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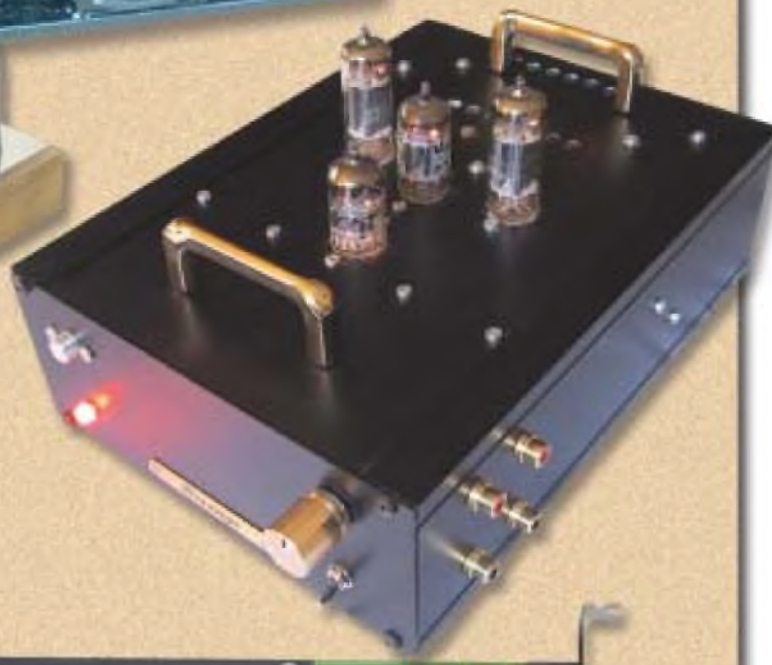

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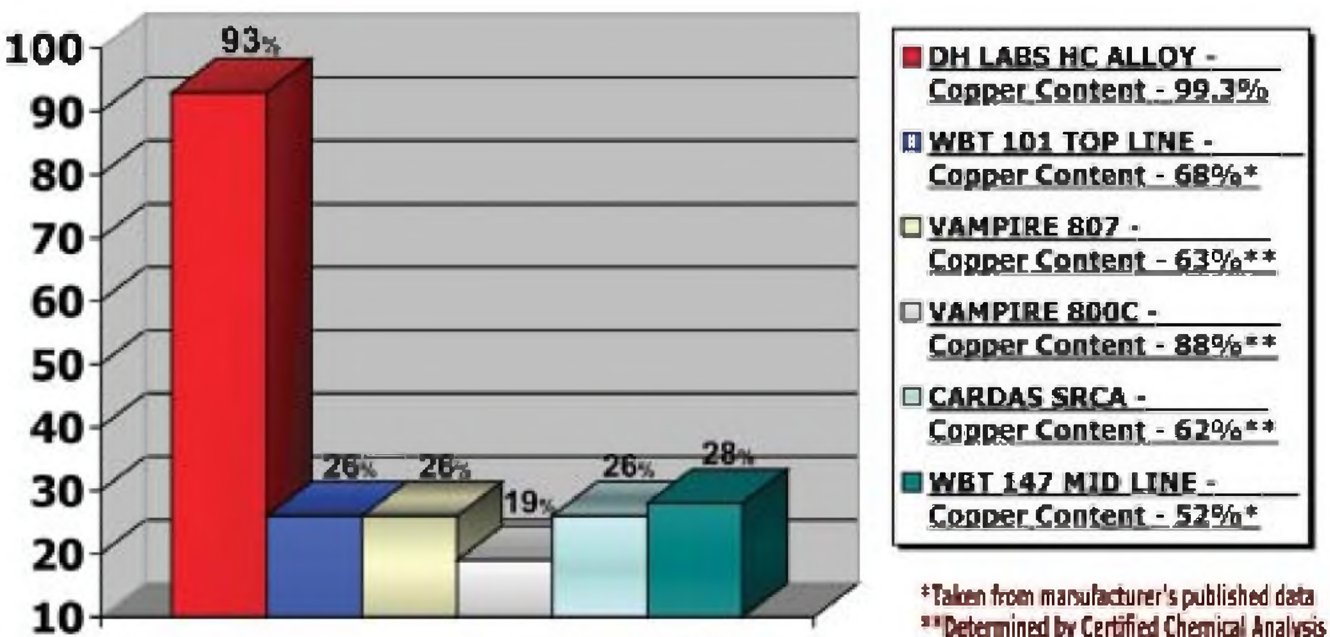

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Audio News

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NEW FROM TDL TECHNOLOGY

TDL® Technology, Inc., announces their Model 601 tunable high-pass/low-pass filter, Model 407 Stereo Phono Preamp, and Model 402 Line-Operated Microphone Preamp. The Model 601 contains tunable 8-pole Butterworth filters, and a microchip PIC microprocessor controls the unit. The Model 407 is especially designed to drive a sound card's line input for restoring 78 RPM records, and the Model 402 microphone preamp is designed for applications where power line operation is preferable to batteries. For more, contact TDL Technology, Inc., 5260 Cochise Trail, Las Cruces, NM 88012-9736, 505-382-3173, FAX 505-382-8810, or download a full data sheet and User Guide in PDF format from the website at www.zianet.com/tdl.

DACT CT 102

DACT introduces the CT102 power supply. This is a ready-made module that comes on a packed PCB with gold plated traces—tested and ready to use. It also includes a (100–250VAC) switch mode mains adapter. CT 102 outputs a stable, noise-free DC voltage, adjustable via jumper to either ±15V or ±20V. Maximum continuous output current is 200mA, but much larger peak currents are available. CT 102 is designed for low-consumption audio electronics circuits. More information about CT102 is available at http://www.dact.com/html/powesupplies.html.

HRS NIMBUS

Harmonic Resolution Systems (HRS) introduces the HRS Nimbus, which utilizes a proprietary elastomer formulation (HRS SPIII) developed specifically for audiophile applications. This formulation is combined with an aircraft aluminum housing, forming a unique design geometry that allows you to "tune" the isolated component to a desired level of performance. The HRS Nimbus can isolate components of any size and weight, making it a highly effective and flexible isolation solution. For more, call 716-667-9896, e-mail info@avisolation.com, or check out the website at www.avisolation.com.

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SYSMATCH "LITE"

A "lite" version of SysMatch software is now available free upon request. Since its introduction and review in 1998 ("Optimizing Your Audio System, Parts 1 and 2," *Audio Electronics*, issues 3 and 5, 1998), SysMatch has become a popular audio optimization software. You can request the freeware and download technical information from www.nexicom.net/~dbhsystems.

Hybrid Solid-State Power Amplifiers

Looking for a better amp design requiring fewer components? Try one of these hybrid models. **By Charles Hansen**

The input stage and voltage amplifier stage (VAS) of an audio power amplifier can consume the lion's share of the component count, because you need only four power transistors for current amplification in a modest (<50W) output stage.

Since an op amp provides both a differential input stage and high overall gain in one package, what are the ramifications of using an op amp as a combined input and VAS in a power amplifier? At least one such amplifier project appeared in *aX*[1], but it retained a fairly high number of discrete transistors between the composite op amp input stage and the output section, perhaps for good reason.

I did not build the four amplifiers shown in this article, but I did run a SPICE simulation to verify that they did perform as audio amplifiers. I provided the circuits, showing waveforms at selected points in the circuitry, as a basis for further experimentation. I selected the feedback values to provide a –3dB frequency response of 2Hz to 86kHz at a gain of 20 (26dB), with a sprinkling of other parts, where necessary, to provide adequate loop stability and avoid excessive phase shift or response peaking.

I simplified the drawings by using diode bias, so in a real-world amplifier the output transistor bias must be optimized with the usual Vbe multiplier bias-spreading transistor. I also do not show components such as output device protection circuits, the zobel network for the speaker, RF oscillation suppression, and power supply filter and decoupling caps. Output devices are yours to select, as are the op-amps unless otherwise specified.

HYBRID OP AMP/POWER TRANSISTOR AMPLIFIERS

Figure 1 shows the simplest circuit using an op amp for the gain stage, with the simulation part values in *Table 1*. The emitter-follower output transistors are inside the op amp feedback loop. The same supply voltage as the op amp powers the output stage, so the absolute maximum power supply voltage specified for the op amp and its output voltage swing limits the output power.

Most audio-quality op amps have an absolute maximum supply limit of ±18V DC. Applying a derating factor of 90%, the maximum safe supply reduces to about ±16V DC. Higher voltage op amps are available, such as the older NE5532 and NE5534A, which have ±22V DC absolute maximum limits, allowing almost ±20V DC for the rails. In what is a happy coincidence, one of the highest-quality audio op amps, the OPA604 and its dual OPA2604, have the highest absolute maximum power supply limit of ±25V DC, allowing a safe rail supply of ±22.5V DC.

Since the OPA604 output swing is limited to 3V below the supply rails, the voltage drive to the output devices is limited to ±19.5V, or about 14V RMS at the speaker. This delivers more than 23W into an 8Ω load, and may serve your audio purposes quite well. Indeed, the Monarchy Audio SM-70 Pro 25Wpc power amplifier uses a hybrid OPA2604 IC op amp/discrete MOSFET design[2].

The circuit in *Fig. 1* is called a current-boosted op amp. If you need more output power, you will require additional voltage gain beyond the limited output voltage swing from the op amp.

TABLE 1
SIMULATION PART VALUES FOR *FIG. 1*

C1	1µF
C2, C6, C7	100pF
C3	10pF
C4	47µF
C5	39pF
R1	1k
R2, R8	47k
R3, R6	1k5
R4	22
R5	100
R7	2k2
R9, R10	47
R11, R12	0R33

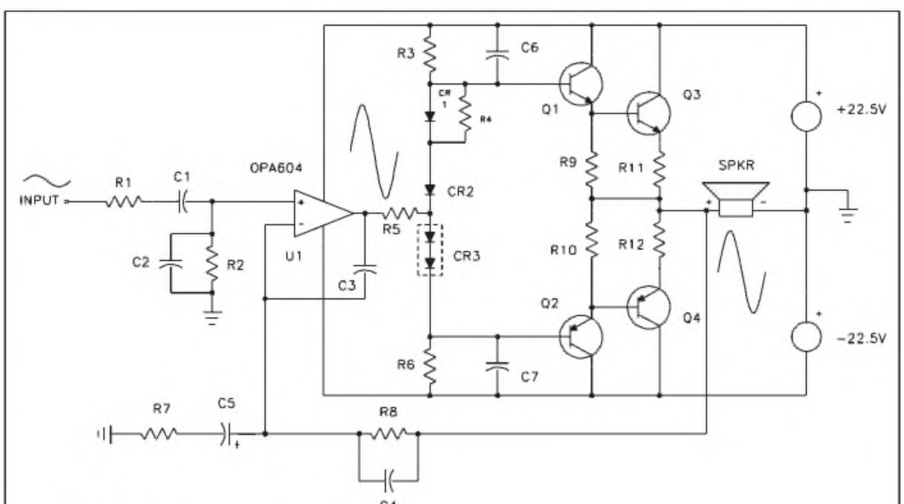

FIGURE 1: Current-boosted op amp.

Behind the Scene

Dr. Joseph D'Appolito has been working as consultant for Usher Audio since early [illegible]. A world renown authority in audio and acoustics, Dr. D'Appolito holds BEE, SMEE, EE and Ph.D. degrees from RPI, MIT and the University of Massachusetts, and has published over 30 journal and conference papers. His most popular and influential brain child, however, has to be the MTM loudspeaker geometry, commonly known as the "D'Appolito Configuration," which is now used by dozens of manufacturers throughout Europe and North America.

Dr. D'Appolito designs crossovers, specifies cabinet design, and tests prototype drivers for Usher Audio, all from his private lab in Boulder, Colorado. Although consulting to a couple of other companies, Dr. D'Appolito especially enjoys working with Usher Audio and always finds the tremendous value Usher Audio products represent a delightful surprise in today's High End audio world.

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CURRENT AND VOLTAGE AMPLIFIERS

The hybrid amplifier in *Fig. 2* is from a Linear Technologies application note[3]. The app note circuit is an inverting amplifier designed to deliver ±120V into 6k, so I modified it for non-inverting gain and the higher current, lower voltage output a loudspeaker requires. Common-emitter transistors Q1 and Q2 provide the voltage gain to the emitter-follower output stage Q3 and Q4, which provides current gain.

Q1 and Q2 cause a signal inversion, so the op amp is configured as an inverting amplifier to make the overall power amplifier non-inverting. The negative feedback connection looks strange, because it needs to be brought to the non-inverting input.

