADCs, a Digital Signal Processor (DSP), and six DACs. Plus a handful of op amps to size the analog signal up or down, and six analog balanced output channels. You can use the DSP for a variety of signal processing before sending a signal to any of the six outputs.

One obvious use of the six processing channels is for a digital three-way stereo crossover. In stereo use, you can link each pair of output channels (low, mid, high) so that any setting you make to one side is automatically copied to the other side. The crossover settings include filter frequency and filter type (Butterworth, Bessel, or Linkwitz-Riley, from 12dB/oct to 48dB/oct).

You can set the level for each output individually, set the input levels, and also insert bandpass or band reject filters, with varying Q and lift/drop levels to correct humps and troughs in your speakers' frequency response. You can modify the lf response to correct room errors. There is even an auto-delay mode (if you have the optional mike) to correct for unequal delays from speakers to the listening position.

There is no limit within the capabilities of the DSP, and the display will show you even how much, in %, remains of its processing power. You can save a particular setup with a name of your choice so you can have a series of setups if you use your unit at different places or systems (or music). And all that for a street price around $300 US. I have used this unit for two years now, and love its quality, flexibility, and well-thought-out user concept, and even bought a second one for my other system. But, being an audio DIYer, I couldn't leave well enough alone.

Because I stay away from DSPs and other magic, this article is about improving the analog portion of the DCX, in particular the output amplifier section. This is a low cost unit, so those op amps aren't so hot, and the judicious use of electrolytic signal coupling caps (arrgh!) and—Ohm forbid!—the use of a transistor for muting means that there is much that can be done better. But, I also wanted it to be fully reversible.

The end result is a plug-in replace-

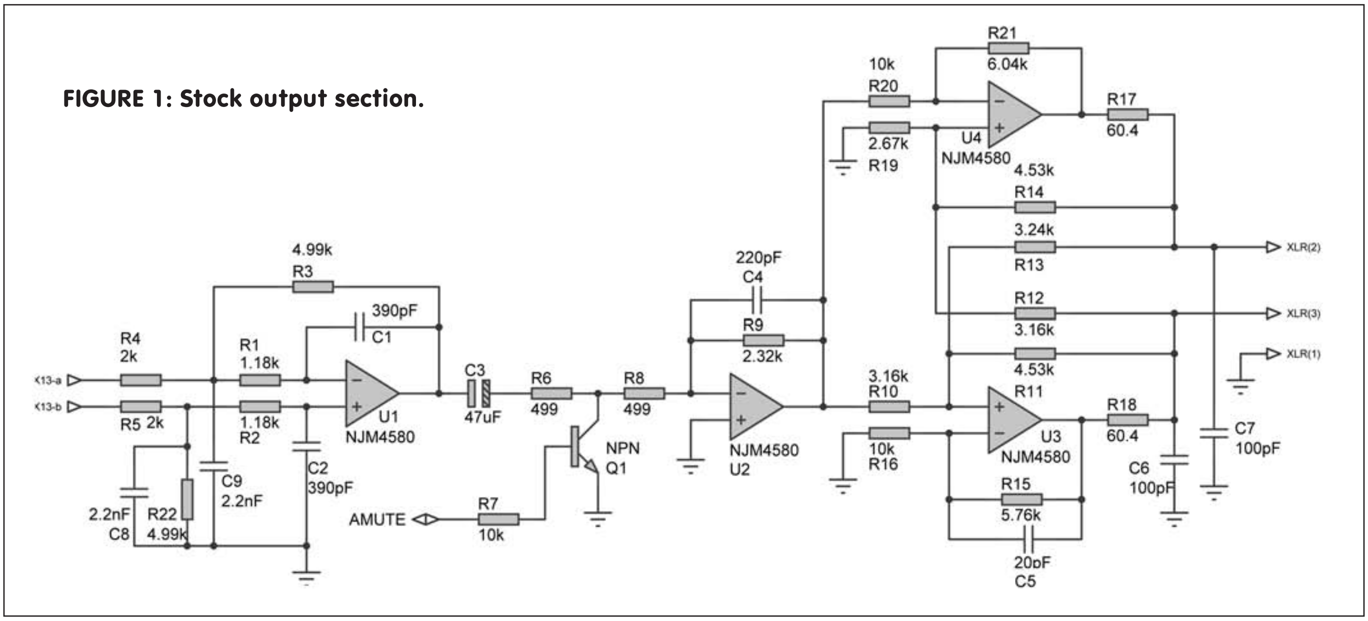


FIGURE 1: Stock output section.

ment board between the DAC outputs and the outside world. The original unit is fully balanced, and I have kept that, but I must confess that—until now—I use it single-ended.

A STACK OF OP AMPS

Figure 1 shows the output section of the stock DCX. This is repeated six times for each of six output channels. Take a walk through it. The DAC is a balanced voltage-output AK4393, and its outputs are connected to X13-a and X13-b, onward through a whole lot of active circuitry, to the balanced XLR output connectors on the rear panel [XLR (1), (2), and (3)]. X13 is an existing flatcable header from the DSP board to the output board, so you can see that by tapping this flatcable you can access the analog signal at will without any hardware changes.

Look at the detailed circuit. Op amp U1 is a second-order active filter, as is usual after a DAC, but it also amplifies the signal 2.5 times and converts the balanced signal to unbalanced or single-ended, if you will. Transistor Q1 is the mute transistor. If the signal AMUTE is high level, the transistor is fully conducting, thus presenting a very low impedance to the signal line and effectively shorting it to ground.

So far so good, but even when the signal is not muted, Q1 is still in the system. Its collector represents a nonlinear impedance and nonlinear capacitance, and together with R6 this will form a nonlinear signal dependent attenuator. I know from experience—for instance, in the early days—when this low-cost mute was often used with CD players, that this can cause distortion of the signal. So, that transistor needs to go!

By the way, that C3 is also not a good idea, but is required here. The DAC outputs the signals on a common-mode voltage of half its supply voltage, or 2.5V DC. Any small DC difference between the two inputs becomes amplified also, and could cause a serious offset further down the line, thus C3.

Next is U2, a buffer with some 2.3 gain, feeding the single-ended to balanced converter U3, U4, which also slightly attenuates the signal (the overall gain of this chain, by the way, is just above 10dB). U3 and U4 are cross-coupled. If you use the output single-ended and ground the unused pin, this will zero the cross-coupled feedback and increase the level of the used output by 6dB.

I wondered about the conversion: first- to single-ended and then again back to balanced. My guess was that it would be much more difficult to implement a mute on a balanced signal at 2.5V DC, and my later experiments confirmed it.

All in all, there is much gain in this chain, and the reason is that this is a pro unit, and pro signal levels are in practice 14dB above home hifi levels. It's a pain because it means you have less control range available on your volume control. So, for home use you really don't need all that gain.

I would like to keep the mute function and the filtering, of course. My new board modification therefore consists of a passive output network with the filtering, and the mute function with reed relays.

GOING PASSIVE

Figure 2 shows the diagram of one channel of the new output stage. The signal from the X13 header is connected to a first-order symmetrical passive filter (fancy name for R3, R4, and C1) through coupling caps. More on these caps later.

Then there is a second single-order filter implemented with R1, R2, and C2. RL1 is the mute relay, which shorts both signals to ground when not acti-

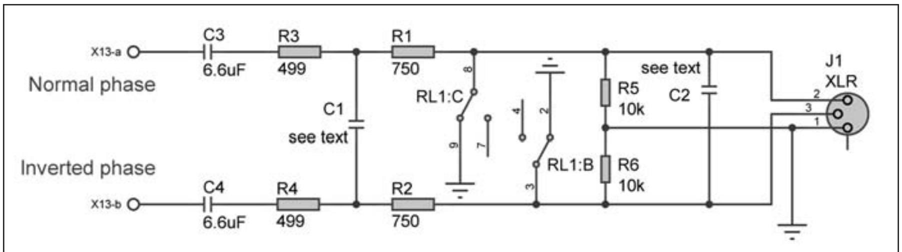


FIGURE 2: New output section.

vated. A resistor from each signal phase to ground, R5 and R6, provides a DC path to ground in case the equipment after the DCX is AC-coupled. They can also be used to attenuate the signal if needed.

The values of the filter resistors are chosen to present at least a resistance of more than 1k to each DAC output even if the output is shorted. You can load the DAC with 600Ω for AC, but the DC load should not be lower than 1k; I selected 950Ω total for a safety margin. Select the filter capacitors to get as close as possible to the second-order active filter shape in the original unit.

Figure 3 shows the frequency and phase response of the original version and my implementation (simulated), which is a tiny bit lower in level at 20kHz but has less phase shift. At any rate, I think the differences are too small to be of audible consequence. Finally, **Fig. 4** gives the full circuit.

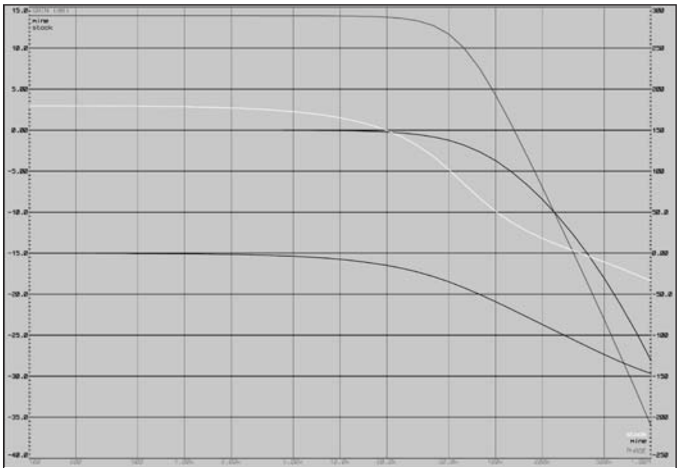


FIGURE 3: Frequency and phase response of the stock and the new output section (simulated).

THE MUTING

The muting circuit gave me some headaches. When you turn on the power to the unit, the AMUTE line (**Fig. 1**) is set high by the DSP and the signal is muted. After a few seconds, the mute is released and sound will be heard. (This is independent of the software mute available for each output which you can activate at the front panel.) In the stock unit, there are no audible clicks at switch-on or switch-off.

I had a choice: mute by shorting either the two output phases together or each phase to ground. I tried the first possibility because it needed only half a relay per channel, but I couldn't get rid of the switch-on noise. The mute signal from the DSP just came too late (it first needs to boot after power-on, I guess). Therefore, I used the relays in normal-on mode (**Fig. 4** again). Both signal lines are shorted to ground until the relays are activated.

Two conditions need to be met for this. First, C26 must be charged up through R26. This takes a second or so, and bridges the time between power-on and setting of AMUTE by the DSP. After the DSP has settled and releases the mute signal, the relays switch and sound is heard. Importantly, at switch-

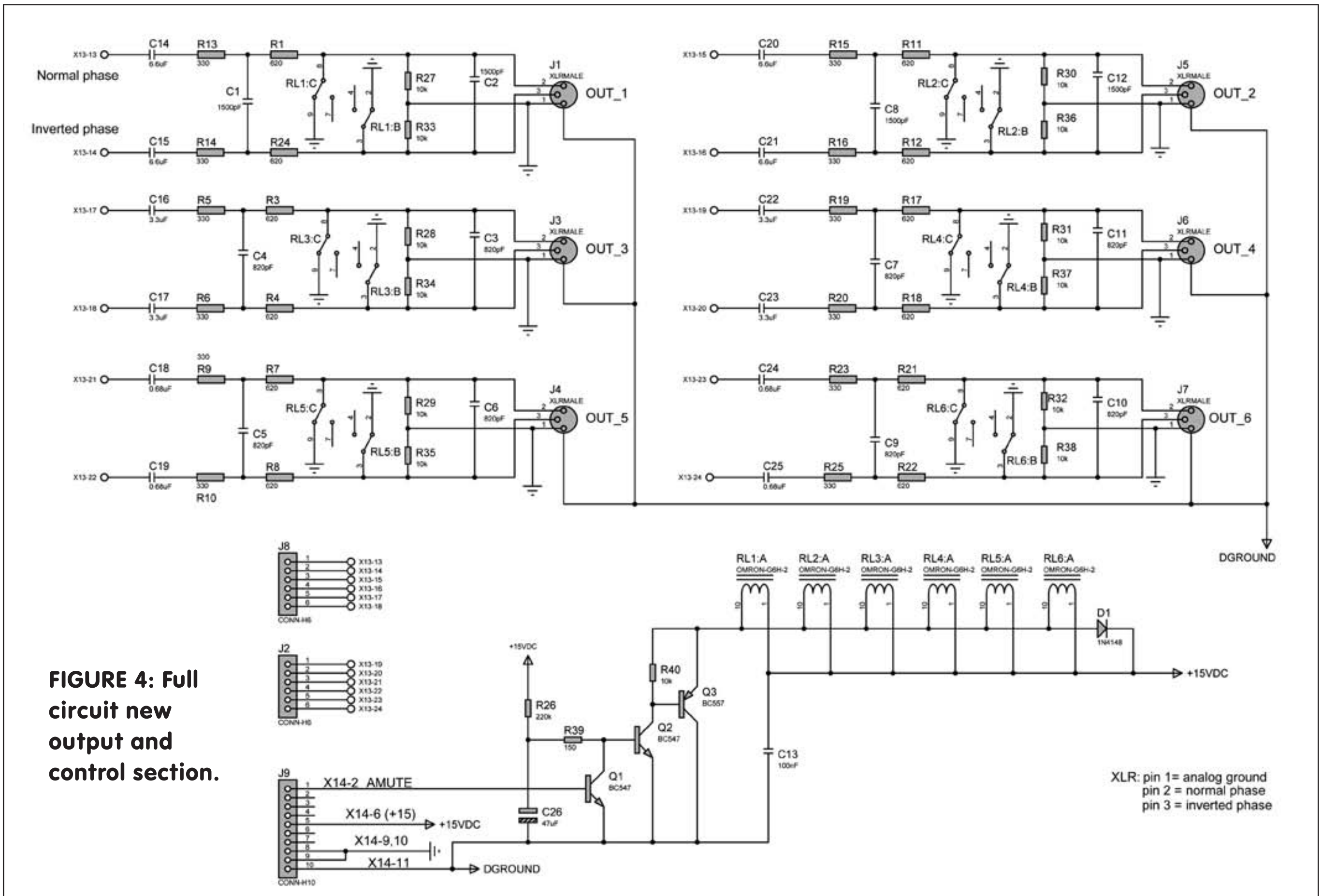


FIGURE 4: Full circuit new output and control section.

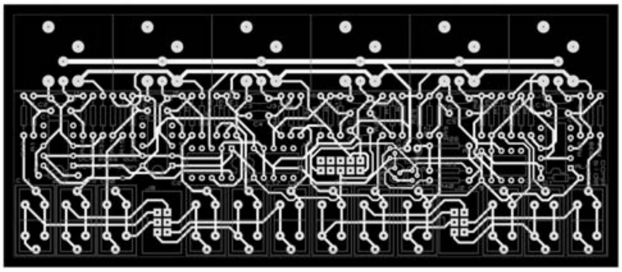


FIGURE 5: Output PCB.

can size the coupling caps with the expected frequency range in mind.

off, or if the power drops out, there is absolutely no noise at all.

BUILDING THE NEW PCB

The final PCB with stuffing guide is shown in **Fig. 5**. **Table 1** is a parts list. I was fortunate to locate (after a long search) the same exact XLR male output connectors Behringer uses, making for a true form-fit and function replacement board. There is only a handful of other passive components. Because you will most probably use this unit as an electronic crossover, you can size the coupling caps with the expected frequency range in mind.

Because it makes sense to use channels 1 and 2 for the low-frequency range, you need to use relatively large caps there (C14, C15, C20, C21). But as they will not be asked to carry a signal more than a few thousand hertz at the very most, they don't need to be exotic types. I used two parallel 3.3µF/50V polyester film types, which will give a low-frequency cutoff of about 5Hz with a 5k load. If the load is higher, of course, the lf –3dB point decreases. Similarly, I use larger passive filter caps (C1, C2 in **Fig. 2**) for the low-pass channels, 1500pF versus 820pF for the mid- and high channels. For the latter, rolloff should only start after 20kHz, but for the low-pass channel we don't care if rolloff starts at 10kHz or so, it only helps to limit hf noise.

On the third and fourth channels (mid frequency band), where I don't expect any signal below say 100Hz, I use a single cap. Finally, for the last two channels (high-frequency band), I used 0.68µF. Because these have a low value, using high quality types here doesn't cost you an arm and a leg. At any rate, I made the capacitor outline with multiple holes to accept several types and sizes of caps.

The resistors used for the filters are run-of-the-mill 2% metal film. Because the reed relay normally is not in the circuit, it also doesn't need to be exotic; however,

TABLE 1.

PART REFS	VALUE
Resistors (all 0.25W 1% metal film)	
R1, R3, R4, R7, R8, R11, R12, R17, R18, R21, R22, R24	620
R5, R6, R9, R10, R13-16, R19, R20, R23, R25	330
R26	220k
R27-R38, R40	10k
R39	150
Capacitors	
C1, C2, C8, C12	1500pF (mica or silver film)
C3-C7, C9-C11	820pF (mica or silver film)
C13	100nF/50V
C14, C15, C20, C21	6.6µF (2 parallel WIMA MKS2 pol. 3.3µF/50V DC Farnell # 107-423)
C16, C17, C22, C23	3.3µF (WIMA MKS2 pol. 3.3µF/50V DC Farnell # 107-423)
C18, C19, C24, C25	0.68µF (EPCOS BS32652 polyprop. 0.68µF/250V, Farnell # 400-3718)
C26	47µF/25 electrolytic
Transistors	
Q1, Q2	BC547 or equivalent small-signal NPN
Q3	BC557 or equivalent small-signal PNP
Diodes	
D1	1N4148
Miscellaneous	
J1, J3-J7	XLRMALE (Neutrik NC3MAH, Farnell # 724-543)
J2, J8	CONN-H6 (flatcable conn 2 × 3 pins 0.1″ grid)
J9	CONN-H10 (flatcable conn 2 × 5 pins 0.1″ grid)
X13	header 26 pin (flatcable conn 2 × 13 pins 0.1″ grid)
X14	header 20 pin (flatcable conn 2 × 10 pins 0.1″ grid)
RL1-RL6	OMRON-G6H-2 or equivalent (see text)
20″ flatcable 26-way 0.05″ spacing	

the board is laid out for Omron type G6H-2-DC12 because I had a box full of those (thanks, Leo). But, since most reed manufacturers use similar layouts, you can probably use other types, too. Just make sure they are dual normally-on 12V types and don't consume too much current (12mA each for the Omrons).