Since the op amp power supply rails no longer limit the output voltage swing, you can employ any high-quality audio op amp. The added complication is the need for a lower voltage supply for the op amp. You can easily obtain this from the ±40V output supply via linear regulators.

When you operate the output devices as source-followers or emitter-followers, as in *Fig. 1*, there is no possibility for output stage oscillation, since the gain of any follower stage is always less than one. With a common-source or common-emitter amplifier, the output device gain and phase shift provide an increased opportunity for oscillation or ringing, so careful frequency compensation takes a high priority.

AN UNUSUAL DESIGN

Figure 3 shows a unique method, employed by Velleman in their K4020 stereo power amplifier[4]. Again, the circuit is greatly simplified, and you could easily implement the MOSFET output stage with bipolar transistors. The preamplified input signal is sent to the non-inverting input of U1, a TL061 low-power op-amp. This is not a low-noise, low-distortion device like the TL070-TL074. I selected it for its low power supply current (250µA versus 2.5mA for the TL071).

Here's where the design becomes unusual. The ±15V supply for the TL061 comes from the emitter of an NPN (Q1) and a PNP (Q2) transistor whose bases are connected to +16V and −16V supply rails, respectively. Each transistor has a collector load resistor, connected to the high-voltage MOSFET supply rail, in what resembles a cascode connection. The emitter current in each transistor is equal to the TL061 supply pin current, which, in turn, is a composite of the op amp quiescent current and the amplified audio signal into loading resistor R5 at the output of the TL061.

An FET-input op-amp, whose non-inverting input is connected to the collector of its "cascode" transistor, drives the gate of each output MOSFET. The

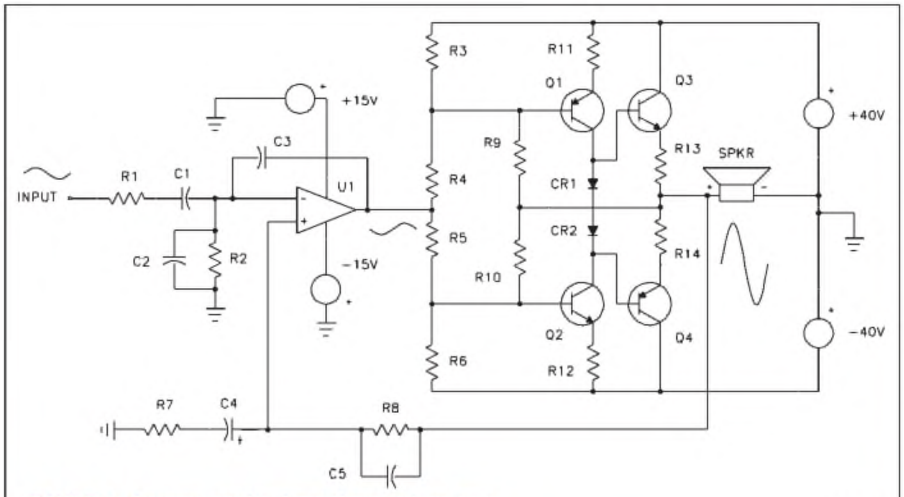


FIGURE 2: Current- and voltage-boosted op amp.

FIGURE 3: Boosted cascoded op amp.

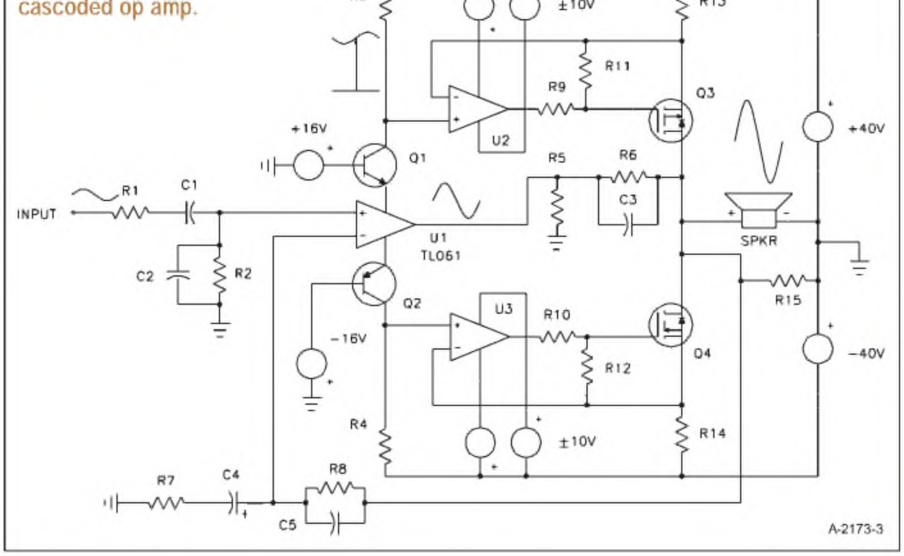


TABLE 2
SIMULATION PART VALUES FOR *FIG. 2*

C1	1µF
C2, C6, C7	100pF
C3	10pF
C4	47µF
C5	39pF
R1	1k
R2, R8	47k
R3, R6	1k5
R4	22
R5	100
R7	2k2
R9, R10	47
R11, R12	0R33

TABLE 3
SIMULATION PART VALUES FOR *FIG. 3*

C1	1µF
C2	100pF
C3	10pF
C4	47µF
C5	39pF
R1	1k
R2, R8	47k
R3, R4	Hundreds of ohms, depending on your selected output bias point
R5	1k5
R6	8k2
R7	2k2
R9, R10	100
R11, R12	10k
R13, R14	0R33
R15	220

split supply for each op amp floats and is referenced to the high-voltage supply rail for its respective MOSFET. Local feedback comes from the source terminals, across R13 and R14 source resistors. Thus each op amp drives its MOSFET gate such that its source operates at the same voltage seen at the "cascode" transistor collector. At idle, the 250µA supply current to the TL061 sets the bias in each MOSFET.

Each common-source MOSFET produces gain in the output stage according to the ratio of the source resistor to the speaker load impedance. Resistor R15, in case of an open speaker terminal, also loads the MOSFET drains.

As with the *Fig. 2* amplifier, the common-source stage gain and phase shift provide an increased opportunity for oscillation or ringing. There are two feedback loops from the output stage back to the TL061. Conventional overall negative feedback is taken from the output drains back to the inverting input of U1. Then there is additional feedback through a parallel network consisting of R6 bypassed by C3, from the output drains to load resistor R5.

The obvious disadvantage of *Fig. 3* over *Fig. 2* is the need for three op amps and four separate split power supplies.

Normally, in a complementary-symmetrical split-supply amplifier, the negative terminal of the speaker is grounded (*Figs. 1–3*), while the amplifier output stage drives the positive terminal. This is called a driven speaker configuration. Since the power section always has less than unity voltage gain, the output power in a driven speaker design sourced by an op amp will be limited by the relatively low swing of the op amp (*Fig. 1*). However, some modern guitar amplifiers[5–7] use op-amps for the front-end voltage gain stage and obtain added voltage gain with an interesting configuration.

DRIVEN SUPPLY

In *Fig. 4*, the ±40V DC power supply common point is left floating. If you ground emitter resistors R12 and R13 of complementary output transistors Q3 and Q4 and connect the speaker + to the floating power supply center-tap, the voltage across the speaker can swing by approximately the full ±40V DC power supply voltage. This is called the driven supply configuration.

During the positive input half-cycle, transistors Q2 and Q4 are turned on (op-amp U1, connected as an inverting amplifier, has a negative output). Current flow through Q4 causes the center-tap of the floating power supply to swing positive, producing a positive half-cycle across the speaker. During the negative input half-cycle, Q1 and Q3 turn on, with opposite results.

With a ±40V DC power supply, the peak speaker voltage at full power can be roughly ±37V, driving 26V RMS or 84W into an 8Ω load. Yet the output of U1 need only swing by V_{out}/gain, or ±1.8V for a voltage gain of 21 (26dB). This completely eliminates the need for a high voltage VAS transistor.

It does complicate the design of the op-amp power supply, however. The ±40V DC supply is not referenced to ground, but the op-amp ±15 supply must be. Either a separate transformer winding is required for the ±15V DC supply, or it must be "sampled" and stored in large capacitors from the high voltage supply rails as they swing up and down at audio frequencies.

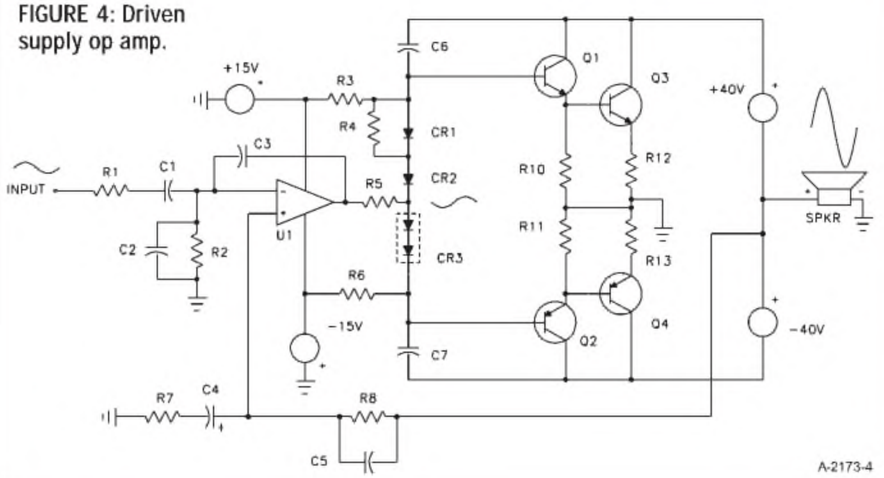
Although the output stage appears to be an emitter-follower, it is a common-emitter amplifier since the emitter resistors are grounded. The ratio of the speaker impedance to the emitter resistors realizes the voltage gain. The earlier caveats concerning the care needed to ensure stability apply here as well. As with the *Fig. 2* amplifier, feedback must return to the non-inverting input of U1.

I hope these hybrid op amp/discrete power amplifier topologies give you some ideas for your own designs. ❖

TABLE 4
SIMULATION PART VALUES FOR *FIG. 4*

C1	1µF
C2, C6, C7	100pF
C3	10pF
C4	47µF
C5	39pF
R1	1k
R2, R8	47k
R3, R6	1k5
R4	22
R5	100
R7	2k2
R9, R10	47
R11, R12	0R33

FIGURE 4: Driven supply op amp.



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3. "Power Gain Stages for Monolithic Amplifiers," Jim Williams, Linear Technology Corp. Application Note AN18, Fig. 9, p. 7. See www.linear-tech.com.

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5. U.S. Patent 5,197,102 "Audio Power Amplifier with Frequency Selective Damping Factor Controls," 1993, Jack Sondermeyer; Peavey Electronics Corp. See www.peavey.com.

6. U.S. Patent 5,524,055 "Solid State Circuit for Emulating Tube Compression Effect," 1996, Jack Sondermeyer; Peavey Electronics Corp.

7. U.S. Patent 5,796,305 "Amplifier Arrangements with High Damping Factor," 1998, Jack Sondermeyer; Peavey Electronics Corp.

NOTE

Peavey mentioned that although they do utilize the "driven supply" circuit in many of their amplifiers, and have for years, they were not the first to do so. Jack Sondermeyer said he was not the person to come up with the idea. In talking to some of their folks, they first remember this type of circuit in an early Sunn amplifier, circa 1970 or so.

The DR5 Horn

The latest horn in this author's collection packs plenty of power, but is compact, so it won't strain your back or your wallet. **By Bill Fitzmaurice**

PHOTO 1: The DR5 Horn.

For a number of years I've been using dual 6½″ woofer direct radiator/piezo horn tweeter cabinets for stage monitors. They aren't particularly loud, but they're cheap to build, easy to carry, and—at the stage volumes my band plays at on the "Rock and Roll Seniors Tour"—they do the job. But every once in a while younger musicians use my PA rig, and they often play with stage volumes so loud that they can't hear the monitors.

I'd tell them they would hear the monitors just fine if they'd turn their axes down, but musicians under the age of 30 don't tend to heed sage advice from their elders. Fair enough, because two decades ago I didn't either. To accommodate these Pete Townshend ("Could you speak up please? I'm almost deaf, you know") wannabes, I needed monitors with some more oomph. But because it will be me hauling them around, they still need to be small and lightweight.

TRIAL AND ERROR

When two 6½s per cabinet aren't enough, you might try two eights or a ten. But that means bigger cabinets and more power to drive them. A better way is to gain sensitivity, and thus power, by horn loading.

Traditional horn theory says you'd need to use a long, ungainly straight horn, because a compact folded horn wouldn't provide the necessary midrange. My own theory, proven by my DR horns, is that folded horns can provide midrange as good as a direct radiator. What I did not yet know was how small I could make a folded horn and still achieve a significant increase in sensitivity.

There are a number of computer programs you might use to answer that question, but those programs are based, for the most part, on the pioneering work of Keele and Leach—which now dates back as many as 30 years. They use throat, mouth, and taper formulas that I have improved upon, and incorporate the parameter of mass rolloff, which my cabinets have proven nonexistent. For the heck of it, I tried a couple of those programs; they said that what I planned to do couldn't be done, but they also said that my DR8, DR10, DR12, and so on, couldn't do what they did. I saw no reason why I couldn't make even a pip-squeak folded horn perform well.

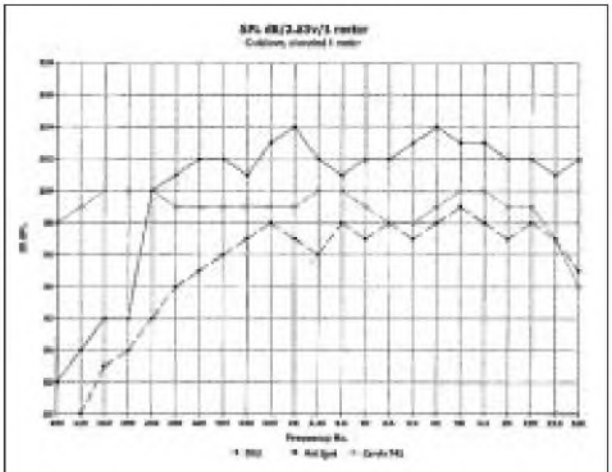

FIGURE 1: Response of Pyle MH516 driver prototype.

Traditional theory states that horns are, by nature, larger than direct radiator cabinets utilizing the same size driver. On this point, traditional theory is correct. My old boxes were about half a cubic foot apiece. I was willing to go somewhat larger to gain more output, but when I tried to design a horn for even a single 6½, I couldn't come up with anything smaller than a cubic foot.

That may sound small enough, but when you're lugging around six monitors, two PA bins, a bass cabinet, and four rack cases of gear in a station wagon, even a cubic foot is too big. I needed to hold the size down to no more than three-quarters of that. The only way I could do so was by loading it with a five.

It just so happened that I had a bunch of old Pyle® fives lying around, so all I had to risk in building an experimental prototype was some plywood and some time. A week later, I fired up the first DR5 (*Photo 1*), and the performance far exceeded my expectations. Look at *Fig. 1* and you'll see why.

A single Pyle MH516 driver (replaced now in their line by the similarly spec'd PDMW5, available at Parts Express for $16) with nominal 88dB efficiency normally wouldn't be up to the job of live-sound vocal monitoring, but when mounted in a DR5 cabinet the resulting 102dB average SPL from 250 to 3.2kHz is more than enough to do the job. As soon as I'd tested the woofer in what I believe to be the world's smallest folded horn (should I contact Guinness?), I went online to Parts Express to order the tweeters (CTS model KSN1036, $7 each) and hardware I'd need to complete it and five more just like it.

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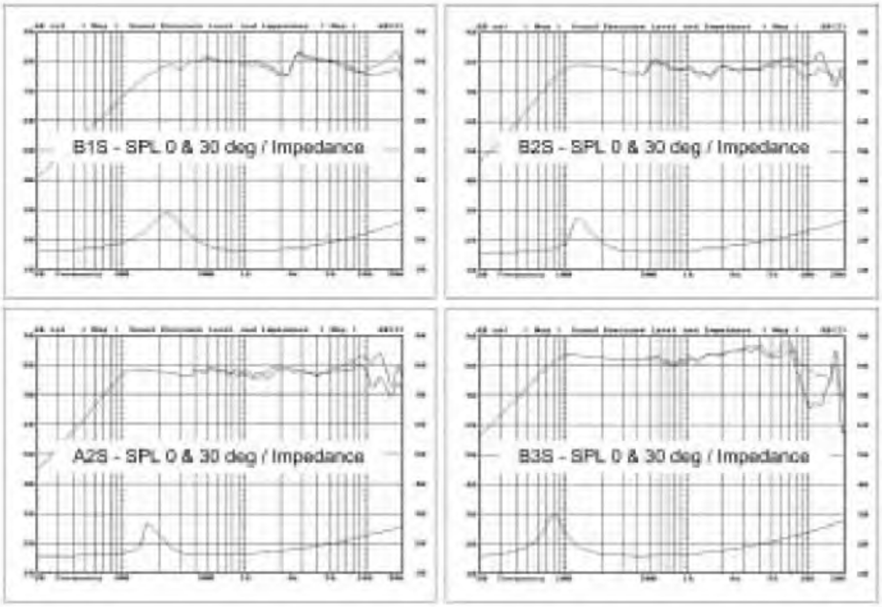

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DESIGN

The design of the DR5 is similar to that of its cousins: the DR8, DR10a, et al. One change is the angle of the top and bottom. Viewed from the side, the box is trapezoidal (*Fig. 2*), eliminating two parallel surfaces that might cause out-of-phase reflections. The throat horn sides are not parallel for the same reason. Since the side panels are so small, there are no side braces, nor is the small cross-section throat horn divided. The radius of the back halves is too small for even ⅜″ plywood to bend tightly enough, so they are fashioned from 6″ schedule 40 PVC pipe.

The major deviation from previous DRs is in the duct location, which, in prior models, was mounted either on the tweeter baffle or through the mouth horn sheathing. Initial calculations showed that a duct long enough to tune the box would be too long to fit into any of the usual locations. Another problem was that the throat size for maximum efficiency was so small that the horn would have been very difficult to construct.

I solved both problems by using a 1″ schedule 40 PVC "tee" to direct the duct output into the throat of the horn (*Fig. 3*) while reducing throat area. The mass of the horn's air column adds to the air mass within the duct, allowing a shorter duct (with the disadvantage of giving little, if any, ability to tune the system, as the horn mass is the dominant one and cannot be altered).

Construction is also similar to other DR horns, though the small size and low internal stresses make even ½″ plywood unnecessary for many parts. I built the prototype mostly from ½″ plywood because I had it on hand, but for the rest of the "fleet" I used ⅜″ Baltic birch wherever I could. The only parts that demand ½″ plywood are those that must hold screws. The horn sheaths require ⅜″ Baltic plywood, but the pieces are so small that if you've built any other DR horns you may have enough left over for a bunch of DR5s.

All listed dimensions are approximate, being dependent on the true thickness of the materials used. I prefer Baltic birch for ⅜″ plywood parts; for ½″ plywood, spruce will do, as long as it has at least five plies. When laying out ⅜″ Baltic birch for horn sheaths, flex the sheet to determine the easier-bending axis.

Some steps call for temporarily screwing parts together— be sure you eventually fill these holes. You may use traditional wood fillers, but those have long

PHOTO 2: Phillips, Torx, and Square drive screws.

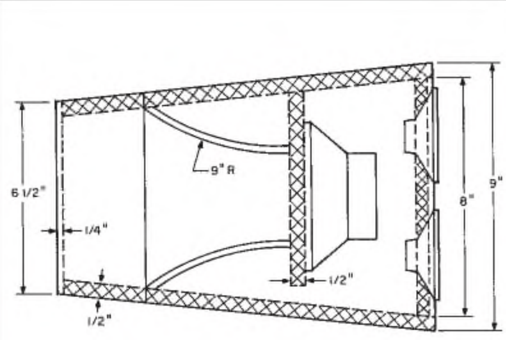


FIGURE 2: DR5 side view—simplified. Overall length dimension: 13¼″.

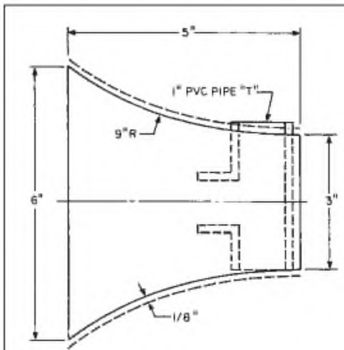


FIGURE 3: Throat horn side view.

PHOTO 3: Simul-cutting throat horn sides.

PHOTO 4: Cutting the throat horn jig.

PHOTO 5: Beginning the throat horn assembly.

PHOTO 6: Aligning the throat horn parts.

PHOTO 7: Attaching throat horn supports. Note right angle ratchet driver.

PHOTO 8: Trimming throat horn assembly with a panel-cutting jig.

set-up times and tend to shrink. Two-part fillers that don't shrink and cure within 15 minutes are available. Aside from color, they closely resemble auto-body filler, and with good reason: that is precisely what they are, though they cost much more. As long as you're not planning on a natural wood finish, "bondo" is the logical choice.

I drilled holes through the interior parts to minimize flat surfaces within the box and open up what would otherwise be sealed chambers. I glued and screwed the joints for the most part, though you can use brad nails in some areas. I prefer caulking gun-applied construction adhesive, especially the urethane type, which expands as it cures to fill voids in less-than-perfect joints. This expansion literally forces the adhesive into the pores of the wood.

PHOTO 9: A PVC pipe "T."

PHOTO 10: PVC "T" glued into the horn throat.

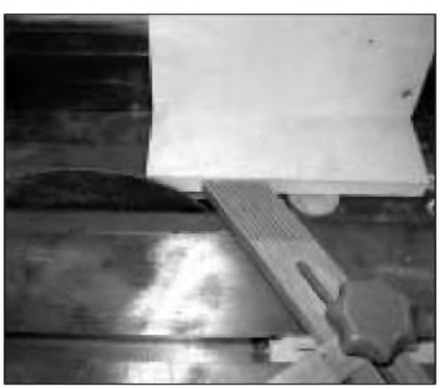


PHOTO 11: Jig and finger-board for cutting PVC lengthwise. Note abrasive table saw blade.

Urethane is less than perfect for joining PVC, so for those joints use a traditional solvent-based adhesive.

Use a screw pilot/countersink bore to prepare all screw holes. As far as the screws are concerned, there seem to be more choices available every day. Phillips-head drywall screws have been joined by square-drive and now torx-drive variants (*Photo 2*). I have always wondered why Phillips bits and screws never seem to fit together very well, and the other options are both far better in that respect. I prefer the square-drive, but the torx is still better than Phillips.

Note when viewing the figures, for the sake of clarity, not all parts are shown. For each step of construction compare the appropriate figure and the accompanying photos.

PHOTO 12: Slicing PVC. Note blade does not penetrate fully at the ends of the cut.

PHOTO 13: Tracing PVC halves onto cabinet top and bottom.

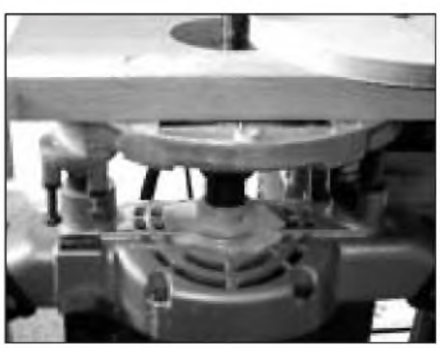


PHOTO 14: Trimming radiused parts with a table-mounted router.

CONSTRUCTION

The first step in construction is cutting out the throat horn sides (*Fig. 3*). Cut the leading and trailing edges at a 5° angle. Use ½″ plywood here, screwing two rough-cut pieces of plywood together, cutting both of them at the same time to ensure they are identical (*Photo 3*). (If you plan on building a fleet of DR5s, and you have a router, make one throat horn side to serve as a pattern and then use a pattern-following bit to make as many duplicates as you need.)

Cut a throat alignment jig (*Fig. 4*) from scrap ½″ or better plywood (*Photo 4*). Clamp it in place down the center of one throat horn side (*Photo 5*), and screw the throat side to it, repeating with the other throat horn side. Cut the throat horn sheaths from ⅛″ Baltic birch, attaching them to the assembly with adhesive and screws while clamps hold them in place (*Photo 6*). Make the sheaths about ½″ too long, leaving the excess overhang at the mouth end.

Unscrew and remove the alignment jig. Cut the horn supports from ½″ plywood, with the edge mating the horn cut to a 5° angle. Make these a bit too long as well, with about ¼″ extra at either end. Drill a few 1″ holes in them, and attach them to the assembly, again clamping them in place first (*Photo 7*). After the adhesive has set, run the assembly across a table saw using a panel-cutting jig to square and trim it to finished size (*Photo 8*), cutting the top and bottom edges at a 5° angle.