GETTING CONNECTED

The new PCB is connected to the DSP board using existing headers. There are two headers on the DSP board—X1 for the input and output signal lines and X2 for the RS232, midi, and power supply lines. You will need to make a new set of flat cables with connectors to hook up the new PCB to the DSP PCB. The new cables from the DSP board are split and fixed to several connectors to join the new output board and the new serial interface board.

The cable plan is shown in **Fig. 6**, while **Photos 1** and **2** show the cable run in the modified unit. **Photo 3** shows the cable set you need to build. Not all wires are used; for instance, there is no need to run all supply lines to the new board (except +15V for the reed relays), and the analog input lines are also not used (but see next section). So you should be careful with the cable alignment in the headers, because in some cases the flat cable has fewer wires than the header pins and the cable needs to be aligned left or right. **Figure 6** and **Photo 2** should make that clear.

Double-check the wire numbers before fitting the connectors. It is not difficult to do, but it's a pain to redo if you make an error. Both my units ran fine on the first try.

FINAL CONSTRUCTION

Mounting the completed board in the unit is straightforward, once you remove the old board (at which point you void the guarantee, I guess). Remove the existing flat cables and all the screws in the various plugs and sockets on the rear panel. You can then take out the complete I/O board. The new board is fixed with the original screws used to hold the XLR output connectors.

I taped a plastic sheet to the bottom of the enclosure to avoid any contact of the solder joints with the metal as the distance is quite short. The small serial port board also uses the original screws that held the RS232 socket at the rear panel. Next put in the new cables—first the analog one, then the one for the control—and serial I/O lines.

WHAT DO I LOSE?

My modification replaces the entire original back panel circuit board, which includes several sections. One section containing the six analog output channels is replaced by my analog passive output PCB. There are two control input sections.

One is for a serial connection to a PC, which allows you to control the DCX remotely, to store and save different setups on your PC and to download firmware upgrades. This functionality is maintained via a small additional PCB

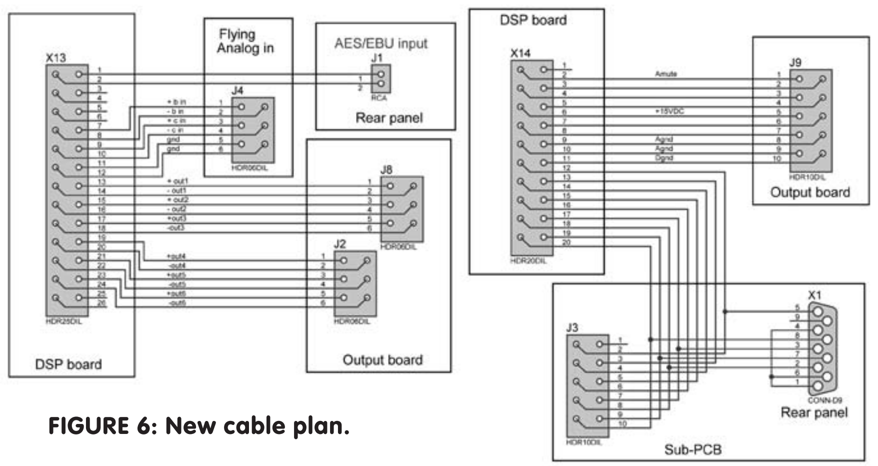


FIGURE 6: New cable plan.

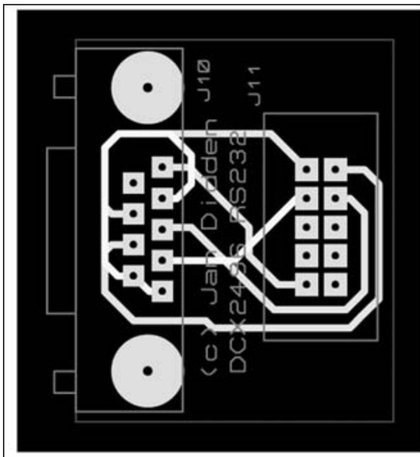


FIGURE 7: Serial port sub-PCB.

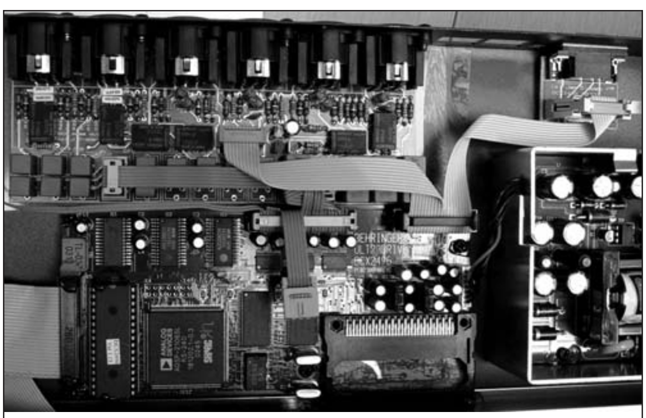


PHOTO 1: Cable layout with new board installed.

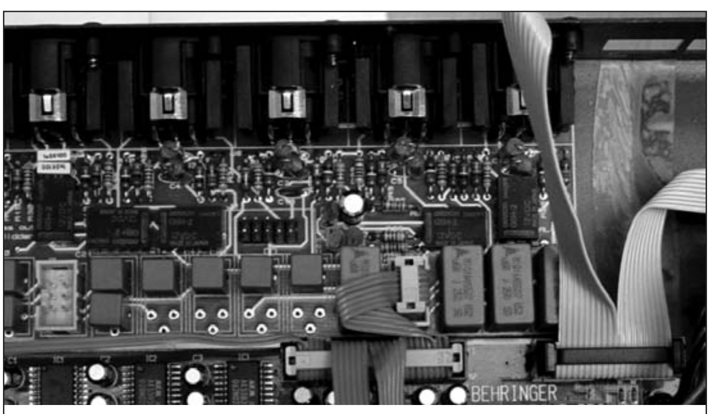


PHOTO 2: Cable layout detail.

(**Fig. 7**). The other is a jack to connect multiple DCX units in series so that they can be controlled by a single PC (this is pro stuff, remember!). That functionality is lost but would most probably not be used in home setups anyway.

The original board also contains analog and digital (AES/EBU) inputs. In fact, in the stock unit the analog and digital inputs share female XLR jacks and are selected from input channel A via a normal low-level relay. Don't ask me what that does to the digital signal integrity!

Because I use the DCX only with digital inputs, I didn't replace the analog input section. Just to keep my (and your) options open and also to have an analog input channel for measurements, I put a flying header on the B- and C- analog inputs (**Photos 1** and **2**). If you want to use the DCX with a high-quality analog source such as a turntable, I recommend a Behringer SRC2496, which is a flexible unit that does all kinds and combinations of A/D and D/A and sample rate conversion and costs less than a DCX.

Currently, I use a WBT RCA jack in one of the unused XLR input mounting holes to connect the digital input (pin X13(1) to RCA center pin, X13(2) to RCA ground pin; see **Fig. 6**). As mentioned before, you also lose about 10dB of excess gain, so this is a good thing for home use.

The next step up is a new digital input PCB to improve cable matching and jitter performance. I am not sure, though, that it can be done without hardware changes to the DSP board, which I would like to avoid. But stay tuned.

PHOTO 3: New cable set.

WRAPPING IT UP

This is a relatively simple project with a great reward. The mod is fully reversible, and not having to touch the hardware minimizes the risk that you damage anything on the DSP, which would make the unit worthless. Best of all, the sound definitely improved after removing all those op amps. The levels are better matched to a home hifi system as well.

You have several options to do this mod. If you have power amps or a preamp that is AC coupled (has an input coupling cap), you can delete the coupling caps on this board (C1... C12), saving considerable expense. In that case, make sure you jumper each capacitor. There is, however, one caveat to this: depending on the exact pre/power amp input circuit, most probably the mute will no longer function correctly when you use DC coupling. In that case, you might as well delete the mute option, saving the expenses for the six relays and associated components (Q1 . . . 3, D1, R26, R40, C13, C26). If you do delete the mute option, make sure you always switch on the DCX before the power amps and switch it off after the power amps, or, as I do, just leave it on always. (You could, of course, delete the mute option anyway, independent of whether you use AC or DC coupling). Each of the deletions mentioned probably saves some 30% of the cost, depending where you buy, and has, of course, no impact on the final sound quality. *aX*

RESOURCES

You can obtain the PCBs from Roger Pilgham Audio (www.roger-pilgham.nl). And the first one who identifies who Roger Pilgham was gets one set for free! The resistors and PCB headers and connectors are quite standard. The XLR connectors are critical for a correct fit on the replacement board as well as in the chassis, so I recommend using the types indicated in the text. The capacitors are a tight fit, but you may have adequate replacement types. If not, use the indicated Farnell part numbers (www.farnellinone.com).

Back

Equalize The Bass Response Of Your Room

Discover how equalization can affect low-frequency reproduction in a room.

By George Danavaras

I spent a lot of time during the last years studying the influence of the listening room to the reproduction of the low frequencies. I read several articles and performed many measurements in my listening room, trying first to understand this influence and then to eliminate it.

This article describes the results of this study and also presents a general method to equalize the low-frequency response of a room using a simple analog electronic equalizer with notch filters. I must admit that these days digital technology could provide more flexible solutions for the bass equalization of a room. But before you decide to follow the digital way, you can try this method, which is simple and cheap—yet effective—and offers many benefits.

BASS REPRODUCTION

It is true that in most rooms the quality of the bass reproduction is determined more by the room than the loudspeaker itself. There are many papers that examine the room's influence on the reproduction of low frequencies, particularly articles by Floyd E. Toole[1,2,3].

Usually, the following three categories affect the room's influence:

1. The dimensions of the room
2. The position of the loudspeaker and the listener in the room
3. The sound absorption and reflection of the surfaces of the room

There is not much you can do about the first point except if you're currently building the room. For the second and third point, you should follow the general rules as given[1], although due to the long wavelengths, sound absorption cannot help much in the low-frequency range.

Of course, it is not always easy to move the loudspeakers or the listening position to the points that have less influence on the reproduction. Sometimes it is just not possible. For these cases the electronic equalization becomes a very good solution.

WHAT TO EQUALIZE

The most important issue when you want to equalize the low-frequency reproduction of a room is to know what exactly should be equalized. There is much discussion about this matter, but here is, in summary, my personal recommendation about this subject:

1. I strongly recommend the equalization of the low frequencies (below 150 to 200Hz). When applied properly, this can help dramatically the clarity of the reproduction.

2. Only a small area of the room, which is usually chosen around the listening position, can be properly equalized. The equalization in the response in one specific place usually makes the response in the other places of the room worse. I concluded this only after many tries to equalize my listening room using simple analog equalization.

3. You should correct only the large peaks of the response. Avoid the equalization of the notches that exist in the room response. For the equalization of the peaks use simple analog notch filters.

FIGURE 2: The response of the notch filter.

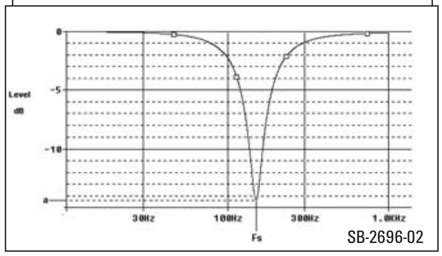


FIGURE 3: The block diagram of the equalizer.

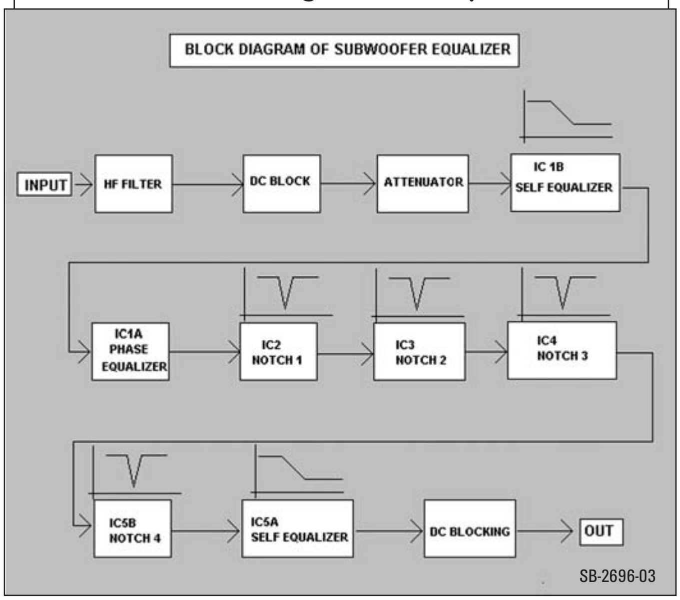
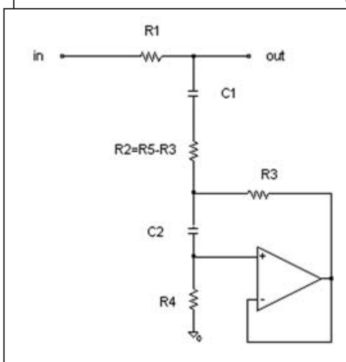


FIGURE 1: The notch filter topology and definitions.



$$Fs = \frac{1}{2 \cdot \pi \cdot \sqrt{C2 \cdot R3 \cdot (R4 - R3) \cdot C1}}$$

$$Q = \frac{1}{2 \cdot \pi \cdot Fs \cdot C1 \cdot R5}$$

$$a = 20 \cdot \log \frac{R2 + R3}{R1 + R2 + R3}$$

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The method that I use to realize the above is as follows:

a. First I take a number of measurements in a small area around the listening position (for example, 20 to 30cm in front, rear, up, and down directions).

b. Then I look to the energy spectral decay plots of these measurements in order to identify all the frequencies that have a very long decay, which exists in the measurements of all positions. This is an easy way to identify the real resonance frequencies of the room that should be equalized and to avoid any other acoustical interference that depends on the position of the microphone.

c. Then I compute the average low-frequency amplitude response of all the places that the measurements were taken, in order to decide about the attenuation and the quality factor Q of the notch filter that I will use for the equalization.

d. The components of the notch filters are computed and the equalizer is constructed.

e. Then I repeat all the above measurements to see whether there is a need to make any corrective action and to confirm the benefits of the equalization.

THE NOTCH FILTER

The basic part of the equalizer is the notch filter. For the design of this notch filter I used the method described in Siegfried Linkwitz' site (www.linkwitzlab.com).

The topology of a notch filter with the relevant definitions for the center frequency (Fs), quality factor (Q), and the attenuation (a) is shown in **Fig. 1**. The components R3, R4, and C2 around the op amp form an electronic inductor, which resonates with the capacitor C1, and the typical notch filter response is produced as given in **Fig. 2**. The required parameters for the design of the notch filter are:

Fs = the notch frequency
Q = the width of the notch
a = the attenuation of the filter at the notch frequency

The steps to design this notch filter (see also **Fig. 1**) include:

1. Define the requested Fs, Q, and attenuation (a) of the filter.
2. Select R1.
3. Then $R5 = R1 \times a/(1-a)$ and $C1 = 1/(Q \times R5 \times 2 \times pi \times Fs)$, where pi = 3.14159.
4. Select $R4 > 20 \times Q^2 \times R5$. Usually you would prefer an even greater value for R4.
5. Select R3 <= R5.
6. Then $C2 = (Q \times R5)/(2 \times pi \times Fs \times R3 \times (R4 - R3))$.

For example, consider the following parameters to compute a notch filter:

Fs = 150Hz
Q = 8
a = -15dB = 0.1778

Then, according to the previous method, the steps to follow include:

1. Select R1 = 2kΩ.
2. Then $R5 = R1 \times a/(1 - a) = 432.5\Omega$.
3. And $C1 = 1/(Q \times R5 \times 2 \times pi \times Fs) = 306.7nF$.
4. Select $R4 > 20 \times Q^2 \times R5 = 554k\Omega$. Let R4 = 560kΩ.
5. Select R3 <= R5. Let R3 = 390Ω.
6. Then $R2 = R5 - R3 = 42.5\Omega$.
7. $C2 = (Q \times R5)/(2 \times pi \times Fs \times R3 \times (R4 - R3)) = 16.8nF$

Usually standard values for the components are preferred, so it is necessary to modify the above values and to re-compute the filter. For this reason, I wrote an Excel sheet that you can download from the *audioXpress* website (www.audioXpress.com).

After the download, when you open the sheet with the Microsoft Excel program, you will see two areas. On the top area you can insert the required filter values, and the program will compute the proposed network components.

You can insert the real values of the components on the bottom area, and the Excel program will compute the filter parameters. By changing the values of the components, you will realize the notch filter using standard components values.