Mount the PVC tee duct (*Photo 9*) ½″

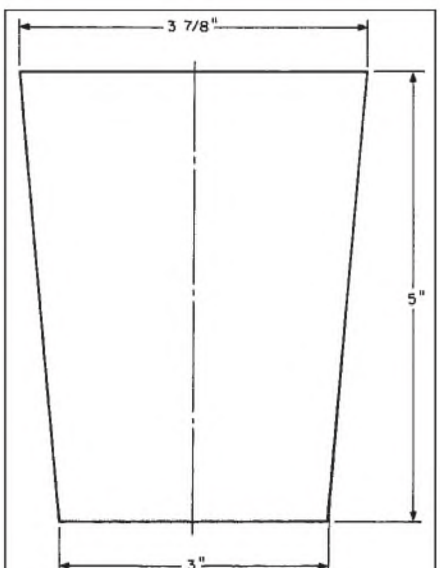


FIGURE 4: Throat alignment jig.

from the horn throat opening. After drilling a hole through one of the throat horn sheaths, push one end of the tee through the hole and rest the other on the opposite sheath (*Photo 10*). First tack the tee in place with hot-melt glue, then thoroughly fill all the joints, finally filling the lower portion of the tee so that it does not act as a resonant tube (*Fig. 3*).

Halving 6″ PVC for the cabinet back requires a jig made of scrap plywood. This consists of a 3″-wide piece and a 6″-wide piece screwed together at a right angle. When running through the table saw, hold the jig tight against the rip fence with a finger-board (*Photo 11*).

Cut the PVC to rough length first; about 8″ will do. Clamp the PVC to the jig and run it through the saw. At the beginning and end of the cut, lower the blade so that it does not quite cut all the way through, otherwise the PVC could close down on the blade (*Photo 12*).

You may use a toothed blade to cut PVC, but an abrasive blade is better. After the first cut, re-clamp the PVC to cut the opposite side, finishing the cuts with a hand saw or utility knife to form two crescent-moon shaped pieces. Use these to trace the outline of the top/bottom (*Fig. 5*) onto a piece of 5/8″ or ½″ plywood (*Photo 13*), in so doing determining their actual finished width.

As with the horn sides, copying the top and bottom for multiple cabinets is a simple job if you use a router and a pattern-following bit. The job is even easier if the router is mounted in a table, or on a board clamped to a workbench or table saw extension (*Photo 14*). Cut the straight edges of the pattern piece to size on the table saw, and then use a saber saw to rough-cut the two arcs of the back joint within about a quarter inch of the line.

Drill 3/16″ holes through the pattern at the center points of the two arcs. One arc at a time, screw the pattern to the

PHOTO 15: An alternate view of the router in action.

PHOTO 16: Making copies with a pattern-following bit.

PHOTO 17: Using a jig to align the top and bottom for assembly.

router table/board so that when you rotate it into the bit, the router will finish the arc with a perfectly round cut (*Photo 15*).

To make perfect copies, first rough-cut another piece of plywood a bit oversize, screw it to the pattern, and then use the pattern-following bit to trim the copy to finished size. If you have a long bit, you may make two copies at a time from ½″ plywood, three at a time from ⅝″ (*Photo 16*); after trimming with the router, you must square the area where the two back halves meet with a saber saw.

Draw the positions of all mating parts on the top and bottom, and mark the porthole cut line on the bottom. Starting with a plunge cut, use a saber saw to cut out the porthole. If you're building a batch of cabinets at once, make sure you mark the portholes and their respective bottoms to make it easy to match them later.

PHOTO 18: Using clamps to align parts before screwing in place.

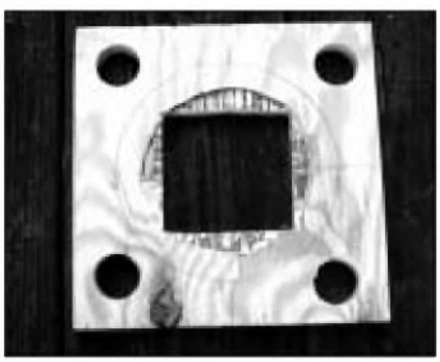


PHOTO 19: The baffle.

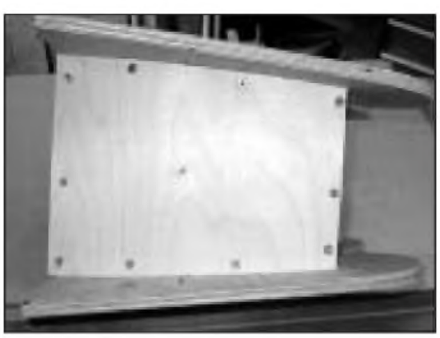


PHOTO 20: A horn sheath in place.

Assembly is easier if you build a jig (a piece of scrap plywood with two strips of ¾″ × ¾″ stock or plywood screwed to it). Clamp the top and bottom to the jig to hold them in position at the proper angle for assembly (*Photo 17*). Cut the tweeter baffles from ½″ or ¾″ plywood—the top and bottom edges at 5°, the sides at 30°, angles.

Using clamps to hold the parts in alignment, attach the tweeter baffle and throat horn assembly (*Photo 18*). Cut the baffle from ½″ or ¾″ plywood with the side edges at 15°, the top at 5°. (Note: The photos show the baffle extending all the way from top to bottom. Inserting the driver is much easier with the baffle left an inch shorter—I did so with the subsequent copies.)

Holding the baffle in mounting position, reach through the throat horn, tracing the throat opening on it. Remove the baffle, cut the throat opening, drill four venting holes on the baffle corners, and rout away a ⅛″ trough to prevent cone-slap (*Photo 19*). Install the baffle.

Cut the horn braces using the technique used on the top and bottom for making multiple copies. The braces are bisected to accommodate the baffle location and installed on the top and bottom. Cut the horn sheaths from ⅛″ Baltic birch, making them about a half-inch too long to allow extra material as fitting may require. Install the sheaths (*Photo 20*). After the adhesive has set, trim and sand the sheaths flush to the horn supports and tweeter baffle.

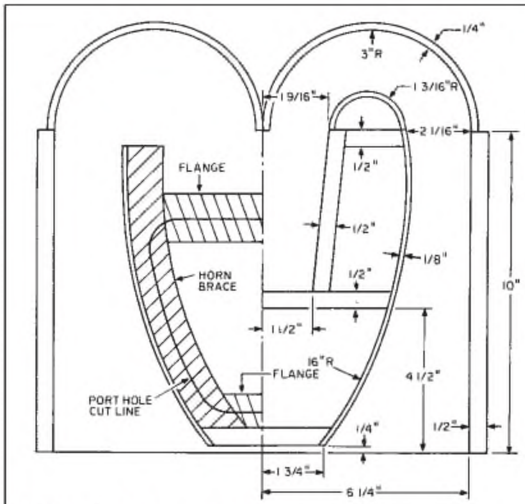


FIGURE 5: Cutaway top view.

Bisect an 8″ length of 1½″ PVC pipe, using the same method as with the 6″ PVC. Trim the two halves to size with the table saw blade at a 5° angle, again using dead reckoning to determine the exact length. Attach the two reflectors to the assembly with hot-melt glue, being sure that all joints are airtight (*Photo 21*).

Trial-fit the two back halves; you can chamfer them for a better fit using the table saw with a panel-cutting jig and abrasive blade (*Photo 22*). Cut the back halves slightly long, again with the blade at 5°, and install. Liberally fill the joint of the two halves with adhesive.

Cut the sides from ⅝″ or ½″ plywood and install them, butt-jointed with the back halves. You may use screws, brads, or paneling nails here. Run the assembly across a table saw atop the panel jig to trim the excess side material (*Photo 23*), and then sand all joints, first flush, and then slightly rounded over.

Hole-saw two 2½″ holes in their baffle for the tweeters (*Photo 24*), saving the resulting discs. Attach one disc to the inside of the porthole cover, centered over the throat horn, making sure it does not hit the baffle when installed. Attach the other disc to the outside of the cover, offset about a half-inch toward the edge.

Using a drill press with the table set at a 30° angle, clamp the cover to the table, using a plywood spacer to keep it flat (*Photo 25*). Use a ⅝″ spade bit to drill not quite through the stacked discs and cover, producing a socket for mike stand mounting (*Photo 26*). Be sure to seal the hole in the interior disc. Drill holes in the cover above the driver for jacks, making sure they don't hit the baffle. I recommend two jacks parallel wired for daisy chaining, using deep jack sockets (*Photo 27*).

Use plywood scraps to complete the porthole flange, trimming it as required to allow the woofer to slide into place (*Photo 28*). Apply the finish of your choice. I finished my boxes with spray-on truck bed liner, which looks good, is durable, and is easily repaired by simply spraying on some more. If you go this route be sure to putty over the screw heads with "bondo" and sand the box well to minimize wood-grain.

Install the woofer, screwing it in place by reaching through the tweeter

PHOTO 21: The PVC reflectors in place.

PHOTO 22: Chamfering PVC edge with the paneling jig and abrasive blade.

holes. The driver magnet will pull screws off the screwdriver, so secure them to the driver with a dab of hot melt glue, which will hold a screw well enough to get it started but will pull off easily afterwards. Loosely—but thoroughly—stuff the cabinet voids with polyfill, keeping the rear of the woofer cone and the duct opening clear.

Install the tweeters, caulking the flanges. The tweeters are a bit wider than the baffle, but are easily sanded down to fit nicely.

Wire the drivers, all in parallel/in phase, to the jacks. CTS recommends a 20Ω resistor in series with piezo tweeters to prevent possible high-frequency insta-

PHOTO 23: Rough-trimming the cabinet side.

bility in the power amp. Because this also pads the tweeters down by at least 3dB, I haven't bothered to do so for years, with no ill result. If you wish to be safe but don't wish to overly pad down the high end, use a 4Ω 20W resistor in series with the paralleled tweeter pair.

Weather-strip the flange and screw the porthole cover in place. Finally, attach a handle to the top and put thick rubber feet on the bottom.

LISTENING RESULTS

Realistically, how much power can you really get from a ¾ft^3 box? For vocal monitoring, the answer is "plenty." *Figure 6* shows the DR5 compared to two traditional-style monitors.

PHOTO 24: Front view of completed raw cabinet.

PHOTO 25: Drilling a mike stand socket.

PHOTO 26: A DR5 on a mike stand.

The first is the Galaxy Hot-Spot™, a dual 5″ driver industry standard. Nominal SPL at 2.83V is 98dB, but that's at 4Ω impedance, so you can use only two per amp channel. The DR5 is 4dB louder for the same voltage input, 7dB louder for the same wattage, and with 8Ω impedance you can use four of them per amp channel. The Hot-Spot uses full-range drivers mounted side by side, which hurts horizontal dispersion, while the vertically arrayed tweeters of the DR5 provide superior horizontal high-frequency dispersion.

The third plot on the chart is for a typical floor wedge, the Carvin™ 742, loaded with a 12 and a horn tweeter. If you're like me and are into playing music, but not lugging equipment, four DR5s take up about the same amount of space and weigh less than one 742.

PHOTO 27: Inside view of the porthole cover.

PHOTO 28: Woofer in place. Note trimming of porthole flanges to allow woofer passage.

PLYWOOD PARTS LIST
SIZES ARE APPROXIMATE, IN ORDER OF ASSEMBLY.

1. Throat horn sides 5″ × 6¼″
2. Throat horn sheaths 5″ × 5½″
3. Horn supports 1¾″ × 7″
4. Top, bottom 13″ × 14″
5. Tweeter baffle 4″ × 8″
6. Baffle 7″ × 7½″
7. Horn braces 2″ × 12″
8. Horn sheaths 8″ × 10½″
9. Sides 9″ × 10″

You can build six DR5s for about the price of one 742.

Performance-wise, below 250Hz the Carvin is obviously better, which is fine if you put bass or kick drums through your monitors. The thing is, unless you are playing on a very large stage or outdoors, you don't want bass or kick drums in the monitors, because that makes it harder to hear vocals and guitars. (Note: always lower the monitor feed EQ sliders below 125Hz to prevent driver over-excursion, preserve amp headroom, and keep the sound from being "muddy." Also, engage the power amp high-pass filters if it has them.) Above 250Hz the DR5 is more sensitive, and when mounted 3–4′ above the stage on a mike stand, it is effectively another 6dB louder still than a wedge at floor level.

In fact, a DR5 about a meter away from your ears with 10W input will deliver 112dB output. In theory you could drive one with 100W for 122dB output. I'd strongly recommend against that, unless you really strive to become the next Pete Townshend—medically speaking, that is. If you can't hear your vocals at 112dB, you're probably well on the way to permanent hearing loss, and need to turn things down, not up. Sage advice, indeed. ❖

FIGURE 6: Comparative responses of DR5, Hot Spot, and Carvin 742.

The DC Revenge E182CC Amplifier

This article describes a novel direct-coupled single-ended triode amplifier, using readily available indirectly heated tubes, a simple power supply, and local end tube feedback. It offers sonic quality comparable to constructions using directly heated tubes and costing an order of magnitude more. **By Mauri Pännäri**

I developed this amplifier after four years of listening to a self-made 300B design based on original British STC4300A tubes through Tango U-808 transformers, with the C3m as a driver. Although I was very satisfied with its musical quality and especially with the honey midrange, occasional listening sessions with good tube push-pull or transistor designs revealed a lack of power or fitness at the lowest bass end, a well-known phenomenon to SE enthusiasts.

Without a doubt, I could have corrected this by implementing overall feedback or heavier end transformers, but the feedback solution would have meant returning to old starting points. A later change to KR300BXLS tubes made a clear overall sonic improvement and also somewhat sturdier low end. For general interest, this is clearly a more dynamic tube than the old STC4300A.

PROTOTYPE

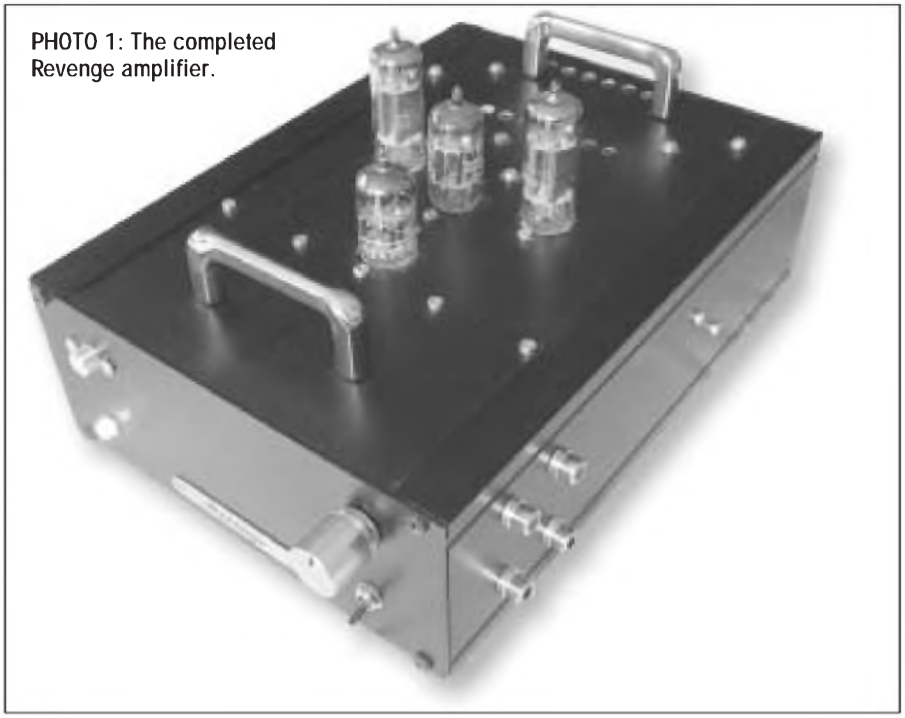
Overall feedback in audio amplifiers has its known merits and questionable drawbacks. Local feedback around output tubes has been an accepted means used for a long time in push-pull designs—e.g., by Quad, MacIntosh, and Köykkä here in Finland, even in 1:1 ratio. This inspired me to try it in a SE design. Another goal was to achieve quality and power compared to good SET direct-heated designs with ordinary cheap tubes of the European E-series or corresponding USA-types, to diminish the enormous hype and mystique around direct heating.

Looking at what I had available, the professional E182CC, long-life dual triode, originally meant for computer use, showed interesting properties. With a mu of 25 and high-current Ri of 1.6kΩ, each triode half withstands 4.5W of loss power, 8W together. In addition, you can drive the tube grid positive with sufficient current (several milliamperes) as specified. With the halves coupled in parallel, you can obtain Ri of 800Ω, comparable to 300B. The operating point could be somewhere around 160–180V/45mA to 1.6–2.5kΩ.

The low anode voltage operational point and the 6.3/250V tube transformer I had on hand allowed me to lift the cathode to about 100V and fit the grid directly to the anode of the driver. One driver tube should handle the gain, so I decided to use a favorite of mine: the ECC81, or actually the telephony type of it, Brimar 33A/101K, with its high mu and good sound, proven when I prototyped it earlier with my 300B designs.

I had no suitable output transformers on hand, so I decided to try my own design. Applying three paralleled identical 8Ω secondary windings, I wound the primary between them as halved, a proven minimalist way of construction. To get some power output, I decided to use a low primary impedance of 2 × Ri of the tube, 1600Ω in this case, at the cost of more initial distortion, cancelled by the local NFB here. The primary inductance targeted to 10H. This construction was wound on a 20VA normal grain oriented silicon steel stack with an air gap by a local private winding

PHOTO 1: The completed Revenge amplifier.



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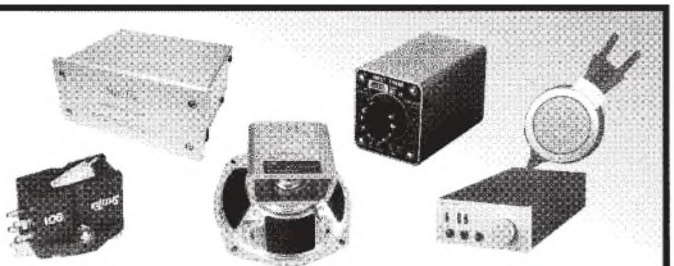

■Cartridge

Model	Price (US$)
Denon DL-102 (MONO)	150
Denon DL-103 (STEREO)	200
Denon DL-103R (STEREO)	250
Denon DL-103 PRO (STEREO)	350
Shelter Model 501 II (CROWN JEWEL REFERENCE)	750
Shelter Model 901 (CROWN JEWEL SE)	1,400

Postage (Air Economy)

Area I $18	China, Korea, Hong Kong, Taiwan
Area II $22	Singapore, Malaysia, Indonesia
Area III $27	North America, Oceania, Europe
Area IV $34	Africa, South America

These Area I ~ IV are for all products except book.

■Japanease Audio Book

Postage $15

Title		Price(US $)
Attractive Tube Amps Vol. 1&2	(Isamu Asano)	30 each
The Joy of Vintage Tube Amps 1&2	(Tadaatsu Atarashi)	30 each NEW
Direct & Indirect Tube Amps	(Kiyokazu Matsunami)	40 NEW
SE Amps by Transmitting Tubes	(Kouichi Shishido)	50
The Remembrance of Sound Post	(Susumu Sakuma)	30
Classic Valve	(Hisashi Ohtsuka)	40
MJ Selected 300B Amps	(MJ)	30
Top-Sounding Vintage Power Tubes	(Stereo Sound)	30
Output Trans of The World	(Stereo Sound)	30
20TH CENTURY OF AUDIO	(Stereo Sound)	30
Vintage Speaker Units	(Stereo Sound)	30 NEW
Tube Amp Craft Guide	(MJ)	30

■MC STEP UP TRANS

Model	Specifications			Price (US$)	Postage**
	Pri.Imp(Ω)	Sec.Imp(kΩ)	Response		
Shelter Model 411	3~15	47	20Hz~50kHz	980	Area I $25
Jensen JE-34K-DX	3	47	20Hz~20kHz	550	Area II $30, Area III $40
Peerless 4722	38	50	20Hz~20kHz	300	Area IV $50

**Air Economy

■STAX

Model	Price(US $)
OMEGA II System(SR-007+SRM-007t)	Ask
SRS-5050 System W MK II	Ask
SRS-4040 Signature System II	Ask
SRS-3030 Classic System II	Ask
SRS-2020 Basic System II	Ask
SR-001 MK2(S-001 MK II +SRM-001)	Ask

■Speaker

Model	Specifications					Price* (US $)	Postage** (US $)			
	D (cm)	Ω	Response	db	w		I	II	III	IV
Fostex FE208 Σ	20	8	45Hz~20kHz	96.5	100	296	62	74	120	156
Fostex FE168 Σ	16	8	60Hz~20kHz	94	80	236	42	50	73	98

*Price is for a pair **Air Economy

■TANGO TRANS (ISO) 〔40models are available now〕

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US $)	Postage** I	II	III	IV
XE-20S (SE OPT)	20	2.5, 3.5, 5	20Hz~90kHz	300B,50,2A3	396	47	56	84	113
U-808 (SE OPT)	25	2, 2.5, 3.5, 5	20Hz~65kHz	6L6,50,2A3	242	42	50	73	98
XE-60-5 (PP OPT)	60	5	4Hz~80kHz	300B,KT-88,EL34	620	62	74	115	156
FX-40-5 (PP OPT)	40	5	4Hz~80kHz	2A3,EL34,6L6	320	47	56	84	113
FC-30-3.5S (SE OPT) 〔XE-60-3.5S〕	30	3.5	20Hz~100kHz	300B,50,PX-25	620	62	74	115	156
FC-30-10S (SE OPT) 〔XE-60-10SNF〕	30	10	30Hz~50kHz	211,845	620	62	74	115	156
X-10SF 〔X-10S〕	40	10W/SG Tap	20Hz~55kHz	211,845	1160	90	110	180	251
NC-14 (Interstage)	—	[1+1 : 1+1] 5	25Hz~40kHz	[30mA] 6V6 (T)	264	30	40	50	70
NC-16 (Interstage)	—	[1+1 : 2+2] 7	25Hz~20kHz	[15mA] 6SN7	264	30	40	50	70
NC-20F 〔NC-20〕 (Interstage)	—	[1 : 1] 5	18Hz~80kHz	[30mA] 6V6 (T)	640	42	50	73	98
NP-126 (Pre Out)	—	20,10	20Hz~30kHz	[10mA] 6SN7	264	30	40	50	70

Price is for a Pair

**Air Economy

■TAMURA TRANS (All models are available)

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US $)	I	II	III	IV
F-7002 (Permalloy)	10	3.5	15Hz~50kHz	300B,50	836	60	70	110	145
F-7003 (Permalloy)	10	5	15Hz~50kHz	300B,50	836	60	70	110	145
F-2013	40	10	20Hz~50kHz	211,242	786	70	84	133	181
F-5002 (Amorphous)	8	3	10Hz~100kHz	300B,2A3	1276	65	80	120	160

Price is for a Pair

**Air Economy

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Card Charge 6%

company that I have used frequently, with a long and respectable knowledge around the subject (Radio Duo, Mr. O. Taavela).

Commercially, you can use the Lundahl LL 1623 for 1.6kΩ, Audio Note TRANS-135, Tango U-808 (if still available), or Sowter SE12, to mention some from the best range. If you have a 16Ω winding in 3.5–5kΩ SE output iron (e.g., for 2A3, 300B, EL84, EL34), load it with 8Ω and there it is!

PRELIMINARY RESULTS

Calculating I could still use extra anode voltage loss, I decided to prototype with a simple CRC anode filter. Results showed less than 1.5mV hum at loudspeaker output with 150μF/330Ω/150μF pi-filter, and I accepted it and its 35V (for stereo) loss, leaving 260V for tube anode operation. With my 92dB/W speakers the hum can be just heard under 1m distance. I then had 100V at the end tube cathode and 160V available over the tube. By increasing the 330Ω filter resistor value, you can get possibly higher transformer voltage to fit; you can also decrease the value if you can tolerate more hum or increased capacitor values, respectively.

The first listening trials were shocking—the sound was superior to what I had been expecting, second to none I had previously constructed. I thought, however, that it could still be pushed to higher listening levels.

I decided to add a cathode follower to feed the E182CC because it is specified for positive grid current. One dual tube could serve both channels. Having a lot of the condemned ECC82s, I decided to give one 2 × 5mA a chance. No more is needed here because it is the tube that pulls up the grid.

That was the cure! Now I could obtain subjective levels comparable to my 5W 300B design. The sound was pleasing, the bass end was surprisingly tight, the imaging was excellent, and I only listened, smiled, and wondered several weeks around the rough prototype.