THE ELECTRONIC EQUALIZER

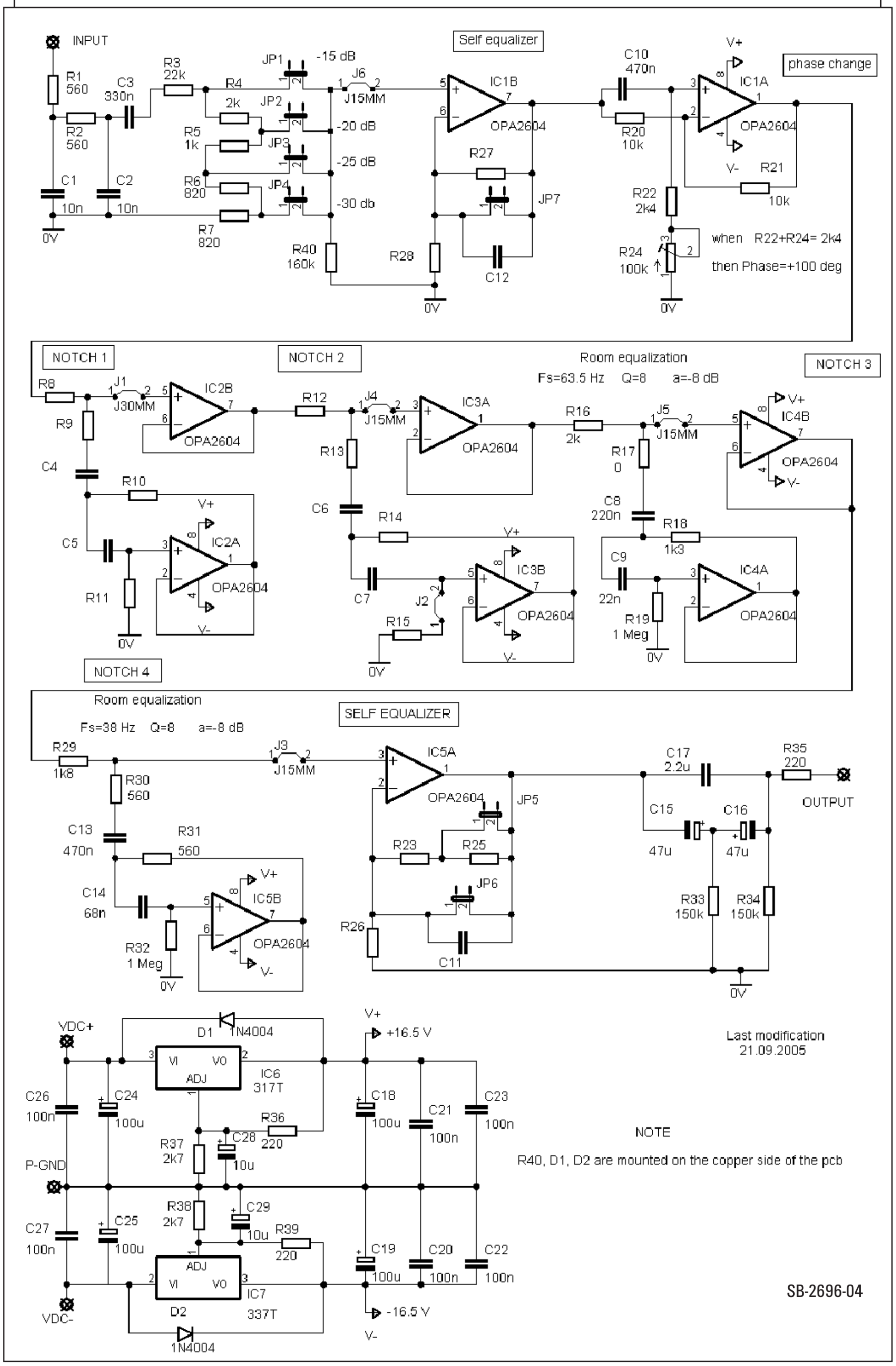
The block diagram of the equalizer is shown in **Fig. 3** and the full electronic diagram in **Fig. 4**. At the input of the equalizer, the components R1, R2, C1, C2 form a low-pass filter for the filtering of the high frequencies above the needed bandwidth. The capacitor C3 blocks any DC level and forms a high-pass filter at about 1Hz.

The components R3, R4, R5, R6, R7, and R40 form a voltage attenuator. By using the jumpers JP1, JP2, JP3, and JP4, you can attenuate the input level from -15dB to –30dB in steps of –5dB. This is very useful for adjusting the level of the subwoofer to the rest of the system.

The components R27, R28, C12 around IC1B form a buffer for the previous stage and optionally a self-filter, which can boost an area of low frequencies. This kind of filter is necessary to correct the subwoofer response in a range of frequencies. If you don't need this filter, then you can set the jumper JP7 to the ON position and the stage becomes a simple buffer.

The components C10, R20, R22, R21, and the potentiometer R24 around IC1A form an all-pass circuit, which

FIGURE 4: The electronic circuit of the equalizer.



you can use to change the phase of the subwoofer signal. When R24 is set to its maximum value of 100kΩ, the phase is zero; set to its minimum value, the phase change is about +100°.

The main speakers interact with the subwoofer near the crossover frequency (because both are playing the same signal) and cancellation can occur if the difference in the phase of the two signals is around 180°. The change of the subwoofer phase can help so that a smooth integration of the subwoofer response to the rest of the system will happen.

The first notch filter is formed with R8, R9, C4, C5, R10, R11, and IC2A. The IC2B is used to buffer the first notch filter and to drive the second notch filter with the components R12, R13, R14, R15, C6, C7, and IC3B. The IC3A buffers the second notch filter and drives the third notch filter with the components R16, R17, R18, R19, C8, C9, and IC4A. The IC4B is used to buffer the third notch filter and to drive the fourth notch filter with the components R29, R30, R31, R32, C13, C14, and IC5B.

The IC5A is used to buffer the fourth notch filter and also forms a second self-filter as described above. The components C15, C16, C17, R33, and R34 form a DC blocking stage, while resistor R35 buffers the equalizer from the capacitance of the next stage. For the op amps, I used the OPA2604 from Burr-Brown.

The regulated power supply of the circuit is based on the LM317 and LM337 regulators. A small heatsink is necessary for each regulator.

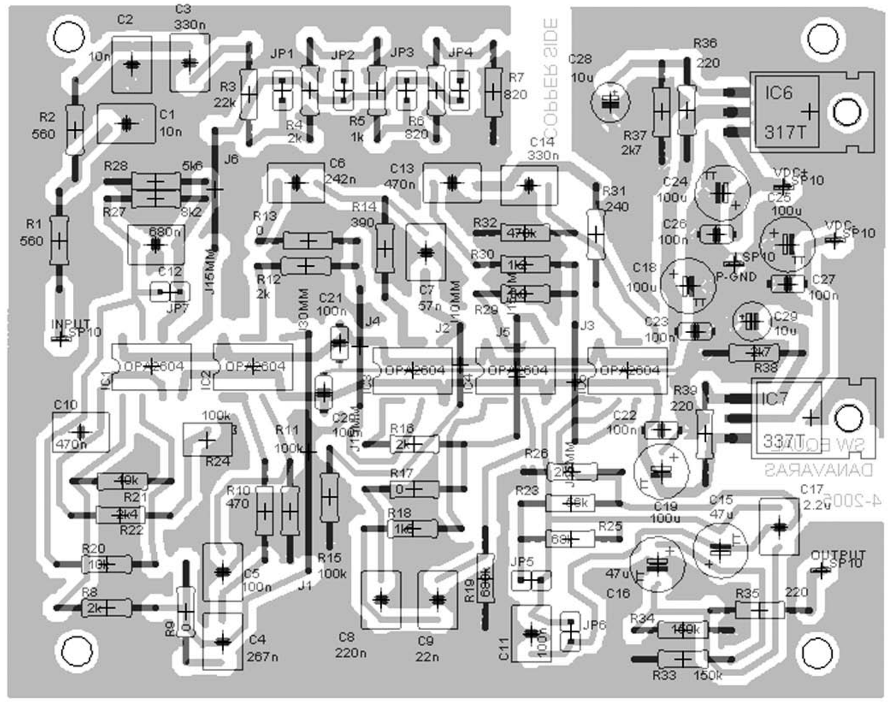
The DC unregulated voltage should be provided from an external power supply. You can use a common power supply, provided that it has an output of about ±24V DC at 500mA with low ripple.

The capacitors C24-C27 filter the input voltage, while the capacitors C18-C23 filter the output voltage. The resistors R36-R37 and R38-R39 set the output voltage at 16.5V. Diodes D1 and D2 protect the regulators during input voltage interruptions.

I also designed a single-layer 112 × 89mm PCB for the equalizer using the demo version of the EAGLE software package (www.cadsoft.de). The Gerber file of this PCB is also available for download on the *aX* website. The file SW_Processor.zip contains the compressed file SW_Processor.bot, which is the copper side of the PCB.

After downloading the file, you can use a Gerber file viewer (like View-

FIGURE 5: The layout of the components on the PCB.



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mate, which is available free from www.pentalogix.com) to print the PCB on transparent paper and manufacture your own PCB. The placement of the components on the PCB is shown in **Fig. 5**, while **Photo 1** shows the assembled PCB of the equalizer.

Please note the following two important notes when you assembly the PCB:

1. The components R40, D1, and D2 are placed on the copper side of the PCB.

2. There are two separate grounds on the PCB—one for the power supply and one for the analog filters. This flexibility could solve any existing ground-loop problems. Usually, the two grounds should be connected together either at the star ground point or directly on the PCB. Remember to connect the two grounds together; otherwise, the circuit will not work properly.

A TYPICAL EXAMPLE

I performed the following procedures for the equalization of my listening room, which measures approximately 4.5 × 10m with a height of about 2.8m. I started by taking several measurements of the impulse response of the room, with the microphone for these measurements positioned about 30cm around the ear height of the listener.

The topside of **Figs. 6–9** show the energy spectral decay as measured at four different microphone positions around the listening point. From these measurements, notice the long decays at the frequencies of 38Hz, 61.5Hz, and 66Hz that exist in all four positions. From this you can conclude that these are the main room modes that exist below 100Hz.

There are also long decays in other frequencies, but it is clear that these strongly depend on the position of the microphone, as you can see if you compare them with the measurements in the other positions. The frequencies of 61.5 and 66Hz are very near; so I decided to choose the mean value of 63.5Hz as the center frequency of the notch filter.

I determined the attenuation and the bandwidth of the notch filter at the above two center frequencies as follows:

Figure 10 is the average low-frequency ⅓ octave frequency response of the microphones for all positions. From this curve it seems that the attenuation should be about 8dB for both notch filters. I chose a value of about 8 for the quality factor Q for the notches. This is a good compromise between the sharpness of the room mode (which always demands a high Q filter) and the realization of the electronic circuit (which means that the practical realization of a high Q notch filter using electronic components is not an easy task).

So here are the two notch filters that are needed for the equalization of my room:

FIGURE 6: Energy spectral decay in position 1 without and with equalization.

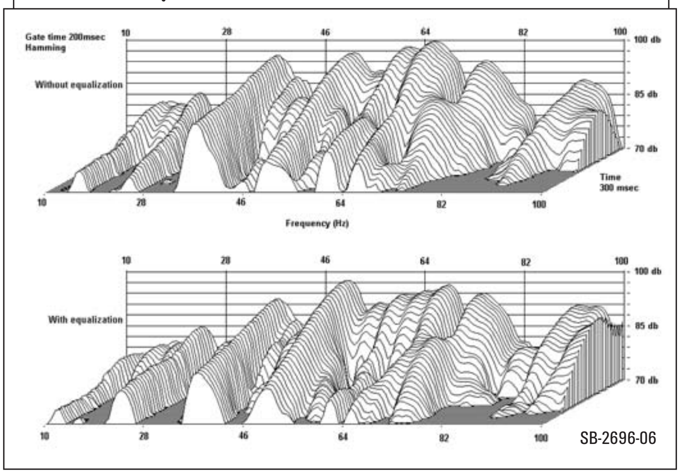


FIGURE 7: Energy spectral decay in position 2 without and with equalization.

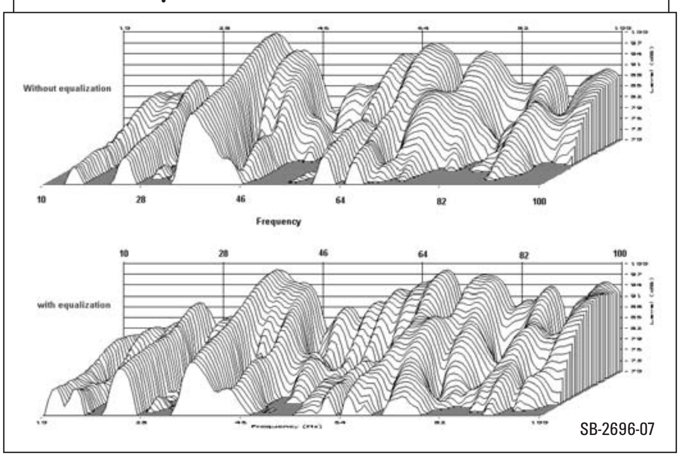


FIGURE 8: Energy spectral decay in position 3 without and with equalization.

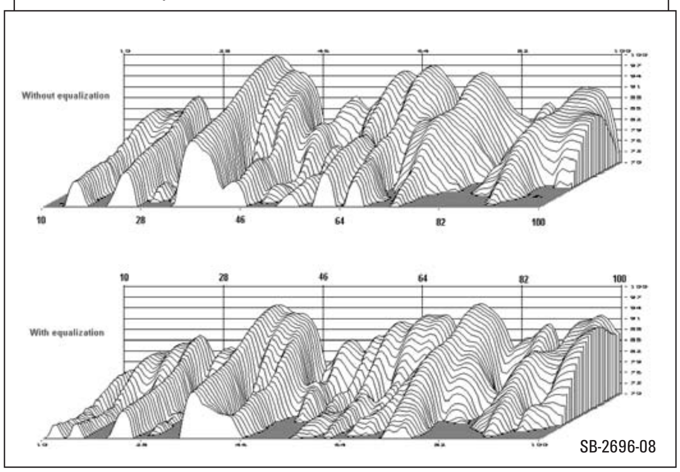
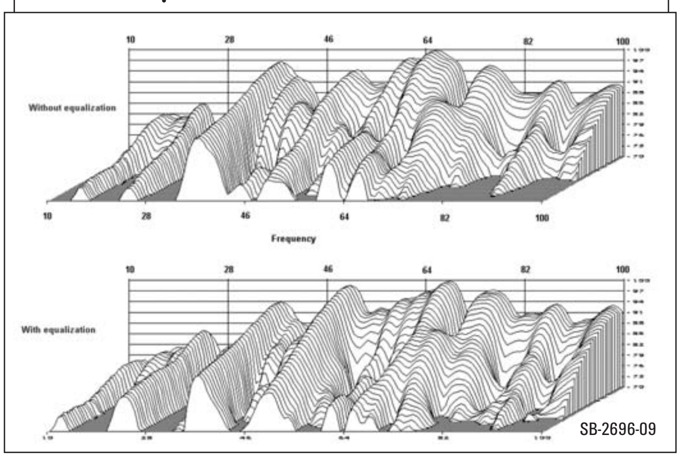


FIGURE 9: Energy spectral decay in position 4 without and with equalization.



a) Central frequency Fc = 38Hz, Q = 8, and attenuation about –8dB

b) Central frequency Fc = 63.5Hz, Q = 8, and attenuation about –8dB.

I computed the components of the filters using the method described previously, and their values are shown in the schematic of **Fig. 4** as Notch 3 and Notch 4.

After the construction and the testing of the equalizer, I again measured the energy spectral decay on the same positions as before. The results of these measurements are shown for comparison on the bottom sides of **Figs. 6–9**. From these figures you can easily see the improvement on the energy spectral decay with the equalization. The decays on the frequencies of 38 and 61.5 and 66Hz are much shorter now.

Also, the ⅓ octave average frequency response is much smoother as shown in **Fig. 11**. Comparing the responses with and without equalization, you can see that the variation in the response drops from about ±6dB without equalization to about ±3dB with equalization. This is a clear improvement.

The impulse response is also shown for both cases in **Fig. 12**. The topside is the impulse response in the listening position without equalization, and the bottom side is the impulse response at the same position with the equalization. It is easy to notice that the equalization controls much better the damping of the response.

All of this proves that the applied electronic equalization to the room response was effective. This was also confirmed by the listening tests. The improvement is dramatic. The bass reproduction is much more tight and articulate with the equalization. *aX*

FIGURE 10: Average frequency response without equalization.

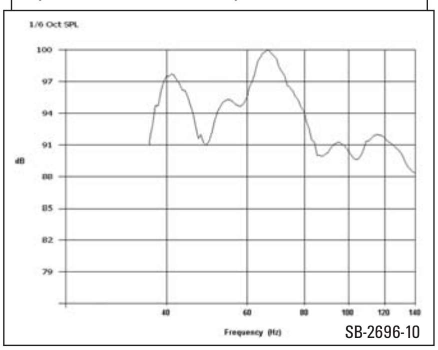


FIGURE 11: Average frequency response with equalization.

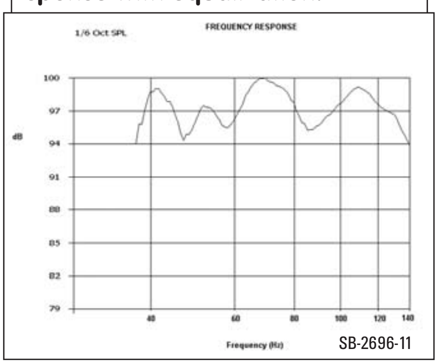
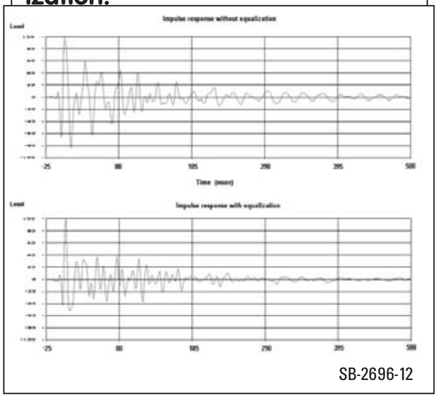


FIGURE 12: Comparison of impulse responses at the same position with and without equalization.



George Danavaras graduated from National Technical University of Athens, Greece, in 1986 with a degree in Electronics Engineering. He currently works in the R & D division for a Greek telecommunication company. His hobbies include design and manufacturing of audio crossovers, amplifiers, and loudspeakers.

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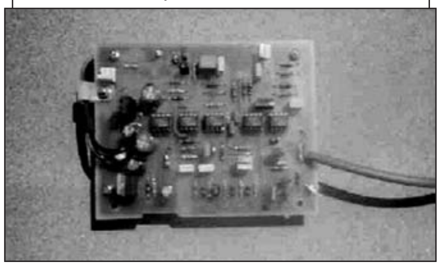
PHOTO 1: Equalizer PCB.



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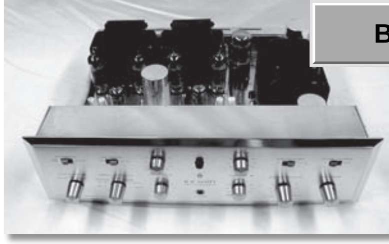


Renovating the Scott 222C Part 4: Listening & Measurements

By Charles Hansen

I connected my sources to the modified unit using the same tubes as in the original listening session, with my NHT satellites plugged into the 4Ω and then the 8Ω taps. The first listening session used the Direct input to the power amplifier (the "fourth" CW position on the input selector).