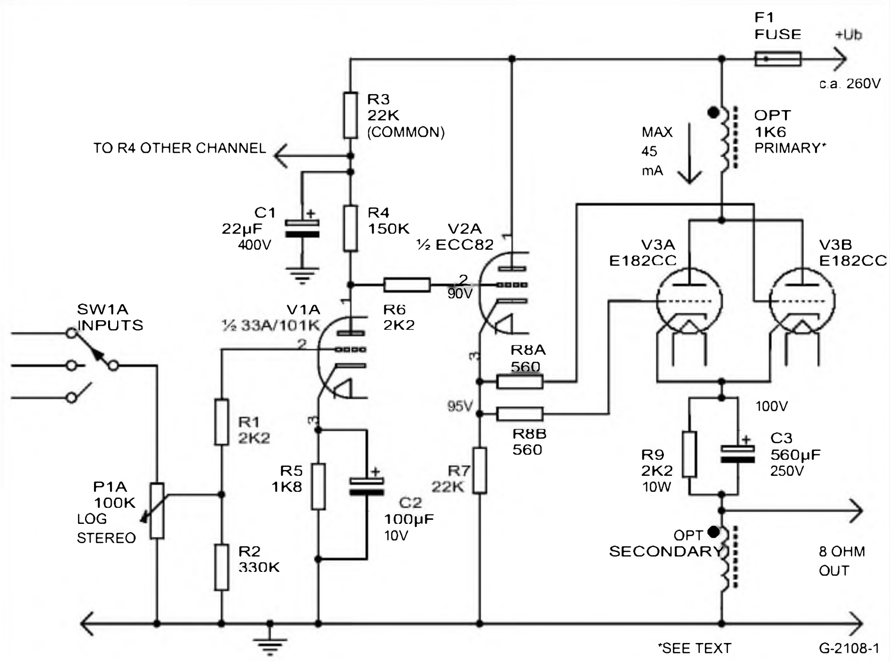
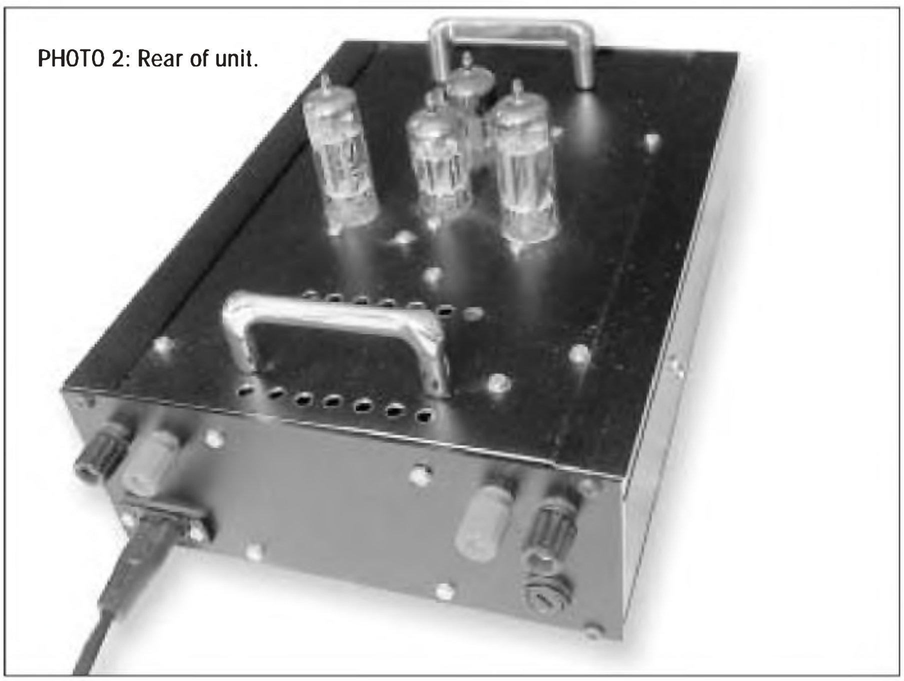


FIGURE 1: The main amplifier circuit. Bandwidth at 1W (–1dB): from 25Hz–90kHz. Power (at onset of visible clipping on oscilloscope): 2.1W. Hum at output: under 1.5mV. Sensitivity: 300mV.

PHOTO 2: Rear of unit.



FINAL CONSTRUCTION

Later I reconstructed my test prototype on a commercial aluminum chassis, hiding all the transformers under it (*Photo 1*). I also added a Bourns "blue" volume control, and two line inputs behind a small two-way toggle switch at the right front corner to complete the amp (*Fig. 1*). Here you can use your imagination quite freely to maintain the short signal path, possibly needing no screened wire at all. Grid-stopper resistors proved to be necessary here. They are always soldered directly near the corresponding input pin with about 3mm wire cut to length.

Overall, use as short wires and compact construction as possible. The wiring distance between tubes in the prototype is only around 3cm. The cathode power resistors lose about 5W each and the common power-supply resistor about 4W, so you need to handle their cooling! Use an IEC three-pole mains connector instead of the two-pole one I used, because I had it on hand (*Photo 2*). Remember to connect the other 6.3V heater end to ground; the tubes used can tolerate the existing cathode to heater DC+AC voltages. I used space wiring between tube socket pins, in addition to some tag strips and bus grounding (*Photo 3*).

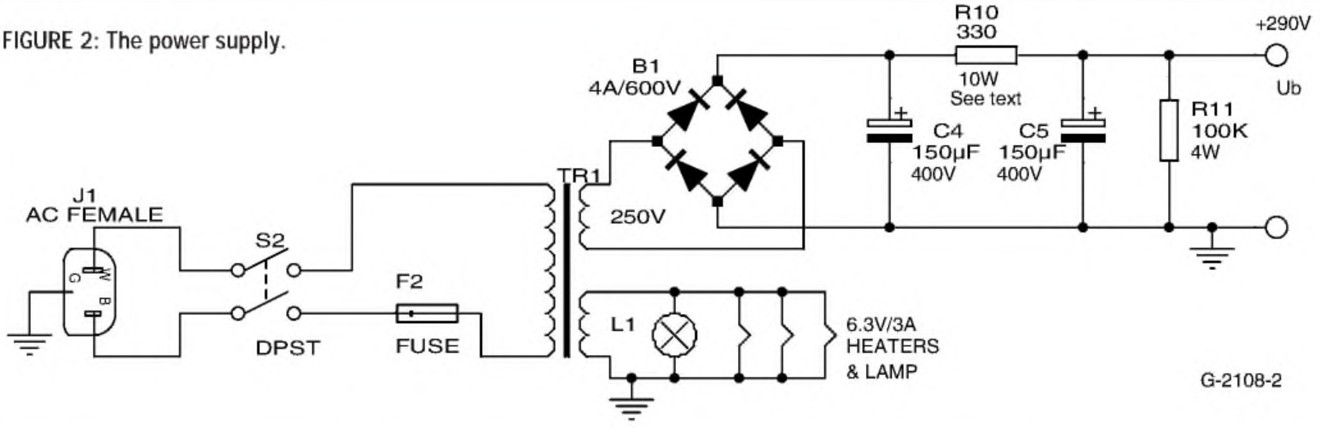
TESTING

My original still runs without anode fuses, but I recommend you use them, a slow 100mA per output tube (you may even come down to 80mA). You must take into account the effects of a direct-coupled design and consider a different test order than usual.

You first fit the first tube and may have about 90V at the anodes. You can adjust this by changing the cathode resistor R5 in a normal way, if needed. The voltage changes slightly downwards when you have the output tube current draw on, but around 90V is good. You notice that this stage works with quite a low current of under 1mA. This gave the best sonic performance with practically no next stage load. You need only 13V of peak anode amplitude for the end tube grids, giving a reason-

PHOTO 3: Underside of amp.

FIGURE 2: The power supply.



able overall input sensitivity of 300mV.

Then you add the cathode-follower tube and measure about 5V more DC at its cathodes than the previous result, 95V or so with the ECC82. 12BH7 also fits directly for the follower stage. I found no remarkable sonic differences, so one Siemens E82CC with its poor hi-fi reputation earned a good home here.

You now have an excellent preamplifier, too; you could even consider taking cathode outputs for this purpose via capacitors to extra RCA connectors and use the front end as a preamp, just taking the output tubes away. For this purpose I would replace the CRC power supply with better filtering; e.g., with a 10H choke or a FET source follower stabilizer (e.g., IRF840) for no discernible hum. 260–270V is a good value, then, to feed all the stages. Also, if you are driving horns around 100dB/W sensitivity, this better filtering is recommended.

I have used this old proven pre-stage topology in my Revenge KR842VHD amp[1]. I was also glad to see it used in the portable mixer construction by Paul J. Stamler, presented some time ago in this magazine[2].

At last, after switch-off, you can add the output tubes, one by one. After warming there will be about 100V at their cathodes, meaning 40–45mA goes through; 90–100V is good. It is good practice to have resistive loads at the transformer outputs when testing or measuring.

There is the possibility that you don't know the phasing of the output secondary. In this case, connect the cathode resistor/capacitor common earth end first directly to ground as normally, leaving the output winding hot end floating with an 8Ω load resistor connected and measure the gain. If you are a feedback hater, you can even leave the amp like this (here you then have the possibility for bridged mono usage with floating, series-connected OPT secondaries). When you now connect the output winding in series with the cathode circuit

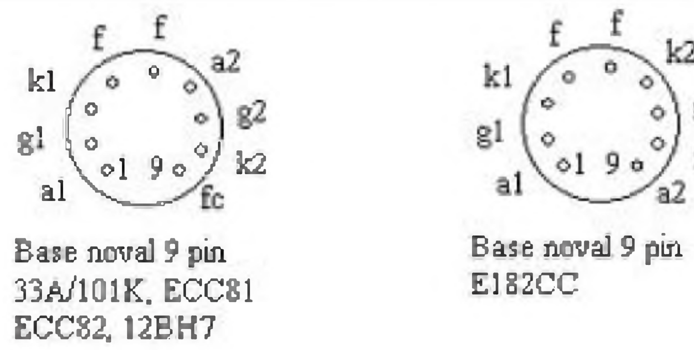


FIGURE 3: Tube sockets.

G-2108-3

OPTIONAL CURRENT GENERATORS FOR THE CATHODE FOLLOWERS

Inspired by the previously mentioned series of articles by Paul J. Stamler in *audioXpress*[2], and out of curiosity, I showed that you can improve the sound by replacing the cathode follower resistors (R7, 22kΩ) with semiconductor current generators. A simple solution I tried is presented in *Fig. A*, both transistors using a common low-current yellow-led reference (1.7V). Instead, you can use two 1N4004 diodes in series. The current is ca. 1.1V/the emitter resistor (here 330Ω, or 0.6V/the resistor, in the case of two usual diodes). The transistors may need a small cooling clip or be insulated/bolted to the chassis, depending on the current used. Note: the LED may not light if the cathode follower tube is off of base.

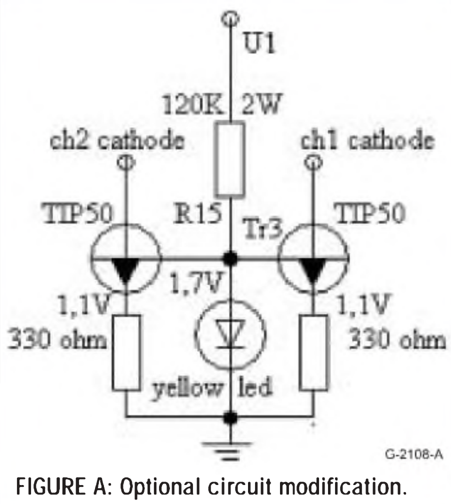


FIGURE A: Optional circuit modification.

phased as shown, the gain should be about 6–7dB less, slightly depending on the turns ratio of the OPT. If you have no opportunity to measure, this change is also clearly audible with music when you don't touch the gain chain.

If the winding is at wrong (positive fb) polarity, the amp very probably oscillates (mine did). Alternatively, it might show 6dB more (audible) gain. If phasing with listening, don't destroy your speakers; use a cheap one here for testing. In commercial transformers the hot ends are usually marked and there is no connection problem. Now, if these tests are OK, play quality music and play it loud! I continue doing so with my 92dB/W Fostex 103S 4″ speaker wonders.