There was still a noticeable hum with my ear against the loudspeaker LF drivers that did not change with the loudness control setting. It was more pronounced in channel A (left), but was fortunately not noticeable from my listening position. The channel A power amplifier tubes, filter capacitor, and output transformer are farther from the power transformer than channel B, so any magnetic field pickup would seem more likely to affect channel B than channel A. Even the on-off power switch is closer to the channel B section of the loudness control.

Because the closest matched tubes were in channel B, I decided to swap the output tubes from left to right to see whether the higher hum level followed the unmatched tubes. Once I reset the bias for each tube, I again checked the hum level, which had not changed one iota. With no real objective reason to do so, I then swapped the 6U8 tubes with equally unsuccessful results—mystery still not solved!

I connected the CD player to the new Direct input and let the 222M run in for ½ hour, then began my listening session. I needed about 3 o'clock on the loudness control for a comfortable listening level. The Direct input is the purest configuration, without the steep high-pass input filter, the added preamp tubes, tone controls, and other response-altering circuitry.

I found the bass to be a bit weak, but with good definition, especially with an acoustic bass. The midrange was nicely presented while the highs were a bit rolled off. The soundstage was wide and the amplifier didn't lose definition with complex orchestration or massed chorals.

With only 13dB of gain available, I had to turn the loudness control almost all the way up for a comfortable listening level. I tried the 4Ω tap, which gave

FIGURE 42: Modified unit AC line inrush current.

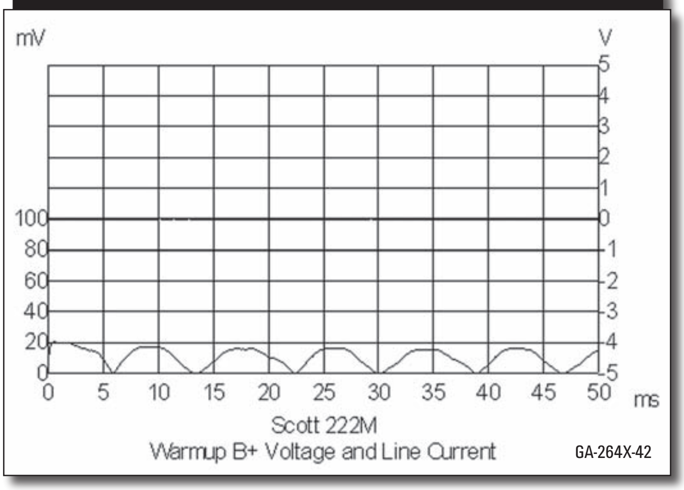


FIGURE 43: RIAA equalization response error.

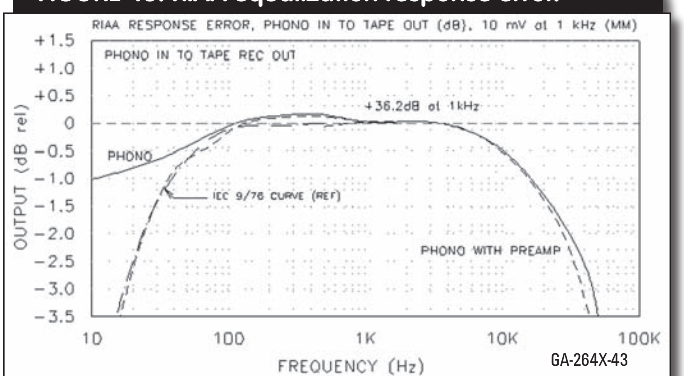


FIGURE 44: Phono section THD vs. frequency.

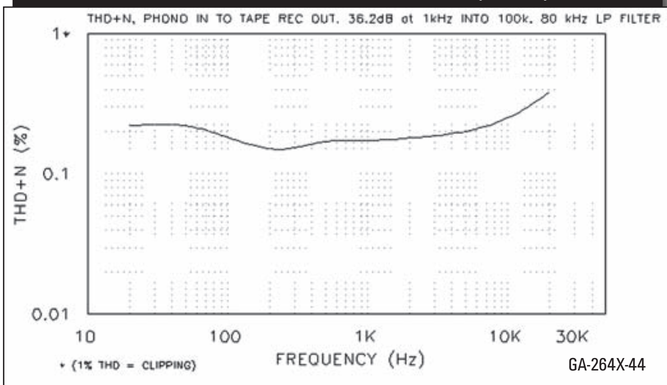
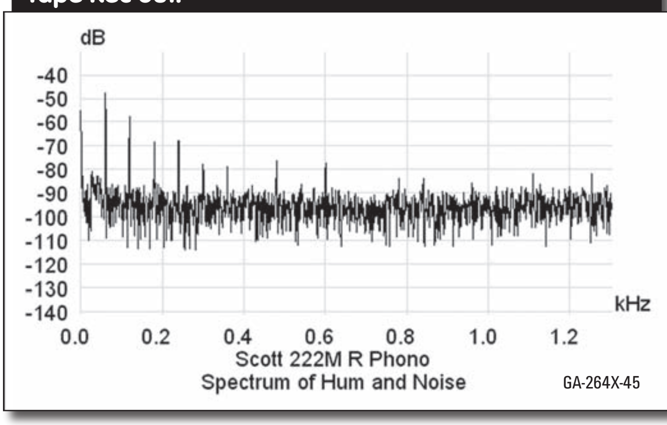


FIGURE 45: Spectrum of hum and noise, phono In to Tape Rec out.



a bit more volume with the 8Ω NHT SuperOnes, but the high frequencies were even more rolled off and took on a bit of edginess. Next, I connected the Center Channel output to the input of my subwoofer amplifier.

Fortunately, the hum was no more noticeable than that from the NHT satellites, so the Center Channel output was quite useful for this purpose. You may need to adjust the value of resistor R13 to provide a center channel signal level that is compatible with your own powered sub. The subwoofer level nicely tracked changes in the loudness control setting, although it was a bit overpowering at low levels with the Loudness switch turned on.

Next I tried the Extra input, which adds the preamp section with its filters and tone controls. With 26dB of available gain, I needed only half the loudness control setting for the same acoustic level as the Direct input. This was fortunate because the two stages of preamp gain add more hum that varies with the loudness control. The deepest bass was noticeably weaker than the Direct input, no doubt caused by the steep LF input filter. The midrange and highs were quite nice, with only a bit less soundstage width. I could bring the highs back up to a satisfying level with just a slight adjustment to the treble controls.

With the subwoofer connected to the Center Channel again, the bass was acceptable down to the open-string E_3 (41Hz), but lacked bass slam with a 5-string electric bass, whose A_3 reaches down to 22.5Hz. The preamp HP filter actually accentuates the 2nd harmonic, which is already dominant over the fundamental on the electric bass.[10] This also made some well-recorded acoustic bass music sound even richer.

My final test was to connect my turntable to the Phono input. There was a noticeable low-frequency thump when I selected Phono on the input selector. The hum level was no higher than that of the Extra input with the loudness control set for the same output level at 1kHz, but there was noticeably higher hiss. I replaced the phono preamp tubes with Mullard ECC83s, with only a marginal reduction in the hiss level. Adding tube shields did not help (which is fine with me since I suspect they shorten tube life).

I played a mix from my collection of jazz and classical LPs. Even with the softest sections of classical chorals and orchestration, hum and hiss was not really objectionable from my listening position. The phono section provided a deep and wide sound stage with bass performance equivalent to the Extra input.

The cannons in the finale of Tchaikovsky's *1812 Overture* didn't suffer too much from the HP filter and all those coupling capacitors, and the hum was much lower than with the 222C as I received it. I was quite impressed, in

FIGURE 46: Extra input frequency response.

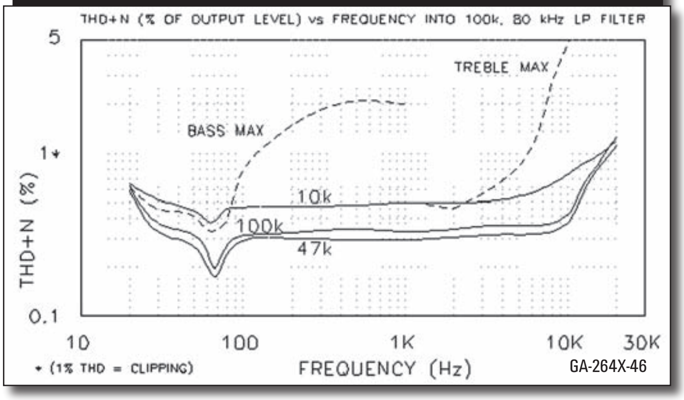
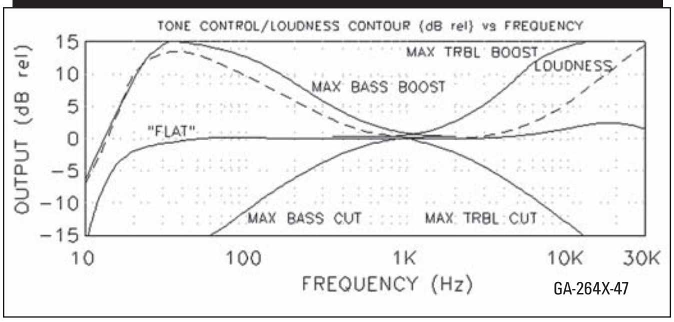


FIGURE 47: Tone control modifications to frequency response.



fact, with the midrange and low treble performance. This is a very usable phono stage.

The preamp section, with its hum, noise, and limited bass, is still the weak link in this project. The power amplifier and phono sections are much improved over the original, but still not in a class with the modern tube amplifiers I have auditioned and tested.

MEASUREMENTS—MODIFIED UNIT

Figure 42 shows the modified amplifier AC line inrush current recorded with the 100:1 ratio current transformer. The 1Ω CT secondary burden is again processed by the precision full-wave rectifier op-amp circuit, with a scaling factor of 10mV/amp.

The initial magnetizing inrush (first half-cycle) is now only 2.1A peak. I took several traces with a cool-down period between each one, and the peak current never varied. The inrush current half-cycles are also uniform, without the 3rd harmonic content or asymmetry from operation on a minor hysteresis loop that would suggest a DC component was impressed on the transformer.

The Phono input impedances were both 47k5. The phono stage output impedances measured at the Tape Rec output jacks had decreased to 7k3 (A) and 7k7 (B) at 1kHz, including series 2k80

FIGURE 48: Extra input/preamp THD+N vs. frequency.

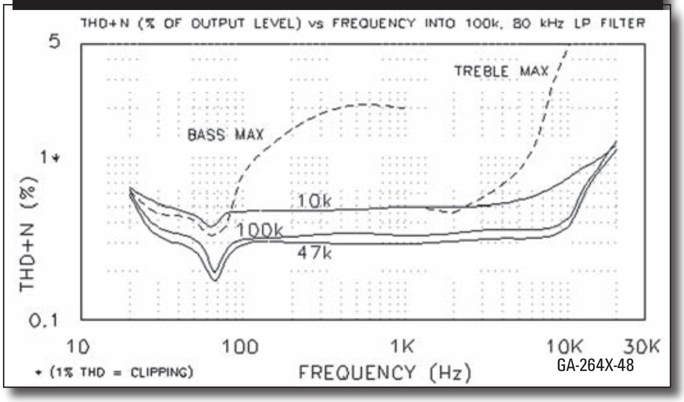


FIGURE 49: Extra input/preamp THD+N vs. output voltage.

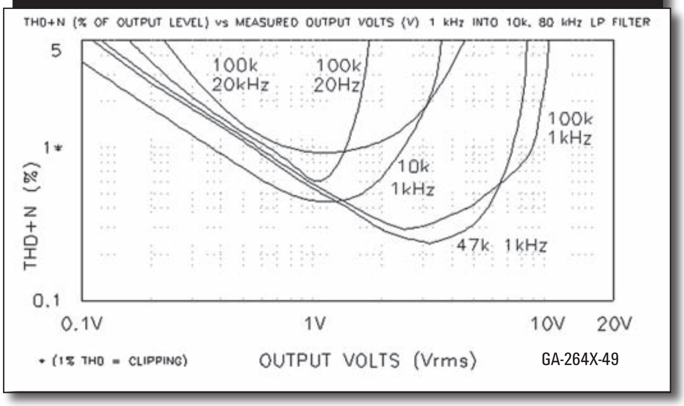


FIGURE 50: Extra input residual distortion, 1kHz 2V 100k.

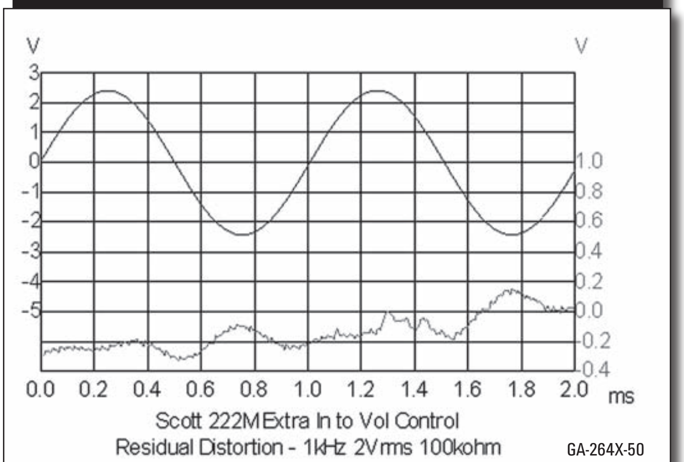


FIGURE 51: Extra input spectrum of 50Hz, 2V 100k.

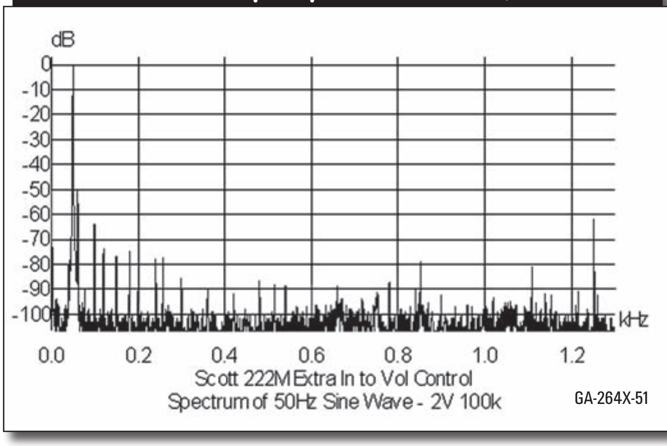


FIGURE 52: Extra input spectrum of 1kHz, 2V 100k.

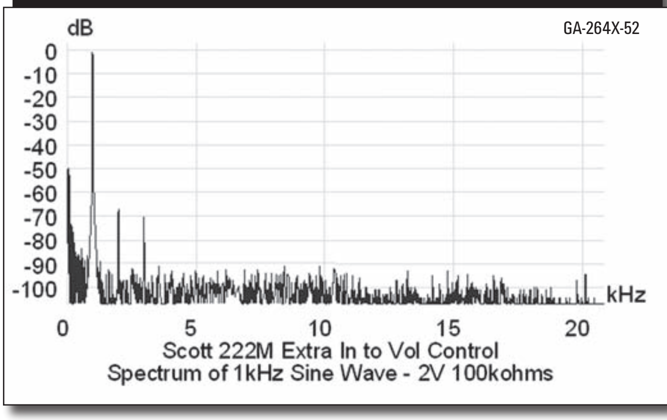


FIGURE 53: Extra input spectrum of 19kHz + 20kHz intermodulation signal.

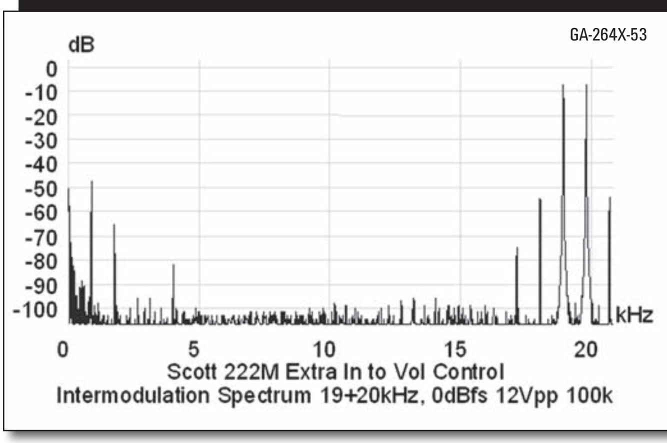


FIGURE 54: Extra input spectrum of 9kHz + 10.05kHz + 20kHz intermodulation signal.

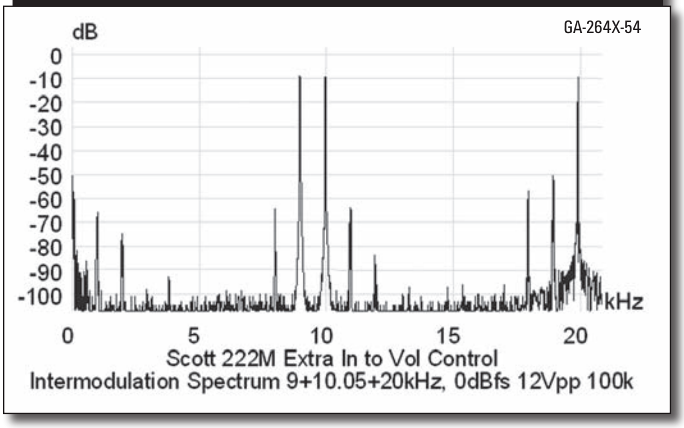
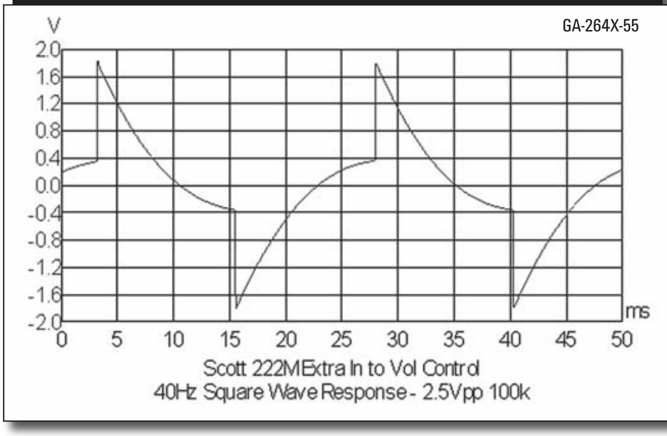


FIGURE 55: Extra input 40Hz square wave response, 100k.



resistors R117 and R217.