TUBE SELECTION

My output tubes bear a Philips label. The first tube is, as mentioned, STC/Brimar 33A/101K, which is the professional 10,000-hour ECC81 widely used in British telephony carrier wave equipment as a telephone speech channel amplifier active element before the intrusion of transistors. My 33A/101K

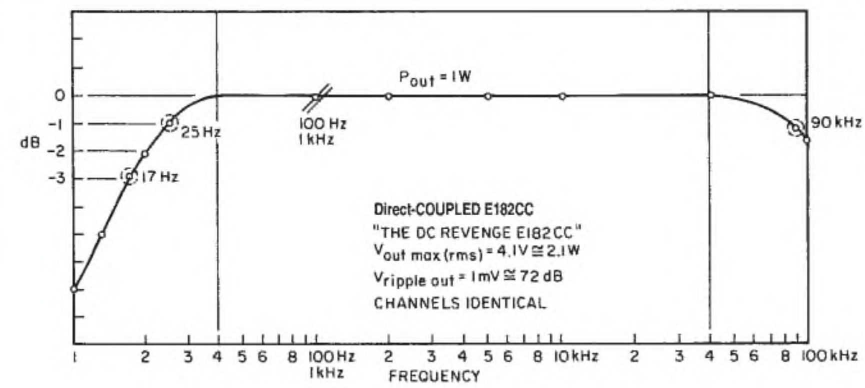


FIGURE 4: System response. G-2108-4

heaters have drawn 150/300mA, not 600 or 800mA as some claims have stated.

I have personally supervised repairs of this kind of STC tube telephony equipment in a large volume at the start of my career in the beginning of the '70s at Telecom Finland. The normal ECC81/12AT7 is very good and cheap here (see photos), and you can get good sensitivity with it. The Mullard vintage M 8162 is excellent.

You can also try your favorite signal tubes and choose your preference. I don't guarantee the amp won't oscillate then! Try, for example, 2C51/6N3P, 6DJ8, 6N1P, ECC83, ECC85, ECC88, or 12AV7 (different pinouts and cathode resistors, trim 90V at the anode). Also comparing manufacturers ("tube rolling") around the same tube type is fun, but always check about 90V at the anodes first without the output tubes fixed.

My new favorite is Philips Miniwatt E80CC Special Quality audio triode, which sonically beats (to my ears) all the aforementioned in a normal low-current common cathode gain stage. E80CC fits here pin to pin in place of ECC81 with no component changes, but has only about half the gain, although it is still sufficient for a CD player.

A general note: Some small-signal tubes give a white initial flame when powered up. Don't be afraid of this. I have tubes in use doing it at every start and they have lasted for tens of years. Experimenters can further try to remove the first stage cathode bypass or put it behind a toggle switch permanently biased with a 2MΩ resistor from the cathode (switching transient elimination), thus having two sonic nuances with different sensitivity for the amp. I have not verified this in practice with this amplifier. Oscillation or more hum is then possible in this high BW, high gain tube block!

LISTENING CRITIQUE

The sound is sweet but detailed, and the bass end is unbelievably tight. The local feedback over the output tube works well and also lowers the distortion by half (I also use this trick with my 300Bs, but with their voltage gain of four you can get only a couple of dBs). The –1dB bandwidth is 25Hz–90kHz with my self-designed transformers! The 20VA stock shows weak signs of saturation on the oscilloscope at full power under 30Hz, so I recommend bigger, better irons (the concept is worth it).

Power is over 2W per channel, which is excellent with about 7W anode loss. The small positive end grid drive is a bonus. This design beats all my tried

PARTS LIST
THE DC REVENGE E182CC AMPLIFIER

PER CHANNEL, RESISTORS ½W UNLESS OTHERWISE NOTED	
R1, R6	2.2kΩ
R2	330kΩ
R3	22kΩ (common for both channels)
R4	150kΩ
R5	1.8kΩ
R7	22kΩ/2W
R8	560Ω, 2 per channel, one for each grid
R9	2.2kΩ/10W, e.g., Dale with cooling/fixing block
CAPACITORS	
C1	22µF/400V, common for both channels
C2	100µF/6V or 10V, tantalum
C3	560µF/250V, high quality
OTHER	
S1	Double throw two-pole miniature toggle switch (or a rotary SW)
P1	2 × 100kΩ log stereo potentiometer, e.g., Bourns "blue"
F1	Fuse 100mA slow, see text
OPT	SE output transformer, see text
THE DC REVENGE E182CC POWER SUPPLY	
S2	Mains switch, two pole
L1	Pilot lamp 6V/50mA
F2	Mains fuse 0.8A slow (T)
MT	Mains transformer, 6.3V/3A and 250V/150mA
DB	Diode bridge 4A/600V (see text)
C4, C5	150µF/400V, high quality
R10	330Ω/10W, e.g., Dale with cooling/fixing block (see text)
R11	100kΩ/4W or 2 × 47kΩ/2W in series
Other	IEC female chassis mains connector (3-pole), fuse holder, knob

SEs considering bass end fitness and is one of my favorites at the moment for listening, as well as in overall compactness and portability. I next intend to try two tubes in parallel to 1.25kΩ biggish Audio Note TRANS–115 irons. I ordered these some time ago from Angela Instruments and they are excellent-sounding with a sturdier tube (6AS7) in my initial tests.

I and my friends have used the unit in daily listening for over three years, and this gives an indication that the DC coupling with its unspecified heating order is able to survive, which was a non-verified question for me. Anyway, I recommend that you use those anode fuses; otherwise, in the case of a failure some smoke may appear. You should also consider an output tube anode voltage switching delay. My anode supply is a normal 600V/4A straight diode bridge. I have yet to test hexfreds—e.g., IR HFA04TB60—or ultrafast diodes—e.g., UF 4007—but they are on the list; interested readers can try them.

CONCLUSIONS

Of course, purists can and will use tube rectification, which also provides the anode delay as a bonus. Note that you must then use a lower value smoothing capacitor (order of 50μF) after it (unless you use a choke input with a proper choke; an ordinary one tends to keep mechanical noise). I tried a tube in my prototype (EZ 80) but changed to the sonically (to me) sturdier semiconductor version.

I recently came across a trick to use two diodes (e.g., 1N4007, negative leg) and a double rectifier tube (positive leg) in a Graetz bridge arrangement with one anode winding (no center tap) only. You can use that here. In addition, this gives you an easy way to compare an all-semiconductor solution: make a plug of two diodes to fit directly to the tube base (note the orientation of diodes in the Graetz bridge) to have a full semiconductor bridge. Verify the stage anode voltage to remain around 90V.

I consider it important to repeat the familiar tube gear safety precautions here. Keep one hand in your pocket when measuring and/or wear thin cotton gloves. Make certain your anode voltage is under about 30V before doing any modifications (this is why the 100kΩ bleeder is at the supply). Always disconnect the mains cable when working on the unit.

As I've said, this is my favorite amplifier at the moment. I can report on even better overall quality with more power from my new Revenge KR842VHD SE design with C3m triode mode driver, E80CC pre and Lundahl iron set, which is ready and playing[1] (this new tube from KR Audio Electronics in old Tesla tube lab premises in the Czech Republic is an enemy of most 300Bs). However, you can easily sink about $1,500 in good components there, compared to $150 spent on mostly surplus items with this design; the price mainly depending on the output transformers. The tubes and the output transformers are the basis for the sound quality. Enjoy the indirectly heated music; this little piece of SET equipment makes it cheaply and pleasantly available. ❖

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Acoustical Shrinkage of Large into Small Rooms

Honey, I shrunk the room! Key parameters you need to consider for improved room acoustics. **By Stephen Harner**

The physics of sound is well understood and the equations that describe the propagation of sound can be derived from the fundamental nature of the molecules of air. You can then use the equations of sound to calculate various parameters for a given set of boundary conditions—i.e., volume, surface area, and so forth—of a room.

Unfortunately, there aren't many widely accepted parameters that acousticians agree will consistently improve the acoustics of small rooms. However, there now exist several widely accepted parameters that you can adjust to improve concert-hall acoustics. In the following sections, I discuss the six most widely accepted parameters for concert halls (large rooms), and then apply them to small rooms by extrapolation, or so-called acoustical shrinkage.

LOUDNESS

Loudness, as applied to concert halls, really means the relative perceived increase in loudness due to the reflected sound field over the loudness that naturally comes directly from the source. In a large concert hall, not being able to hear the quiet passages (i.e., soft violin notes) would be bad for the overall sound quality. Because small rooms require listening in close proximity to the source, loudness is not really a factor.

INITIAL TIME DELAY GAP—ITDIG

The initial time delay gap, which I call ITDiG (it dig, you dig?), is the time between the arrival of the direct sound and the arrival of the first reflected sound. Beranek[1] says that the ITDiG should be less than 20ms for optimum quality of concert-hall sound, while longer values have a negative effect. Small rooms will generally always have an ITDiG less than 20ms, and typically will be around 2ms, unless the sound source is placed within 1′ of the wall (ignoring the low-level edge-diffraction effects).

Advocates of live-end/dead-end theory say that early reflections "smear" or "blur" the direct sound, and that eliminating the early reflections improves the sound. This involves treating the front half of the room with acoustic absorbers. Others suggest using a mirror moved around on the wall to locate the speaker reflections as viewed from the listening position, then treating these spots with acoustic absorbers. Both of these methods are akin to lengthening the ITDiG.

Does too short an ITDiG make for bad sound, or does it improve the sound by adding width and fullness (like reverberation)? Very short ITDiGs—less than 1ms—do affect the way we localize sound sources. Hafter's Auditory Perception Lab[2] has shown that when the left ear is fed an impulse incrementally increasing from 0.1 to 1.0ms before or after the right ear, the brain hears the sound as coming from further and further left or right, depending on which ear is fed the first impulse. Certainly, if the two impulses smeared each other, then the brain would not be able to discern them. The fact that the ear/brain auditory system can discern impulse signals spaced apart by only 0.1ms is almost unbelievable!

If the second sound came from the same direction as the direct sound, and at the same time, then some phase cancellation (generally referred to as comb filtering) would occur, causing peaks and valleys in the frequency response. In fact, you can hear this when you place two speakers in close proximity and feed them the same signal. But the direction and timing of reflected sounds are typically different from the direct sound, and there is no conclusive evidence that you actually hear comb filtering with speakers placed close to walls. It is widely accepted that the ear/brain auditory system can key in on the direct sound, or first sound, then fuse or compensate for any subsequent similar sound without detrimental effect to the overall sound.

For small rooms, in light of the localization effects, I agree with Ludwig[3] that it would seem prudent to ensure at least a 1ms ITDiG by placing the sound source at least 2′ away from the walls. However, placing an acoustic absorber on the wall to absorb the first reflection may actually hurt more than it helps, because most absorbers commonly used are not wideband, but rather have very little absorption in the low frequencies, thereby producing a very unbalanced first reflection spectrum.

BASS RATIO—BRat

Bass ratio—what I call BRat—is a measure of the level of lower-frequency re-

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verberation versus the level of higher-frequency reverberation. Beranek defines it as the average reverb time at 125Hz and 250Hz, divided by the average reverb time at 500Hz and 1000Hz (see subsequent definition of reverb time). Although it is defined as a time measurement, it is perceived as a level (or gain versus loss).

For example, a room with a high BRat sounds warm or bassy, while a low BRat sounds harsh or bright. The main factor affecting BRat is absorption. Thin paneling and drywall absorb bass by flexing and vibrating. Most acoustical absorbers absorb treble and midrange, but little bass (just like carpet).

Beranek's recommended range is from 1.1 to 1.45, for concert halls with lower reverberation times. Note that most studies of concert-hall acoustics use symphonic music as the main subject, which in general has much lower levels of low-frequency sounds than does modern music (i.e., rock and roll). This would require a higher BRat for good-sounding symphonic music and a lower BRat for good-sounding modern music.

Small rooms should sound natural and balanced, and since most rooms inherently have more high-frequency absorption than low, a slight emphasis in the bass is desirable, which agrees with the range of 1.1 to 1.45. However, overuse of foam-type absorbers is all too common, resulting in too high BRat with too much bass. Since most live-end/dead-end rooms use foam on the dead end, they sound unnatural—half live, half dead (frequency-wise).

INTERAURAL CROSS-CORRELATION (SUBTRACTED FROM ONE)—I-ACk

Interaural cross-correlation subtracted from one, what I call I-ACk, is a measure of apparent source width, or more specifically, I-ACk=1-IACC (see Ando[4] for a technical discussion of IACC). A sound source directly in front of you without any reflections will have zero I-ACk. A sound source fed to your left ear, while blocking your right ear, will have I-ACk equal to one.

Lateral reflections from the side walls increase I-ACk, because sounds coming from left or right will arrive at the ears differently, except if the source is dead center in front of you, and you are positioned exactly half way between two identical walls, but even then tiny movements of your head will increase I-ACk.

Since live-end/dead-end theory reduces early lateral reflections, favoring later reflections from behind, it also reduces I-ACk. This negatively affects acoustical quality. For small rooms you should make the walls as live as possible, with any acoustic absorption used only on the floor or ceiling.