I again made frequency response measurements with a test signal level into my inverse RIAA network that produces 10mV at 1kHz at the Phono input jack. Gain at 1kHz, 10mV input improved from 32dB to 36.2dB, which is closer to the ideal 40dB.

Figure 43 shows the RIAA equalization error relative to 36.2dB (the line labeled PHONO), with 1kHz the 0dB point. The RIAA accuracy varied –1dB to +0.15dB from 11Hz to 17kHz, and –3dB to +0.15dB from under 10Hz to 48kHz. When measured at the output of the preamp section at the top of the loudness control, it again closely follows the IEC 9/76 LF rolloff curve shown in the curved dotted line. The two channels varied from each other by only 0.2dB (not shown).

The input overload for visible output clipping now exceeded the RIAA requirements across the board before hitting 1% THD. It was 52.6mV at 20Hz (5.42mV required), 208mV at 1kHz (50mV required), and a greatly improved 1042mV at 20kHz (477.5mV required).

The THD+N from the Phono input to the Tape Rec output is shown in **Fig. 44**, relative to 10mV input at 1kHz. Recall that it was immeasurably high due to the excessive hum level in the as-received 222C.

A spectrum of the improved hum and noise from the Phono input to the Tape Rec output is shown in **Fig. 45**. The 60Hz component has been reduced from –26dB (5%) to –48dB (0.4%), while the noise floor dropped from about –70dB to about –95dB. I think a major portion of this improvement is due to the power supply modifications and the replacement filter caps.

The input impedance at the Extra and Tuner inputs was 918k. Hum and noise at the top of the volume control with the inputs terminated at 600Ω was 3.4mV (A) and 2.7mV (B) for –55dB and –57dB. The A-weighted values were 0.45mV and 0.14mV, or –73dB and –83dB, respectively. This shows the noise performance of the preamp A channel to be worse than the B channel. Switching preamp tubes between the two channels did not materially alter the readings.

Loudness control tracking between the two channels with a 0.5V input signal favored left channel A throughout the rotation. Its worst divergence was at about 11 o'clock (1.6dB), decreasing to about 0.6dB until 1 o'clock, and then increasing again to just over 1dB from 2 o'clock to the maximum. Crosstalk was identical from L to R and R to L, at about -39dB referred to 0.5V RMS input from 1kHz to 20kHz. This result is dominated by noise.

The preamp output impedance (at the top of the loudness control) was 700Ω at 20Hz, 560Ω at 1kHz, and 630Ω at 20kHz.

The frequency response of the preamp section, from the Extra input to the top of the loudness control, is shown in **Fig. 46**. The graph is normalized to the +13dB available gain, with 0dB at 1kHz and 100k load. This graph differs from the preamp response graph of the original unit in **Fig. 6** (Part 1) in that regard.

Note that the SPICE-predicted response based on the 222D schematic is not as good as the actual response of the modified unit. The HF response peak drops off above 20kHz, while the 222C response and

FIGURE 56: Extra input 1kHz square wave response, 100k.

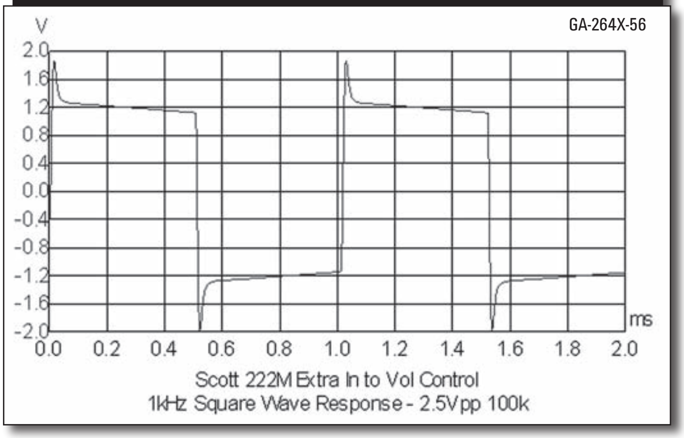


FIGURE 57: Extra input 10kHz square wave response, 100k.

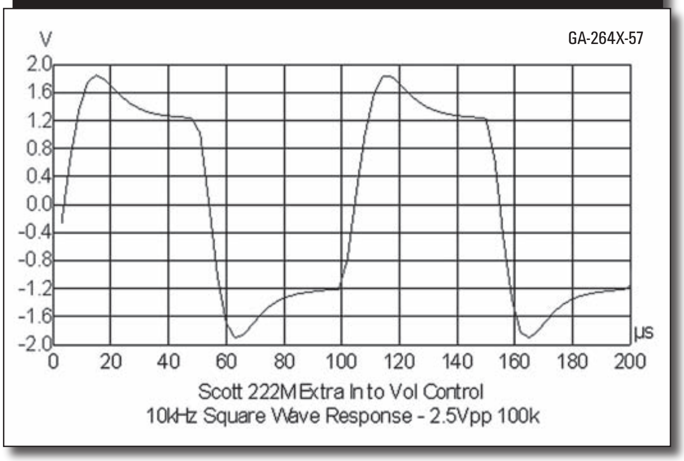


FIGURE 58: Power amplifier frequency response.

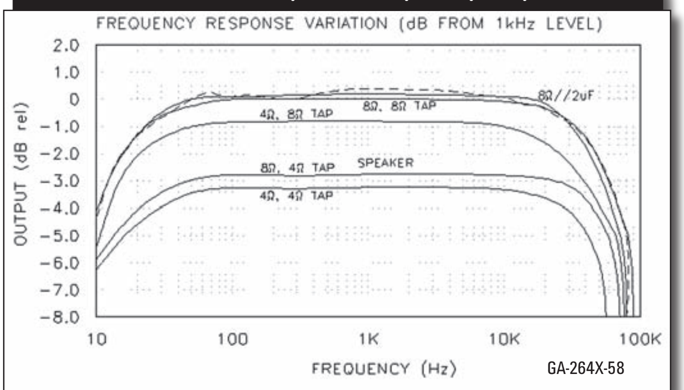


FIGURE 59: Power amplifier frequency and phase response, 8Ω, 3% THD.

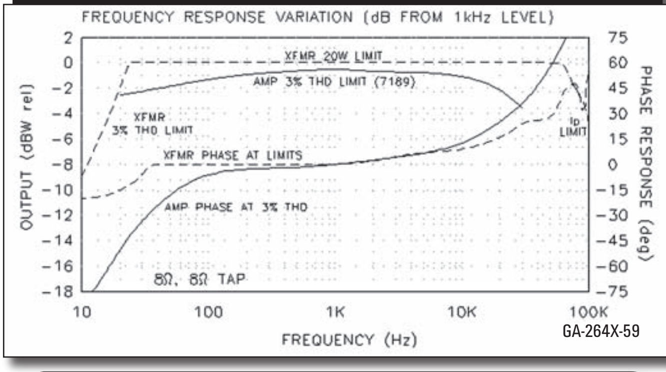


FIGURE 60: Power amplifier THD+N vs. frequency.

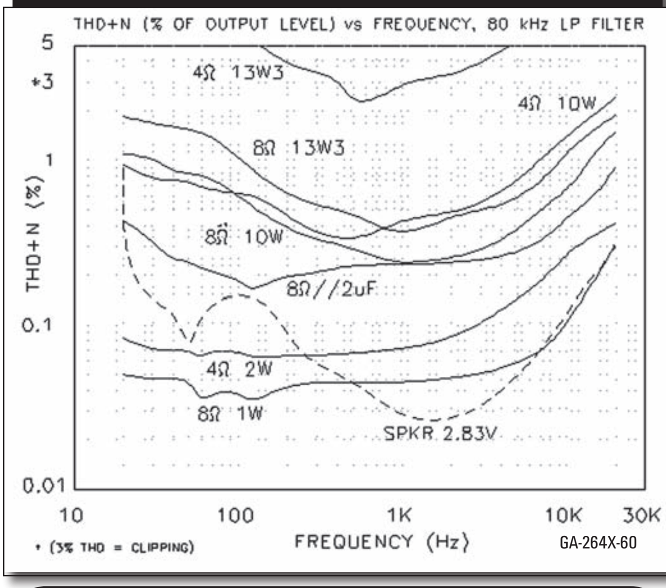


FIGURE 61: Power amplifier THD+N vs. output power.

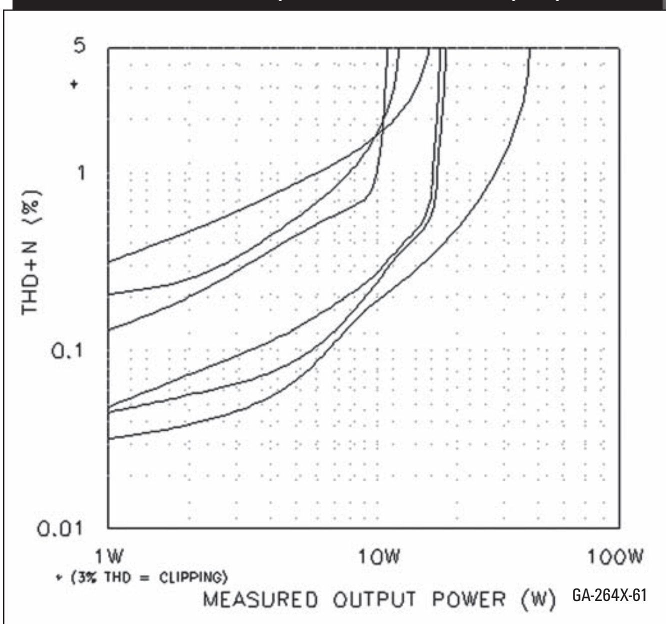


FIGURE 62: Residual distortion, 1kHz 8Ω load.

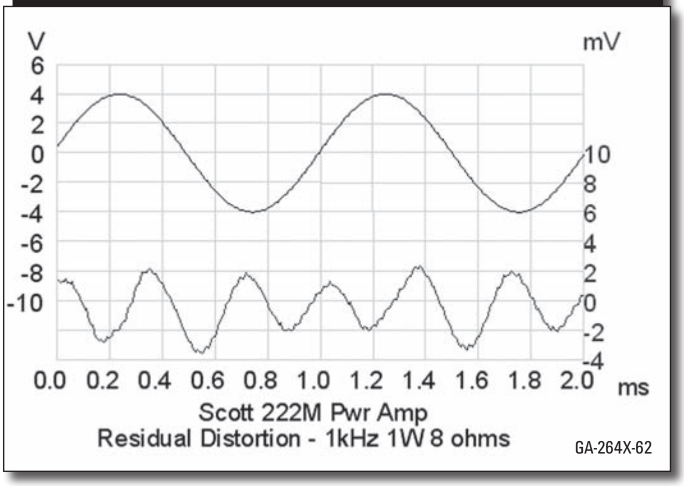
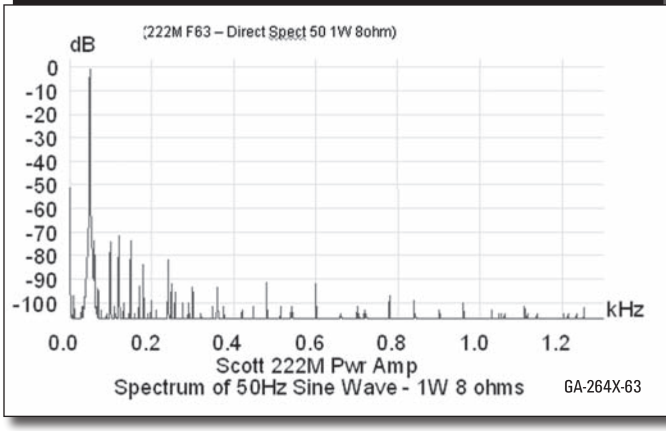


FIGURE 63: Spectrum of 50Hz 1W sine wave, 8Ω load.



the predicted SPICE response continue to rise. I show the actual response at both 100k (the input impedance of my test equipment) and 47k loads. The load impedance is higher in the actual circuit, which is one reason why SPICE predicted a higher 17.5dB gain.

The range of response modification by the tone controls is shown in **Fig. 47**. The loudness contour curve (dashed line) shows more of a peak at each end than the earlier 222C tests in **Fig. 7**, but that may be due to the difficulty of setting the loudness control at exactly the same 9 o'clock point. The revised tone control circuitry also has a bit more authority at the cut end of their settings.

Figure 48 shows the THD+N versus Frequency for the modified preamp section from the Extra input to the top of the loudness control. The solid lines are with the bass and treble controls set flat. The dashed lines are with the bass and then the treble control set to maximum. The hum notch out at 60Hz is quite a bit smaller than in **Fig. 8** (Part 1), showing the improvement in the power supply.

Note that the distortion with a 47k load is lower than that of the 100k load. I also show a 10k load distortion line for information. The preamp load actually connects directly to the volume control with no external connection available. This load impedance will vary from 500k at the minimum loudness control setting to about 250k at the maximum loudness control setting (assuming the balance control is centered).

The THD+N versus output voltage is shown in **Fig. 49** at 20Hz, 1kHz, and 20kHz with a 100k load. Curves for 47k and 10k loads are also included at 1kHz, again only for information. Again the distortion for the 47k load is lower than the 100k load until the output voltage exceeds 6V RMS. The downward slopes of the noise content of THD+N are dominant until the output exceeds 1V RMS or more, where the distortion component takes over on the upward slopes.

The preamp residual distortion for 1kHz from Extra input to the top of the loudness control is shown in **Fig. 50**. The output sine wave is at the top while the distortion residual signal after the distortion test set notch filter is the bottom trace. The residual shows mainly a third harmonic dominated by noise. The trace is gradually rising from left to right, which shows it is modulating the hum from the lower frequency AC line.

You can see the preamp intermodulation of the AC line 60Hz with the 50Hz test sine wave in the spectrum in **Fig. 51**.

I applied the 2V RMS test signal to the Extra input and monitored the spectrum at the top of the loudness control. Hum pickup in the circuit produced 60Hz at –50dB (0.32%). Nonlinearities in the preamplifier produced IM products at 70Hz and 80Hz, and AC line harmonics out through 480Hz. There is one unexplained –52dB spike at 1250Hz. The noise floor is low at -106dB.

Repeating the spectrum with a 1kHz 2V RMS input signal produced the results shown in **Fig. 52**. Here the only harmonics are the 2nd and 3rd at about –70dB, with the power line harmonics shown grouped below the 1kHz fundamental.

Figure 53 shows the output spectrum reproducing a 12Vpp combined 19kHz + 20kHz CCIF IMD signal into 100k. The 1kHz IMD product is –47dB (0.079%) and the 18kHz and 21kHz products are –55dB (0.035%). There are also products at 2kHz, 4kHz, and 17kHz. Taken all together the products equal 0.51%.

Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, **Fig. 54**) produced odd and even order IM products spaced about 1kHz apart with a total calculated value of 0.32%. The multi-tone IMD is more representative of music than a sine wave test signal.

The 2.5 Vpp square wave at the Extra input at 40Hz showed a significant amount of tilt and leading phase shift (**Fig. 55**) as a result of the steep HP filter in the preamp section. The 1kHz square wave (**Fig. 56**) has a peak at the leading edge, possibly coinciding with the slight rise in frequency response at 20kHz in **Fig. 46**. This response peak is more obvious in **Fig. 57**, where it exactly splits the two halves of the top of the 10kHz square wave.

The final batch of figures shows the tests done on the Direct input to the power amplifier section, where the test signals are applied to the top of the loudness control. **Figure 58** shows the frequency response of the power amplifier with various loads, with 0dB representing 2.83V RMS at 1kHz across 8Ω at the 8Ω tap. Both channels were driven for all these tests. The loudspeaker response is shown with a dashed line. I made measurements for the loudspeaker load with my NHT SuperOne satellites at the end of 6′ of 12-gage Monster cable.

Figure 59 compares the frequency and phase response of the power amplifier section of the modified 222C at 3% THD with the transformer limits that were established earlier and shown in **Fig. 26** (Part 2). The output load is 8Ω at the 8Ω tap. The maximum power over the audio band (solid line) is below the transformer power limit (dashed line) down to 18Hz.

At that point, the amplifier LF rolloff pretty much follows the transformer 3% THD limit line. This tells me the steep HP filter in the preamp was there to roll off the phono response to prevent LF record warp from getting through to the speakers of the day, and not to prevent saturation of the output transformer.

FIGURE 64: Spectrum of 50Hz 4V sine wave, speaker load.

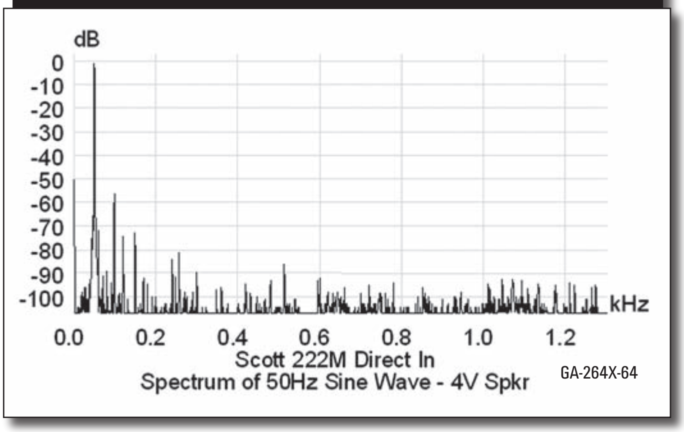


FIGURE 65: Spectrum of 50Hz 5W sine wave, 8Ω load.

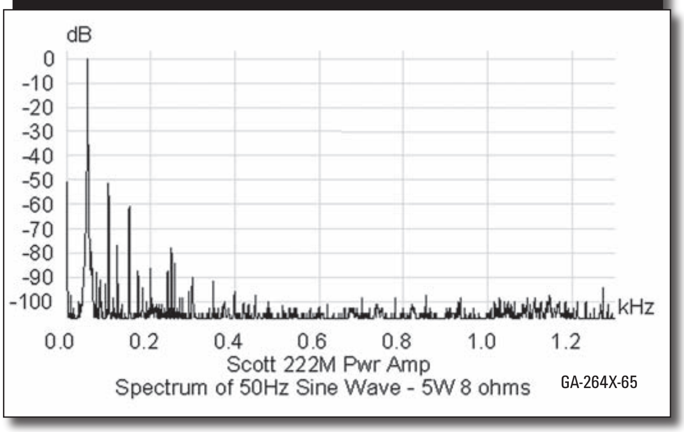


FIGURE 66: Spectrum of 19kHz + 20kHz intermodulation signal.

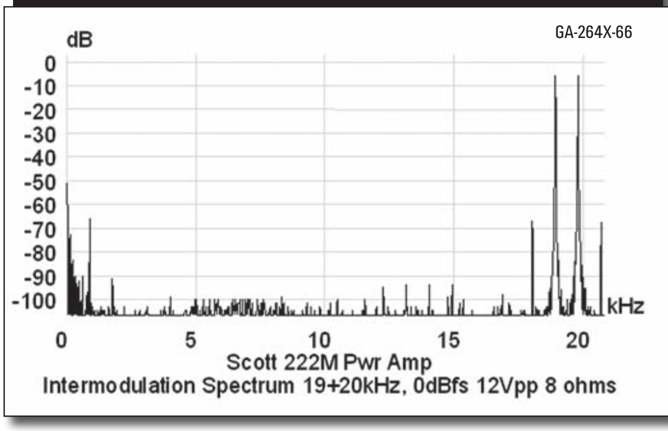
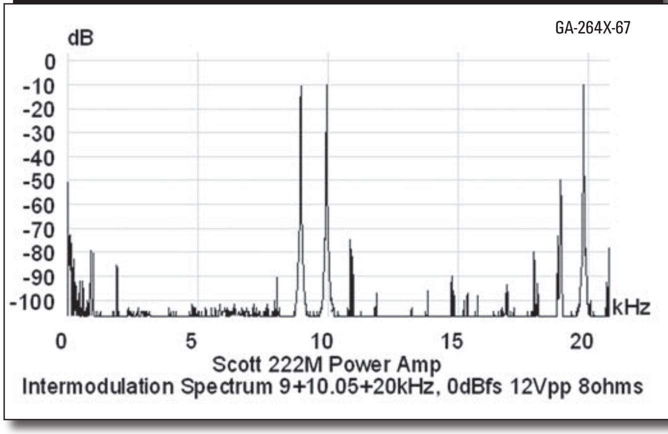


FIGURE 67: Spectrum of 9kHz + 10.05kHz + 20kHz intermodulation signal.



C127/C227 (**Fig. 36**, Part 3) provide the dominant feedback pole to roll off the HF response below the limits of the output transformer.

Because the amplifier high and low frequency rolloff is faster than the inherent rolloff of the transformer, the rate of change in the amplifier phase response is proportionally higher.

The power amplifier THD+N versus frequency is shown in **Fig. 60** with the loads indicated on the graph. Note the large excursions in distortion with the speaker load as the amplifier responds to changes in speaker impedance with frequency. The dips in response at 60Hz and 120Hz at 8Ω 1W are at much lower dB levels than with the preamp section in **Fig. 48**, indicating there is much less hum to notch out in the power amplifier. The distortion only exceeds 3% in portions of the 13.3W curve into a 4Ω load.

Figure 61 shows the amplifier THD+N increase as the output power is pushed higher by increasing the input signal, with the loudness control at maximum clockwise rotation. Both channels are driven. The loads from bottom to top at 1W are 8Ω at the 4Ω tap, 8Ω at the 8Ω tap, 4Ω at the 4Ω tap, 4Ω at the 8Ω tap, the preceding all at 1kHz; then 8Ω at the 8Ω tap at 20Hz and 8Ω at the 8Ω tap at 20kHz.

The residual distortion in **Fig. 62** shows mainly the third harmonic with some noise. The test signal is 1kHz at 1W into 8Ω at the 8Ω tap.

The spectrum of a 50Hz sine wave producing 1W into 8Ω (8Ω tap) in **Fig. 63** shows the harmonics and AC line intermodulation products all below –70dB, with a low noise floor. Replacing the 8Ω load with the loudspeaker in **Fig. 64** produces an increase in the second harmonic and a bit more complexity in the noise floor. Increasing the output power to 5W (**Fig. 65**) increases the percentage of all the harmonics but leaves the AC line product levels lower than they were at 1W.

Next, I applied my two intermodulation test signals. The CCIF 19 + 20kHz IMD spectrum is shown in **Fig. 66**, at 12Vpp into 8Ω. While the 1kHz, 18kHz, and 19kHz products are all at –68dB (0.04%), there are no other multiples of 1kHz above –92dB. This is a much better result than in **Fig. 53** with the preamp.

Switching the IM test generator to the multi-tone signal produced an overall lower level of IMD products (**Fig. 67**), except for the 18.95kHz product that remains at –50dB. The total calculated value of the even and odd order products is 0.103%.

The next series of tests are all responses to square waves at the Direct

FIGURE 68: 40Hz square wave response, 8Ω.

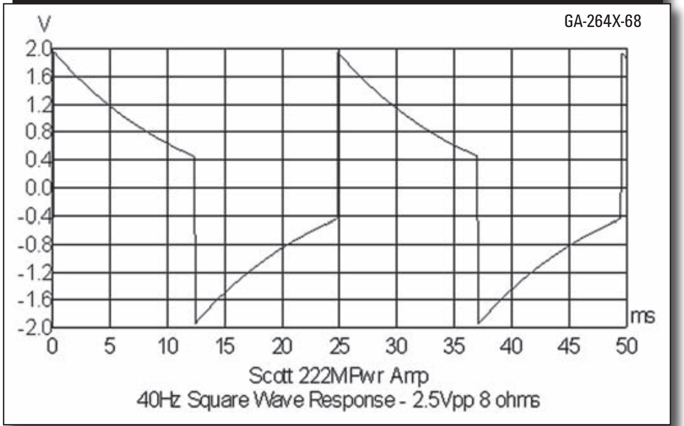
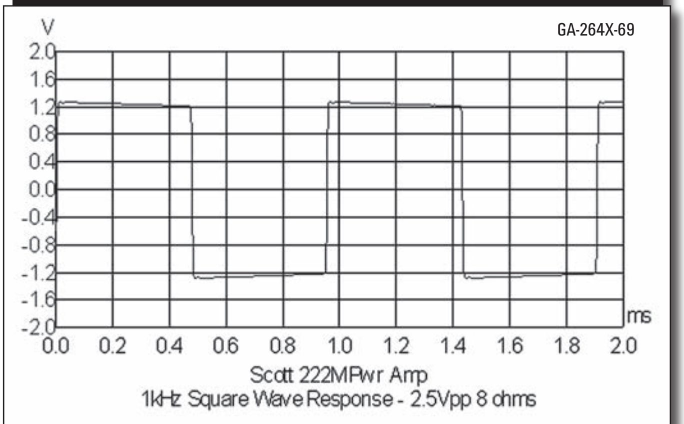


FIGURE 69: 1kHz square wave response, 8Ω.



input to the power amplifier. I adjusted the output for 2.5Vpp into the 8Ω load at the 8Ω tap. **Figure 68** shows the response to a 40Hz square wave. It shows only moderate tilt when compared with the response of the preamp in **Fig. 55**.

The 1kHz square wave in **Fig. 69** is just about perfect. Increasing the square wave frequency to 10kHz (**Fig. 70**) produced a waveform with no peaking and a slight rolloff of the edges. When a 2μF capacitor is switched across the 8Ω load, the response is little changed (**Fig. 71**).

Crosstalk performance for each of the sections of the modified 222C is shown in **Table 7**. A summary of the modified 222C test data versus the original specifications is shown in **Table 8**.

EPILOGUE

While work on this unit may be finished for now, I still want to do some more testing and modifications. I'm fairly pleased with most of the modifications to the 222C, and I learned quite a bit along the way. The hum and noise remaining after all the modifications to my power supply points to the reasons why modern tube amplifiers use voltage regulators in the B+ circuits for all the voltages below that of the output tubes.

The power-supply rejection of single-stage amplifiers is not very good. Most modern push-pull amplifiers also use ultralinear (UL) output transformers. Dennis Colin made an interesting comparison of KT88 plate curves for triode, 70% UL, 43% UL, and pentode output stage topologies in a letter to *audio-*

FIGURE 70: 10kHz square wave response, 8Ω.

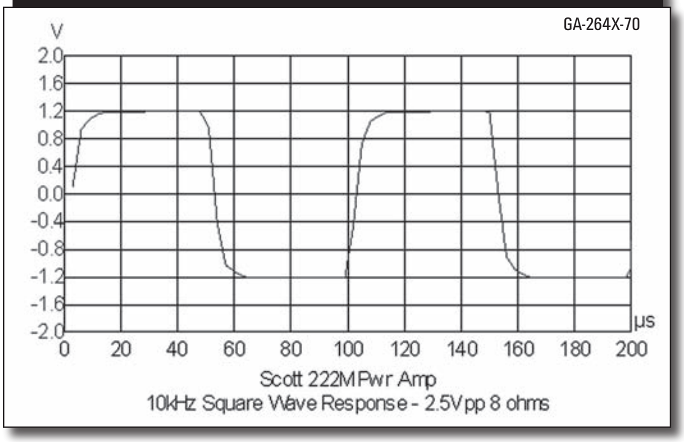
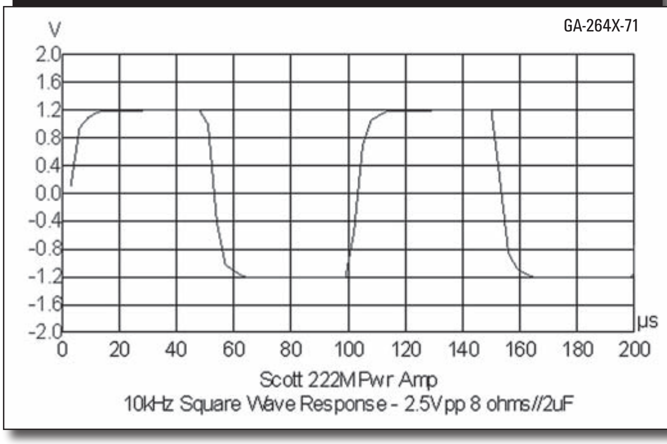


FIGURE 71: 10kHz square wave response, 8Ω//2μF.



Xpress, (Aug. '05, p. 60).

I'll try to find some reasonably priced 7189A output tubes and compare them with the 7189 and 6n14n types. I may remove or modify the preamp input high-pass filter because it does not seem to be necessary to prevent output transformer saturation. There is an undesirable HF peak in the preamp frequency response that I may be able to eliminate with a small compensation cap from the plate coupling cap output to the grid of one or both tubes. And the hum and crosstalk performance of the preamp are inferior to both the phono section and the power amplifier.

TABLE 7: CROSSTALK PERFORMANCE.

	Phono dB		Preamp dB		Amp dB	
Freq	L to R	R to L	L to R	R to L	L to R	R to L
100			–58	–57	–65	–52
1k			–52	–44	–65	–52
10k	–44	–54	–40	–39	–64	–52
20k			–39	–38	–61	–52

TABLE 8: MEASURED PERFORMANCE (CHANNEL B, MODIFIED 222C).

Parameter	HH Scott Specifications	Measured Results, Unmodified
Max power output, 1000 cycles	Music waveforms - 24Wpc Steady state – 20Wpc	No specified test method 17.8Wpc, 8Ω at 8Ω tap
Total Harmonic Distortion	0.8% rated power, 1000 cycles	3% THD, 17.8W, 1kHz, 8Ω 0.8% THD, 10W, 1kHz, 8Ω
Frequency Response, 20W steady-state, 1.5% THD	20 to 20,000 cycles*	18Hz – 30kHz (8Ω relative to 10W), 1.5% THD
Max usable power output, 20 cycles	Music waveforms – 28Wpc Steady state – 24Wpc	No specified test method 11.1W, 3% THD, 8Ω at 8Ω tap
Power Bandwidth at 1.5% distortion (IHF method)	Below 19 cycles to above 20,000 cycles (test equipment limits)*	20Hz – 30kHz (8Ω relative to 10W), 1.5% THD
Intermodulation distortion	Below 0.5%	0.045% CCIF 8Ω
Signal for rated output at 1 kc		
NAB (NARTB) tape	3.0mV	Circuit Removed
RIAA (MAG LOW)	3.0mV	19.5mV RMS
RIAA (MAG HIGH)	9.0mV	Circuit Removed
Tuner, Extra and Tape Playback	0.50V	0.67V 10W 8Ω
Hum and noise		
High level inputs	80dB below rated power	–53dB ref 10W
Low level inputs	10 microvolts equivalent	No specified test method
Scratch Filter	Above 5 kc	Circuit Removed
Treble boost, treble cut, 10 kc	15 dB ± 2dB	14dB boost, –13dB cut
Bass boost, treble cut, 50 cycles	15 dB ± 2dB	15dB boost, –17dB cut
Input Impedance		
Low level (MAG LOW)	47kΩ	47k51
High level (MAG HIGH)	150kΩ	Circuit Removed
High level inputs	500kΩ	918kΩ
Output loads		N/A
Tape load resistance	200kΩ minimum	
Tape out cable capacitance	200pF maximum	
Line voltage range	105–125V, 50-60 cycles	Optimized for 120V AC
Power Consumption, 60 cycles	170W	174W max, 132W at 2Wpc 8Ω
Damping Factor	N/S	4.7 4Ω, 6.3 8Ω; 1kHz
Input Overload, Mag Low		
20Hz		52.6mV RMS
1kHz		208mV RMS
20kHz	N/S	1042mV RMS
Gain, 1kHz, Mag Low	N/S	10mVin produces 646mVout, or 36.2dB
RIAA Accuracy	N/S	–3.0/+0.15dB
S/N, A-Wtd, Ref 5mVin	N/S	–64dB (phono), –73dB (Extra)
Crosstalk, 10kHz	N/S	–44dB (phono), –38dB (Extra)

** Sharp cutoff filter (12dB or sharper per octave) becomes fully operative below 20 cycles.*

I hope this article encourages you to look at the vintage tube hi-fi gear, much of which is available on eBay. I found an average of two pages of listings each for many vintage manufacturers: Scott, Fisher, Pilot, Eico, Heathkit, and so on. The Japanese manufacturers also produced tube hi-fi equipment towards the end of the era.

It is important to find equipment that is in good cosmetic condition with good transformers. Understandably, this will cost you more money. If the unit is operating, all well and good, but remember that I replaced most of the electronic parts in my unit. If you want do a more faithful restoration, the condition of switches, potentiometers, and the other mechanical items that are no longer available takes on more importance. **aX**

References

10. "Modifying Peavey's TKO 80 Bass Amp, Part 1," Hansen, C., Koonce, GR; *audioXpress*, pp. 16-27, 67, Oct '03; "Part 2," Hansen, C., Koonce, GR; *audioXpress*, pp. 40-49, Nov '03.

Next time...

I will wrap up this series in next month's issue with a look at tube specifications and alternatives you can use in the Scott 222C. I audition and measure the performance of these tubes.

VENDORS

Audiophile components: JFETs, MOSFETs, Tantalum, Caddock, Dale resistors, Black Gate, ELNA Cerafine, Nichicon Fine Gold Muse, Jensen Four-pole, Mundorf Tin-foil caps, Stepped attenuators, Teflon cables, Connectors.
www.borbelyaudio.com
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Selected BORBELY AUDIO kits in Japan:
http://homepage3.nifty.com/sk-audio

www.UnityAudio.ca Canadian hand crafted cables and loudspeakers, great with tube gear!

Several new products including models 444 headphone amp and 465 6-channel volume control.
www tdl-tech.com (505-382-3173).

WANTED

stereo gear old/new/quality, speakers, amps, preamps, turntables,
reel to reels, vacuum tubes, tube testers, guitars, parts, etc. **850-314-0321**
Email: **sonnysound@aol.com**

Ad Index

Yard Sale

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6. Maximum 50 words. No accompanying diagrams or illustrations or logos will be used.

"Yard Sale" is published in each issue of *aX*.

HEATHKIT TUBE AMP

After reading the excellent article "Upgrading Heath's W6 Power Amp" by Mr. Brown (March '06, p. 6), I can relate to the bias/stability problems that I had with these amps. The 12BH7 cathode-follower circuit was not friendly to the 6550 output tubes in regard to stable bias voltage, and when the 12BH7 aged just a little bit, everything was in limbo as to a possible "runaway" plate current situation.

I have enclosed a partial schematic of the driver circuit that I now use on my 120W power amplifiers. The 6BL7/6BX7 twin triodes are wonderful for this application, and no overall "nasty" feedback is needed for this circuit. Having the .22mF/600V DC capacitors in the signal path might be undesirable to some designers, but I have yet to have an output tube fail due to a defect in the cathode-follower circuit. Because the 6BL7/6BH7 tubes have large, massive cathodes, there should be no problems (tube driving, aging, and so on) as experienced with the 12BH7 in this application. However, you could modify this circuit for the W6 amplifier or any related circuit that uses tubes as the 12BH7, 6FQ7, 6CG7, 6GU7, and so on.

MOV/Genelex used direct-coupled cathode followers in several of their high-powered amplifiers using KT88 tubes in multiple pair format, with capacitor coupling before the output tubes. For applications with two output tubes, I have had excellent results with 6SN7s.

As always, you provide an excellent publication and I hope to contribute more during the year.

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

CLASSICAL COAXIAL APPROACH

Mr. Koonce's excellent article on Driver Interference and Display Techniques (April 2006, p. 6) brought to mind the classic RCA LC1A loudspeaker of the '50s (**Photo 1**).

The LC1A was a 15″ coaxial driver, having a 2″ tweeter at its center. Harry Olson attempted to solve exactly the same problem described by Mr. Koonce: interference of the tweeter by a concentric ring. In the case of the LC1A, the concentric ring was the woofer itself.

Dr. Olson's innovative solution was the seven "diffractor cones" glued to the woofer cone. Close inspection shows that the seven diffractors are not uniformly placed on the woofer cone. This served to further randomize the distribution of reflections.

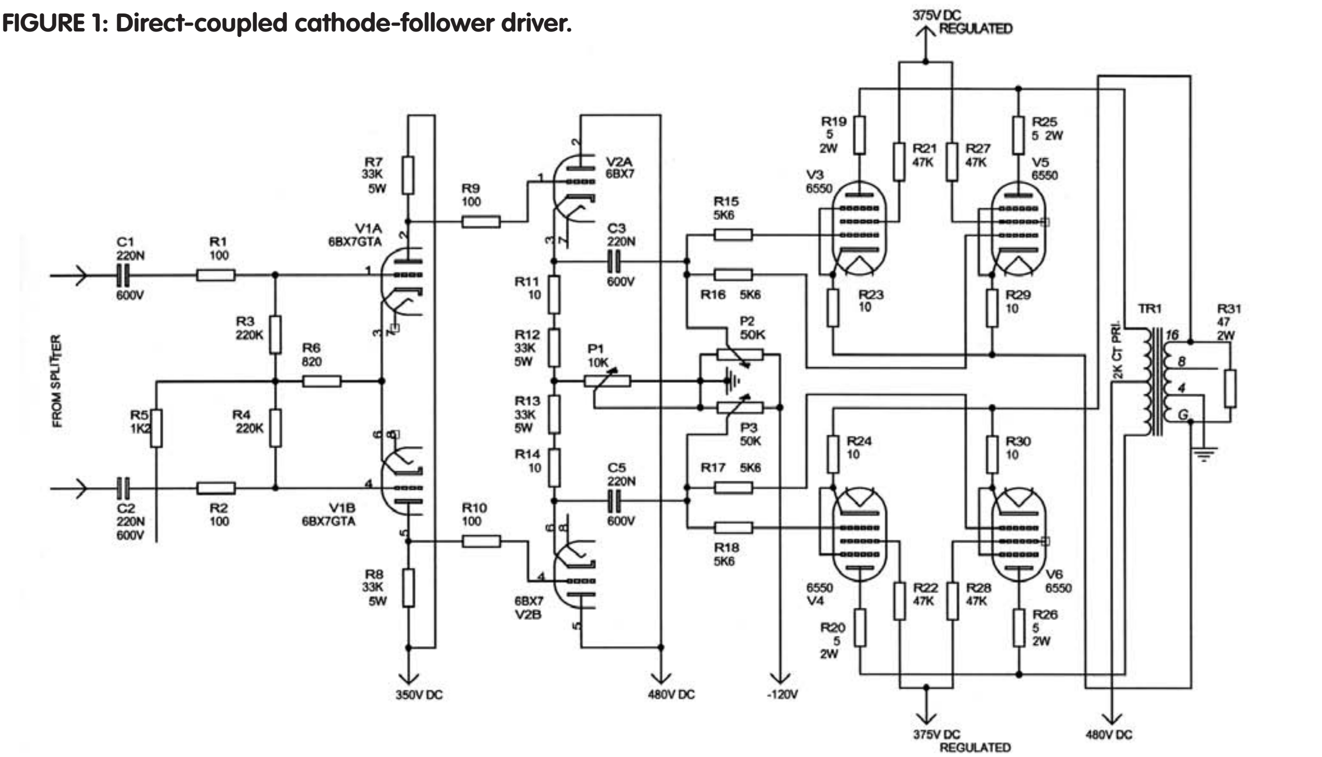
I wonder whether this strategy could be used to reduce the adverse effects of a rectangular grille frame. It should be possible to glue "diffraction disks" to the baffle in a semi-random pattern around the tweeter or near the grille frame.

Mark Rumreich
mark.rumreich@thomson.net

G.R. Koonce responds:

I thank Mark Rumreich for his interest in and comments on my article. Regarding the RCA LC1A loudspeaker, it mortifies me to admit I'm old enough to have listened to one; hi-fi was monaural in those days. Compared to

FIGURE 1: Direct-coupled cathode-follower driver.



popular drivers of the time, the RCA driver seemed to lack presence and high-frequency response. I don't know whether the driver was being used properly or was in some way defective, but after 50 years I simply remember being "unimpressed."

This driver is covered in Dr. Olson's book *Acoustical Engineering*[1]. Olson points out that the smaller conical domes were attached to the large cone to reduce the velocity of wave propagation in the large cone. Quoting Olson: "This broadens the directivity pattern of the low-frequency cone. In the high-frequency range, the conical domes attached to the surface of the low-frequency cone improve the performance in three ways: by decreasing the angle into which the high-frequency cone feeds, thereby increasing the output of the high-frequency cone; by diffusely reflecting some of the sound emitted by the high-frequency cone, thereby eliminating direct reflections; and by diffracting some of the sound emitted by the high-frequency cone, thereby broadening the directivity pattern."

This RCA driver was developed at a time when woofer cone excursions were rather limited. Today, when many large woofers have huge excursion capability, I would worry about Doppler distortion problems with a driver using the woofer cone as the "waveguide" for the high-frequency driver.

In general when attaching the "diffraction/reflection problem" with loudspeaker enclosures there are three basic approaches. The first is to eliminate the structure causing the problem with such methods as flush-mounting a tweeter to eliminate diffraction at the faceplate edge. In general this approach ultimately fails. As noted in my report, as soon as you mount anything else on the enclosure front panel, the response of that flush-mounted tweeter becomes compromised. Unless you are mounting the enclosure flush into a large wall (infinite baffle), you will eventually need to account for the box edges. Additionally for driver protection, you must also devise some way to accommodate a grille on the enclosure.

A second approach is trying to "spread out" the interfering structure. If you can do this sufficiently, then it is possible to reduce the interference signal to an acceptable level. The "Diffraction Ring" developed by David Weems and myself to smooth the response of a surface-mounted tweeter works this way. Its development and performance are covered in reference #3 of my article. As Mr. Rumreich indicates, this approach was used with the RCA driver and may be applicable to front panel treatment. I think such an application would require a rather large front panel to work with, but with sufficient work and testing could be successful.

Someone suggested to me that painting the front panel with one of those paints that produces a very rough surface might be helpful. I doubt that this is effective, but have not yet tested it. As demonstrated in my article, avoiding symmetry in any diffractive structure is very important.

The third approach is to prevent the acoustic energy from getting to diffractive/reflective structures. This is generally done by attempting to absorb the acoustic energy via damping materials. Fibrous-tangle materials such as fiberglass and felt have shown to be effective, with limits, if properly applied. The literature indicates certain foam materials are also effective, but I have yet to identify one that is cheaply available to the home builder.

The best way to handle diffraction/reflection interference problems today seems to be by applying a combination of all three of the above approaches. Even then the in-system response of the drivers will probably vary somewhat from their anechoic response. Tweeters are especially sensitive to how they are mounted, what diffractive structures exist on the front panel, and the grille frame/cloth. The midrange driver is very sensitive to the chamber in which it is mounted, how it is coupled to that chamber, and again diffrac-

PHOTO 1: RCA's LC1A 15″ coaxial driver.

tive structures on the front panel. There is no such thing as taking too much care in these areas.

REFERENCE

1. Harry F. Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., New York, 1957, page 139.

NOT-FOR-PROFIT

I found a copy of *Speaker Builder* (from several years ago) and was really fascinated by its contents. I've been collecting—and rebuilding—vintage audio gear since I was a teenager. . . everything: ranging from 1960s Acoustic Research speakers to (1973-era) AKAI, analog, quadraphonic (!) reel-to-reel tape decks.

What piqued my interest (in the particular copy I'd found—6/99 *SB*) was an article by Mr. Charles Hansen, concerning the restoration of an old pair of Lafayette "Criterion" bookshelf speakers. My view on such a project would've been this: Why waste $235 (the cost of the replacement parts—from Madisound—as quoted in the article) on a pair of speakers which were, undoubtedly, garbage to begin with? I mean, just a trip to your average weekend yard sale and/or flea market (where I get most of my wares) could land you (vintage) equipment (speakers, in this case) *far superior* to what Mr. Hansen chose to work with. Here are some examples of *other* two-way, 8″-10″ woofer/cone tweeter systems (like the Lafayettes, but better): AR-4x, Advent 1, Bose 301, Dynaco A25, JBL 96, KLH 17, KLH 22, KLH 33, Rectilinear XI, Rogers LS7, and Wharfedale W25 (to name a few).

I—with the help of one of your advertisers—rebuilt a pair of (1968-vintage) AR-2ax speakers. I use them with a (1971-vintage) Marantz 1060 integrated amp (on which I replaced all the filter caps as well as upgraded the internal wiring), a (1973-vintage) AKAI reel-to-reel tape deck (I've a substantial collection of pre-recorded "albums" in this format), and a (1977-vintage) Thorens TD-160MKii turntable.

James Hoone
South River, N.J.

Charles Hansen responds:

Point well taken, Mr. Hoone. You must shop some pretty ritzy yard sales, and congratulations on your finds. I wrote the article on the Criterions [Criteria], which are still serving nicely almost 7 years ago, before I did anything with speakers at all (see p. 4 of that same issue). So it started out as a curiosity and turned into a valuable learning experience. To paraphrase the credit-card commercial:

Cost to refurbish two Criterions: $235
Lessons learned and experts I met as a result, especially GR Koonce: Priceless

I went on to do a couple more loudspeaker-related articles with GR, who showed me that there is indeed a science behind loudspeaker design. I agree that this kind of hobbying is indeed satisfying. I recently upgraded a Harman-Kardon HK-460i receiver (designed in part by John Curl) in the same way you described for your Marantz. If you restored any of those vintage loudspeakers, I encourage you to share the results with an article of your own. *aX*

Outlaw Model 200

By Charles Hansen

Outlaw Audio
PO Box 975
Easton, MA 02334
866-688-5297
www.outlawaudio.com

Web-direct price $299 US
Weight: 18 pounds
Dimensions 1.75″ H × 17″ W × 11.5″ D
Warranty: 5 years

This review is a revision of a first one done by our Equipment Review specialist Charles Hansen. After our first measurements discovered some less than ideal results, the Outlaw management asked whether they could make suggested changes in the device and resubmit a second sample. This new review is the result.

We asked the manufacturer whether the changes made would be part of a new version of the amplifier. They indicated that it was not since they had a quantity of the boards which are the basis of the amp topology. These have been modified to achieve the improvements and will be part of regular offerings to buyers until a new batch is required. Charles Hansen assures me that this is often standard practice in the industry. We are satisfied that the revised version of the amplifier performs as we report it in this review.

I cannot resist pointing out that the above incident is not unique. Our reviewers often find problems, discuss them with manufacturers, and this leads to product improvements that might otherwise not be made. Please note that this is a service of audioXpress *to its readers and those in the industry whose products we review. —E.T.D.*

The Outlaw Audio Model 200 M-Block, which is sold only through the Internet, is a monoblock Class-G power amplifier. According to the Outlaw website, it uses a proprietary hybrid Class-A/B/Class-G design and is conservatively rated at 200W into an 8Ω load, 20Hz–20kHz, at less than 0.05% THD; or 300W into 4Ω. It also has a short-term dynamic power rating of 300W into 8Ω. The amplifier is compact at just 1¾″ (one rack-unit) high with a footprint that matches conventional 17″ wide components. There are two screws on each forward side that look as though they might accept rack mounts for pro audio applications.

The Model 200 is designed to be either operating or in standby mode, so there is no front-panel power switch. In the standby mode the Model 200 is turned on or off by a standard 12V DC (6–35V) trigger signal applied to a rear-panel jack, or when the "music sense" circuit detects the presence of an audio signal at the input jack. When the audio signal stops for 10 minutes, the amplifier automatically turns off.

INSIDE THE AMPLIFIER

Photo 1 shows the front panel, which has a three-color LED status indicator just below the Outlaw logo. It shows green when the unit is on, yellow when the unit is in standby, and red when the unit goes into self-protect mode.

The heavy gauge steel chassis is quite rugged, and there are a large number of cooling slots in the top cover and bottom plate to enhance cooling. The cover is secured to the chassis with a total of nine screws. The amplifier sits on four plastic disks with rubber inserts. There is only 5⁄16″ of finger space under the unit, making it a bit awkward to pick up from a flat surface. Weight is heavily biased toward the power transformer side of the unit.

The rear panel (left-to-right in **Photo 2**) has a gold-plated RCA input jack, the pair of 12V trigger in-out jacks, and a trigger mode switch with On, Music, and 12V positions. The gold-plated 5-way speaker binding posts are on ¾″ centers, so you can use dual banana plugs. Next comes the line voltage selector switch, the on-off rocker switch, a fuse post with T5AL fuse, and the IEC power receptacle. The unit is furnished with a heavy power cord.

Photo 3 shows the amplifier with the cover removed. All the jacks and switches are PC board mounted types. The input jack center contact is tin plated, and there are two ferrite beads on the PC board near the input. The AC input board has a common-mode toroidal choke and capacitor RFI filter followed by a chas-

PHOTO 1: Model 200 front view.

PHOTO 2: Model 200 rear view.

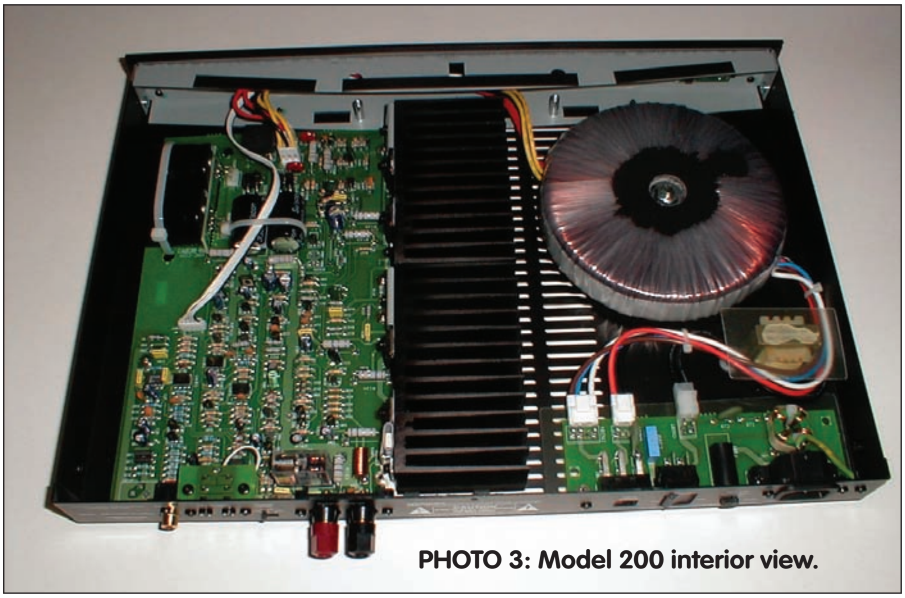


PHOTO 3: Model 200 interior view.

sis mounted choke. Keen Ocean Ltd., a Hong Kong supplier, makes the large 400VA toroidal power transformer.

A pair of hefty heatsinks occupies the center of the chassis. The rear heatsink has a thermal sensor switch to disconnect the speaker in case of excess power dissipation. The amplifier PC board is on the right, and it has a *lot* of components. The power supply is at the front of this board.

TOPOLOGY

A Class-G amplifier design was unveiled in an article by Len Feldman in the August 1976 issue of *Radio-Electronics* titled "Class-G High-Efficiency Hi-Fi Amplifier." In 1977, the Hitachi Dynaharmony HMA 8300 was generally recognized as the first Class-G power amplifier that made it into production.

The design uses two levels of power supply rails to power the output stage. The low voltage handles most of the audio up to some fraction of the rated output, and looks like a conventional amplifier. At some higher level of peak output voltage, the output devices are switched to the higher voltage rails. The advantage of Class-G is efficiency. The only power dissipated for most of the amplifier's operating range is that needed to bias and drive the output stage from the low voltage rails.

There are actually three different types of Class-G output stage[1]. In the series design, the higher voltage output stage is in series with the lower voltage stage. In the shunt design, both the high and low voltage stages drive the speaker load directly. The higher voltage stages remain off until the peak voltage demand reaches the point where they are biased into amplification.

The third type uses a single set of output devices and the power supply rails are shifted between two levels. The two rails are isolated by power diodes. The Outlaw 200 uses this third topology.

Switching spikes from the commutation diodes that deliver the low voltage rails, and then become back-biased when the high voltage rails take over, used to be a major problem. Modern Schottky diodes have pretty much obviated this as a noise source. The two commutation diodes in the Outlaw 200 have RC snubbers soldered across them.

A schematic was not furnished with the Outlaw 200. The power transformer has tapped secondaries (five leads) with two rectifier bridges to provide the two levels of power supply rails. The higher rail supply is filtered by two Su'scon 10,000µF 100V DC aluminum capacitors, and the lower by two Su'scon 3,300µF 50V capacitors.

The Model 200 operates in Class A/B up to 80W. Outlaw Audio says it transitions to Class G within 2µsec above 80W (one AC half-cycle at 20kHz has a duration of 25µsec). Power MOSFETs are used to switch the high voltage rails to the collectors of the output devices and their drivers. The output stage uses two parallel pairs of 15A Sanken bipolar transistors (types 2SC3519 and 2SA1386). High-power 40A MOSFETs switch the collectors of the output devices and their drivers to the high voltage rails.

An air core inductor is connected between the output stages and the positive speaker connector. The lower level audio circuits use carbon film, metal film, and flameproof power resistors. The capacitors are generic mylar film types, X7R ceramic disks, and Capxon aluminum capacitors. A few NP0 ceramic caps are used where critical values are needed. There are also four DIP op amp packages: two TL072s and one each TL071 and LM833.

The board component designations are numbered in five distinct groups. The parts beginning with 100 seem to be associated with the primary audio signal path, since the bipolar driver and output transistors (Q113 through Q118) are in this group. The LM833 and TL071 are also in this group, so these two op amps may handle the input amplification. The 200-series component designations are associated with the raw DC power supplies and the rail-switching Class-G MOSFETs (Q202 and Q203).

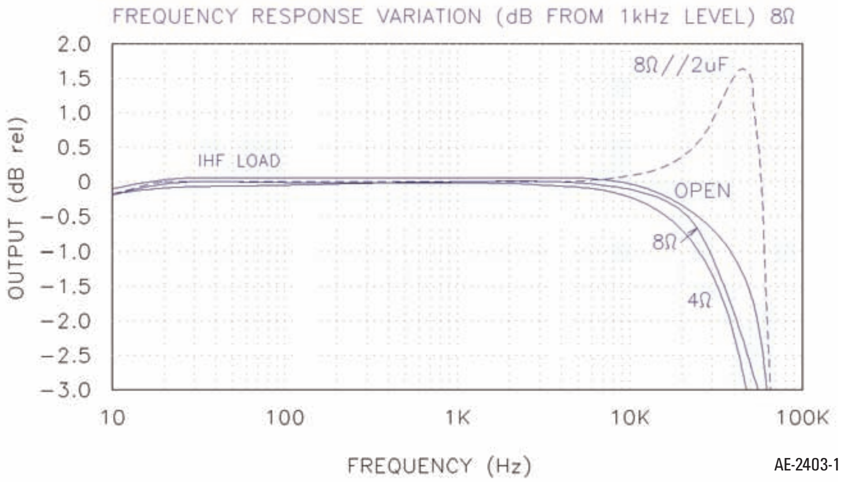


FIGURE 1: Frequency response.

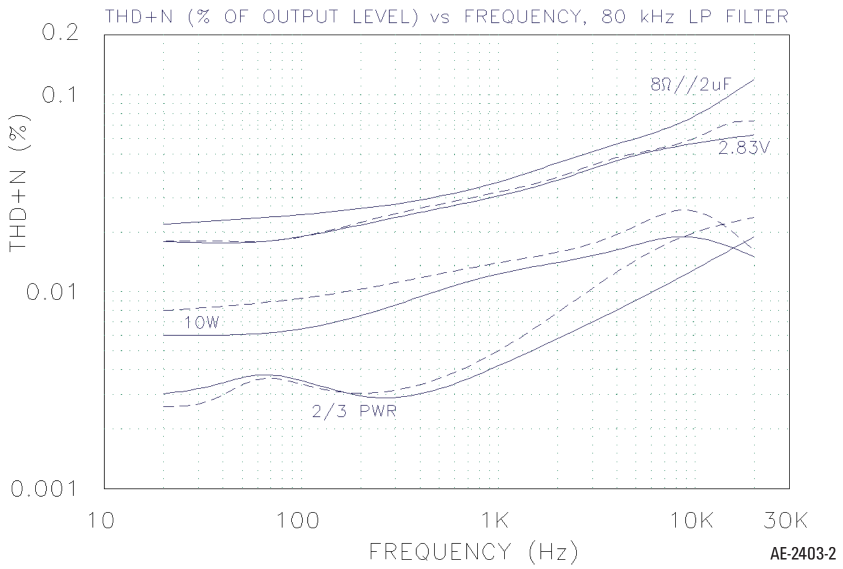


FIGURE 2: THD+N vs. frequency.

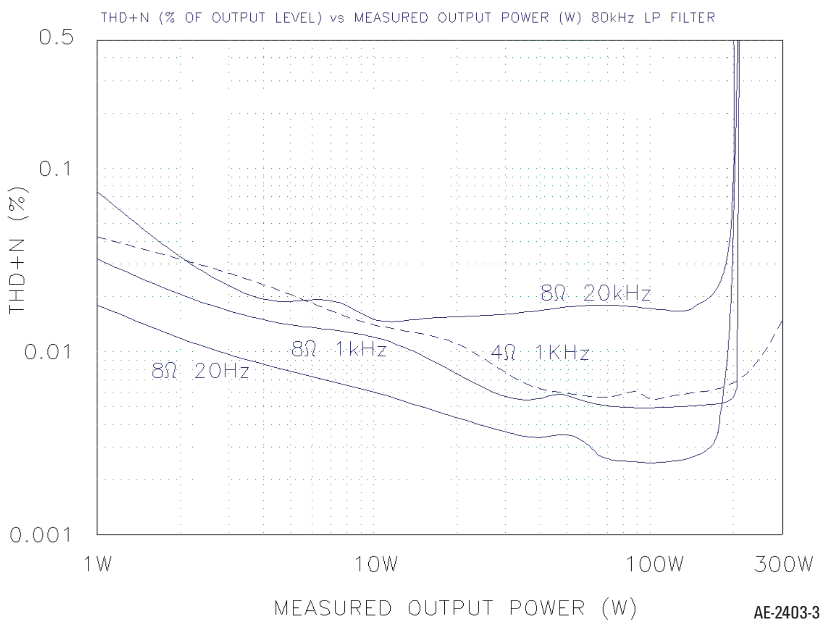


FIGURE 3: THD+N vs. output power.

The function of the 300 series parts is a bit harder to determine. The resistors in this group are mostly carbon film types. They and the other 300-series parts, including one TL074, are grouped around the protection relay, so I assume they form the amplifier protection circuitry. The opto-isolator for the 12V-trigger (and perhaps the "music sense") circuit also falls in this group.

I have no idea of the function of the 400-series parts, which includes the other TL074 op amp. Based on the parts placement, they seem to be associated with the LM833 and TL071 op amps, and include many 1% precision resistors. The 500-series parts are located around two bipolar power transistors that appear to provide the two supply rails for the op amps, derived from the lower set of power amplifier DC rails.

MEASUREMENTS

I operated the single Model 200 amplifier I received at 10W into 8Ω for one hour. The chassis temperature increased to a comfortable 36° C right above the heatsinks. The THD reading at the end of this run-in period was the same as when I began: 0.012%.

There is a brief click in the speaker when starting up, and no noise at all shutting down the amplifier. The output noise consisted of a low-level hiss, measuring 2.8mV RMS (input terminated with 600Ω). I also measured +5mV of DC offset.

The Model 200 does not invert polarity. The input impedance measured 114k with my ohmmeter, but the AC input impedance was a much lower 14k, indicating a capacitively-coupled input stage. The gain at 2.83V RMS output into 4Ω and 8Ω loads was 28.2dB and 28.3dB, respectively. The output impedance at 20Hz and 1kHz (10W) was 0.06Ω, rising to 0.16Ω at 20kHz. The speaker negative terminal and the shell of the input phono jack are connected together, but there is a DC resistance of 220Ω between these audio connections and the enclosure, so the amplifier circuitry is not solidly grounded to the chassis.

The frequency response (**Fig. 1**) was within –1dB from 10Hz to 29kHz, with an output of 2.83V RMS at 1kHz into 8Ω. It was down –3dB at 54kHz. There was just a bit more HF rolloff with a 4Ω load, the –3dB point being 48kHz. The IHF speaker load, which has an impedance peak at 50Hz, produced only 0.1dB rise in the frequency response.

The Model 200 amplifier will be insensitive to variations in speaker impedance with frequency. When I connected a complex load consisting of 8Ω paralleled with a 2μF cap (a test of compatibility with electrostatic speakers), the output voltage began to increase just below 10kHz (dashed line in **Fig. 1**). The output peaked at a benign +1.65dB at 45kHz before dropping off rapidly.

THD+N versus frequency is shown in **Fig. 2** for the loads indicated in the graph. During distortion testing, I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. The peak distortion frequency point varies with the output power. The THD+N increases the most with the complex load. The 4Ω load data is shown with dashed lines.

Figure 3 shows THD+N versus output power for the loads and frequencies shown in the graph (4Ω 1kHz data shown with a dashed line). You can see the slight discontinuity from a smooth curve in the THD just below 100W. This is where the Class-G operation begins as the amplifier switches from its low voltage rails to the higher voltage rails. The 20kHz curve has an additional bump around 6W.

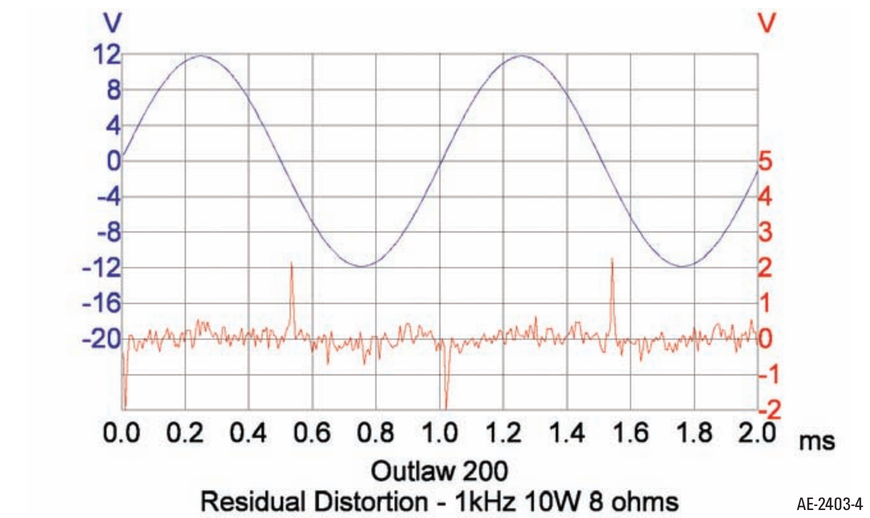
There was absolutely no strain right up to and beyond the point of maximum power, but note the slight increase in THD from 60Hz to 120Hz at two-thirds power in **Fig. 2**. This indicates that the mains power supply is beginning to show up in the output waveform at higher output power levels, albeit at very low levels.

The amplifier reached its 1% 1kHz clipping point at 214W with the 8Ω load (0.47dBW of headroom) and 313W with the 4Ω load (0.18dBW of headroom). Both output AC half-cycles clipped at essentially the same level. While I was determining the 1% and 3% THD levels with the 8Ω load at 20kHz (205W and 207W, respectively), the heatsink temperature sensor switch tripped the amplifier and the front panel LED turned red. I had to wait five minutes for the amplifier to cool down and reset before I could resume testing.

The top of the amplifier reached a maximum temperature of 46° C with 133W into 8Ω. The heatsink would be at a much higher temperature because there is about 1/8″ of air space between the heatsink and the cover.

The distortion residual waveform for 10W into 8Ω at 1kHz is shown in **Fig. 4**. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test set notch filter), not to scale. This dis-

FIGURE 4: Residual distortion.



AE-2403-4

FIGURE 5: Spectrum of 50Hz sine wave.

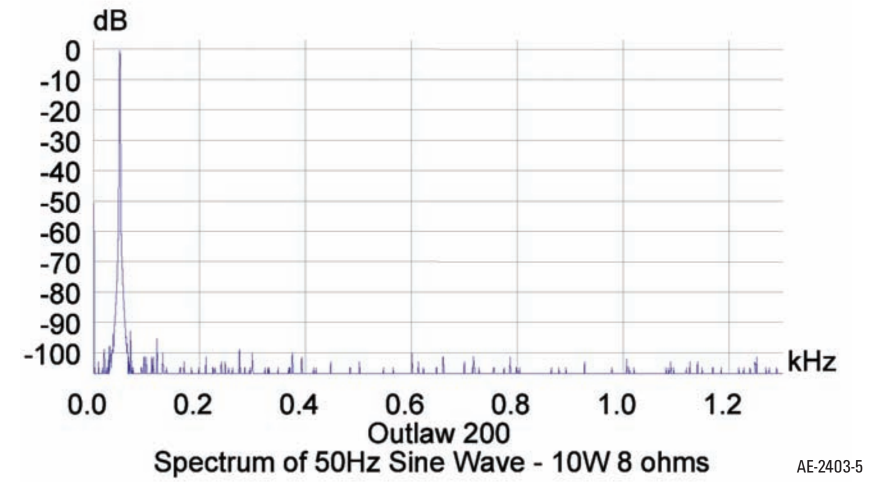


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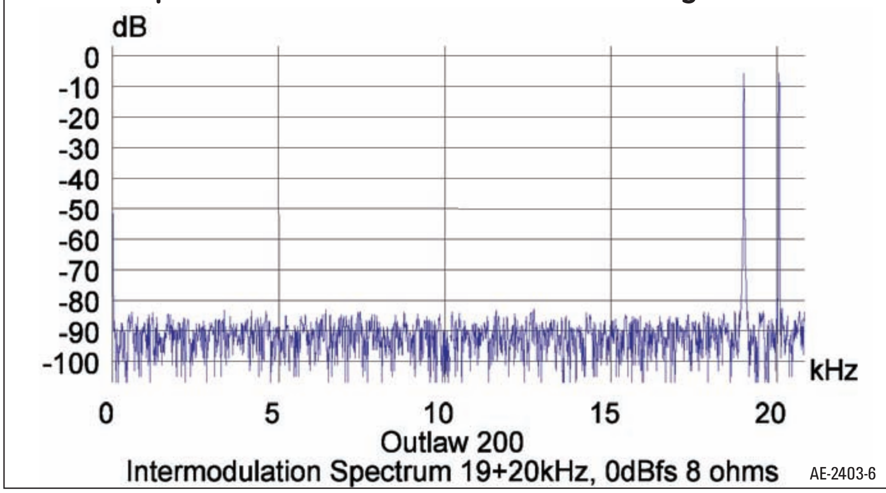
FIGURE 6: Spectrum of 19kHz + 20kHz intermodulation signal.



AE-2403-6

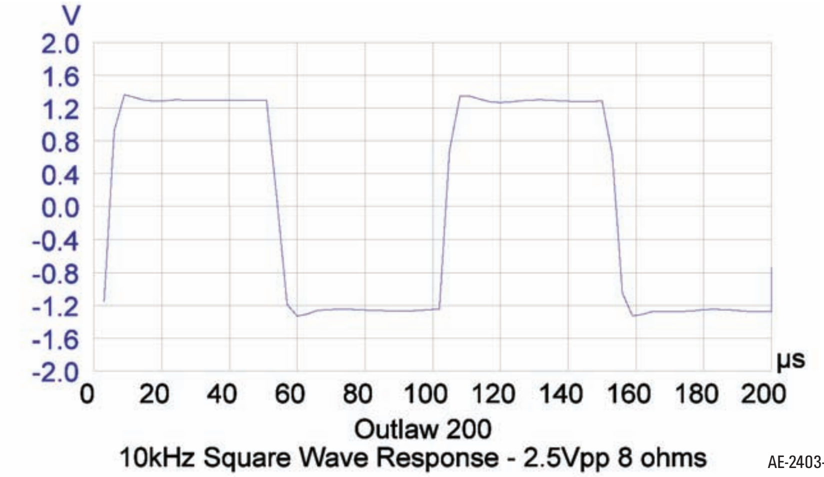
tortion residual signal shows noticeable crossover distortion spikes at each zero crossing. The THD+N at this test point is 0.012%.

The spectrum of a 50Hz sine wave at 10W into 8Ω is shown in **Fig. 5**, from zero to 1.3kHz. The THD+N here measures 0.0069%. The 2nd harmonic measures –98dB, while the 3rd, 4th, and 5th harmonics are all below –103dB. Low-level power line and power supply rectification artifacts are also present at 60Hz and 120Hz. The spectrum of a 1kHz sine wave (not shown) had a nearly identical distribution of harmonics.

Figure 6 shows the amplifier output spectrum reproducing a combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 45Vpp into 8Ω. The 1kHz IMD product is –93dB (0.0022%), and the 18kHz product is –94dB (0.002%) and just barely shows above the noise floor of my IMD generator signal. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, not shown) resulted in a 950Hz product of –88dB and a 1050Hz product of –89dB. This gives a better indication of the amplifier's nonlinear response, because it is a closer approximation to music than a sine wave. The Model 200 produced excellent IMD results.

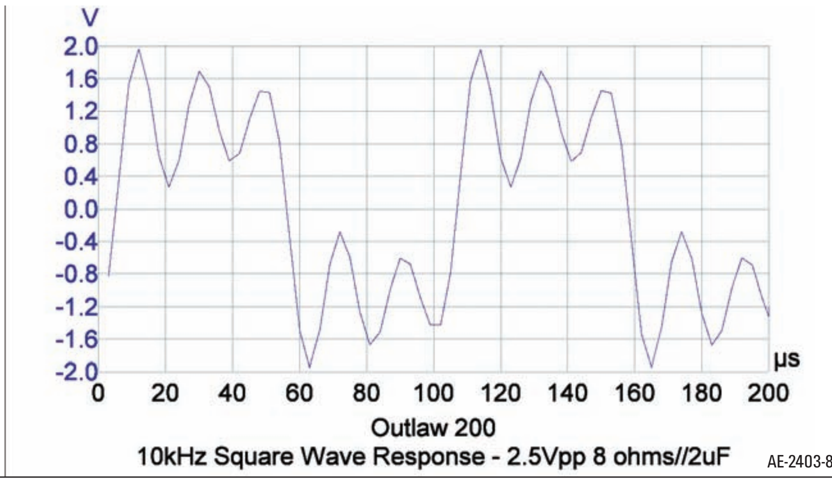
2.5Vp-p square waves of 40Hz and 1kHz into 8Ω were almost perfect, with just a hint of peaking on the leading edges of the 1kHz wave. The leading edge of the 10kHz square wave contained a noticeable peak about 18µsec wide (**Fig. 7**). When I connected 2µF in parallel with the 8Ω load, the Outlaw 200 displayed an underdamped 45kHz ringing (**Fig. 8**) on top of the square wave. This corresponds to the frequency response peak plotted in **Fig. 1**, and the amplifier remained stable during this test.

FIGURE 7: 10kHz square wave response into 8Ω load.



AE-2403-7

FIGURE 8: 10kHz square wave response into complex load.



AE-2403-8

TABLE 1 Measured Performance

Parameter	Manufacturer's Rating	Measured Results
Power Output (RMS)	200W, 8Ω 300W, 4Ω	205W 8Ω 20-20kHz 313W 4Ω 1kHz
Frequency Response	20Hz – 20kHz power bandwidth	10Hz – 25kHz ±1dB, 4Ω and 8Ω
Total Harmonic Distortion	<0.05%, 20Hz - 20kHz Rated Power	0.015%, 300W 4Ω 0.05%, 200W 8Ω
IMD - CCIF (19+20kHz) MIM (9+10.05+20kHz)	N/S	0.0022% CCIF 0.004% MIM
Input Impedance	>10k	14k at 1kHz
Signal to Noise Ratio	100dB unweighted	
Gain	+27dB, 1.7V sensitivity for full output	28.2dB 4Ω, 28.3dB 8Ω
Power Requirements	600W maximum, <3W Stand-by	
Output Impedance	N/S	0.06Ω 20Hz, 1kHz 0.16Ω 20kHz

CONCLUSION

The Outlaw 200 is a good performer and pretty much bulletproof. While I was performing tests beyond 200W, I accidentally shorted the output while adding a second speaker load. The amplifier protested loudly for a second before its protection circuit tripped. It reset immediately after I corrected my connection error and reset the power switch. It continued to produce lower distortion and higher levels of power than the manufacturer's ratings, so there were no apparent ill effects.

A comparison of the measured results and the manufacturer's ratings is shown in **Table 1**. aX

References

1. *Audio Power Amplifier Design Handbook*, by Douglas Self, p. 36–38, Second Edition, 2000, Newnes.