REVERBERATION—REVERB

Everybody loves a little ambience, and some love it a lot. Reverberation time, or simply reverb, is the time it takes a sound to drop in level by 60dB. It is the most obvious difference between indoor and outdoor sound.

Talking in a dead room or an anechoic chamber is immediately disconcerting. Music played in this way is universally considered abnormal, and some may say it is actually unpleasant. So there is no question that rooms should have at least some reverb, but how much is best? Beranek and many other acousticians agree that about two seconds is best for most symphonic music in concert halls. They also agree that different types of music require different reverb times.

Beranek, among others, goes on to say that the early decay time (EDT) is more important than the later decay time, and defines EDT as the time it takes a sound to drop in level by 10dB then multiplied by six to normalize it. Some concert halls, such as the Meyerson/McDermott in Dallas, have very different early and late decay times. The three most widely praised concert halls in the world—Symphony Hall in Boston, Concertgebouw in Amsterdam, and Musikvereinssaal in Vienna—have reverb times close to 2s, with some, but not much, variation in their early versus late decay times.

The consensus is that smaller rooms should have less reverberation than larger rooms. They typically do anyway, because reflections are attenuated each time they bounce off a surface, and sound will have more bounces per second in smaller rooms. Thus the density of reflections will be higher in a small room, but sound will decay more rapidly.

The most important factor in determining the optimum reverb time is the type of music. Modern music is faster and more impulsive than classical music. This requires less reverb time.

You could use acoustical absorbers to reduce the reverb time, but most absorbers work on higher frequencies with very little effect on lower frequencies. Of course, specialized bass traps absorb narrow bands of lower frequencies, but their absorption is proportional to their area, so covering a large area (and a wide band of frequencies) will require many of them. Contrary to popular opinion, putting one or two bass traps in the corner won't do much.

The right amount of reverberation in a small room is whatever sounds good to the listener, which may vary from one person to the next (along with the type of music). Beware of excessively dead rooms with reverb times much less than 0.5s, which will have a negative effect on sound quality. A possible exception would be a control room for monitoring a recording, where you might choose to hear exactly what is being recorded without any pleasant reverberation added. Although, in that case, headphones would be a better solution.

SURFACE DIFFUSIVITY—SurDif

Surface diffusivity, which I call SurDif, is a property of highly irregular walls. Flat walls have no SurDif. Diffusion is not directly related to absorption, but absorption tends to limit the effects of diffusion. That is, dead surfaces cannot have good diffusion because they absorb most of the sound. The degree of diffusivity is proportional to the amount of surface irregularities, but is inversely proportional to the absorption. Unfortunately, there is no objective way to measure it.

Beranek says that diffusivity is an architectural feature that must not be underestimated. The three most widely praised concert halls in the world have excellent surface diffusivity.

Diffusion is always cited as very important to good sound. Haan and Fricke[5] say that it is the definitive quantity for concert-hall sound, and the following is how they define the surface diffusivity index:

High Diffusivity (1.0)—coffered or checker-designed with deep recesses or deep beams (greater than 4″), or random diffusing elements over the full area (greater than 2″ in depth), and all of the area must not embody any sound-absorbing materials.

Medium Diffusivity (0.5)—angled array of broken surfaces, or ornamentally decorative treatment applied with shallow recesses (less than 2″ in depth), or flat concrete surfaces behind a semi-acoustically transparent screen with mostly reflective materials.

Low Diffusivity (0.0)—large separate paneling, or smoothly curved surface, or large flat and smooth surface, or semi-acoustically transparent mesh screen, or heavy absorptive treatment applied.

SurDif is the most important factor in small room acoustics, so the higher the diffusivity, the better. Several diffusors are available, but they tend to be quite expensive. Some of them use mathematical formulas to help ensure random frequency reflection distribution. Fortunately, there are some household objects that make great diffusors, like bookshelves!

SUMMARY

Irregular surfaces, such as bookshelves, brick fireplaces, cabinets, or manufactured diffusors are the panacea for small room acoustics. They will help your room sound as marvelous as it can. Moderate use of acoustical absorbers may help in some rooms, but as far as placing the absorbers following live-end/dead-end theory, forget about it!

TABLE 1
HARNER'S SUBJECTIVE PREFERENCE RANKINGS FOR SMALL ROOMS

1. SurDif—40%
2. Reverb—25%
3. I-ACk—20%
4. BRat—10%
5. ITDiG—5%
6. Loudness—0%

Table 1 ranks the parameters just discussed according to relative importance for room acoustics. However, equally important as perhaps all of these parameters combined, are standing waves, which many other authors have covered extensively. I do have a somewhat unique solution for the problems caused by standing waves that I may present in the future. ❖

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Line-Transformer-Driven SV300B SE Amplifier

This article on the latest from a noted Japanese designer originally appeared in Japan's premier high-end tube magazine, *MJ Audio Technology*, August and Sept., 2002. **By Satoru Kobayashi**

PHOTO 1: Completed SV300B-based amp.

As an advocate of Plitron toroidal transformers, I have built several amplifiers using their products, taking advantage of their ultra-wide frequency response and sound clarity. I present here a toroidal line-transformer-driven SV300B single-ended amplifier, featuring a wide power bandwidth at 1W and even 10W output (*Photo 1*).

Even though Plitron's line transformer (PAT-4126-02) was originally designed for a line-stage amplifier because of its winding structure with 10kΩ (primary) and 600Ω (secondary) taps, I was able to find a new way to use this as an interstage transformer to produce over 200V pp driving the signal directly into a SV300B final tube, by driving it in reverse.

As I expected, the amplifier produced a clear and strong sound, very similar to a SV300B push pull-amplifier[1], with a wide and deep presence of stereo sound as compared to CR-connected single-ended amplifier[2].

CIRCUIT

The keys to the success of this amplifier are: 1) to define a correct circuit to drive the low-impedance tap of 600Ω with a valve, and 2) to get a proper driving signal over 200V pp at a 10kΩ tap to drive a SV300B.

Line Transformer

Menno van der Veen designed this transformer (PAT-4126-02) as a line transformer for a preamplifier with an ultra-wide frequency over 600kHz, utilizing a toroidal-wound architecture, providing 10kΩ and 600Ω impedance taps, respectively (*Figs. 1* and *2*). I have already confirmed that this line transformer has a 300kHz bandwidth at 20V pp with my line amplifier design driven by a 6N1P

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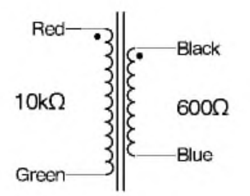

Primary Impedance :	10.02kΩ
Secondary Impedance :	600Ω
Winding ratio :	4.087
Frequency Response :	
0dB ;	1.854Hz~151.411kHz
−1dB ;	0.791~340.024kHz
−3dB ;	0.402~641.933kHz
Maximum Power :	40mW
−3dB Power Bandwidth (Low end frequency) :	10Hz
Primary Inductance :	82.7H
Primary Leakage Inductance :	0.935mH
Primary Leakage Capacitance :	1.23nF
Primary Internal Resistance :	33.5Ω
Seconday Internal Resistance :	3.01Ω
Recommended Driver Internal Resistance:	90Ω

Size : 96.9±0.3 (Diameter) × 44±0.2(Height) mm
Weight : 1.0kg

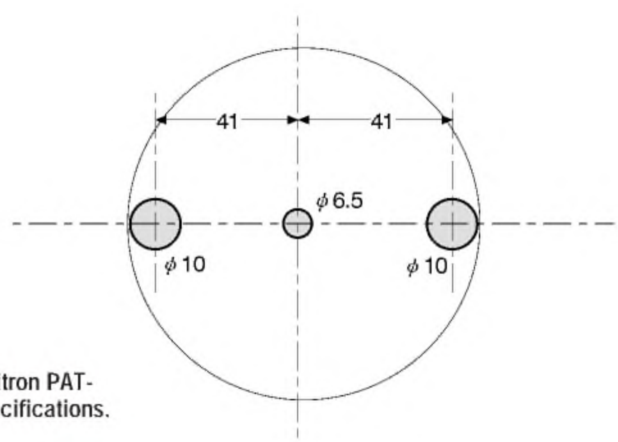

FIGURE 1: Plitron PAT-4126-02 specifications.

A-2174-1

(from page 32)

cathode follower, which was introduced in the March 1999 issue[3] of *MJ Audio Technology* magazine in Japan. So I believe that this might also demonstrate a wide bandwidth as well—even if I could drive a 600Ω secondary tap at 50V pp, expecting to get 200V pp at a 10kΩ primary tap.

Gain Distortion

A 200V pp is the minimum requirement to drive SV300B efficiently under the operating condition of B+ 450V, –95V grid bias and 60mA of idling current[2] (*Fig. 3*). Assuming the input sensitivity of this amplifier is 1V pp at the input terminal, then the entire gain of voltage amplifier stage should be at least 200×. First of all, you should be aware that a 1:4 winding ratio of such a transformer provides 4× of gain, which works as an amplifier in this application. So the voltage driver circuit including a cathode follower needs at least 50× of gain anyway.

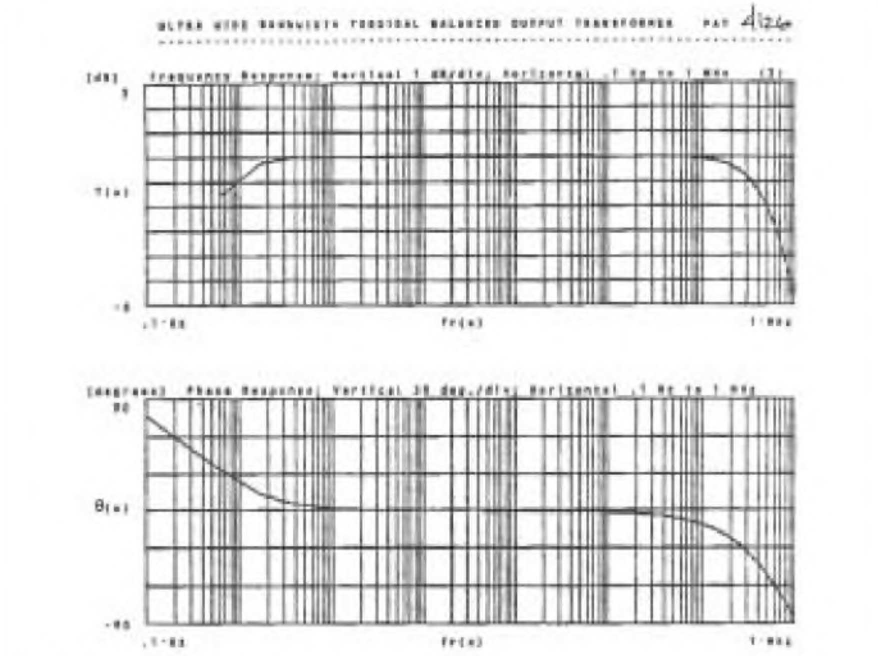

FIGURE 2: Frequency response. A-2174-2

Voltage Amplifier—First Stage

I chose Svetlana 6N1P because of its inexpensive price in Japan, and because I had several extras in my parts box. The 6N1P also shows a good linearity when using this for a voltage amplifier. The circuit I used is a basic resistor loading voltage amplifier, copied from the Svetlana technical bulletin no. 43[4], which allowed me to save time in setting parameters.

The alternative tube is not available, even though 6DJ8 might be a substitute for 6N1P, since the plate to cathode voltage of 6N1P is over 200V, which exceeds the maximum plate voltage value of the original 6DJ8 in this particular application. The 6N1P voltage amplifier provides approximately 28× of gain.

Cathode Follower

The cathode follower (*Figs. 4* and *5*) might be the most suitable circuit to drive such a low-impedance load of 600Ω. I chose the Svetlana SV83 (6BQ5/EL84 compatible), mainly because it showed a better performance when compared to popular voltage driver tubes such as the 12AU7, 12AT7, 12BH7A, 5687, and 6N1P, which I had in my parts box.

Prior to selecting the right valve, I ran a TubeCAD simulation that showed SV83 can generate the maximum voltage in the range of 30 to 40V pp under a 600Ω load with 40 to 70mA cathode current, while the other valves generate less than 20V pp at a 600Ω load with 15–20mA cathode current. SV83 generates approximately 30V pp with –40mA of idling current. However, the value obtained by simulation is not enough to drive this transformer. Thus, I chose SV83, even though I needed to experiment to see how the SV83 and cathode-follower (CF) circuit generate up to a 50V pp driving signal with this transformer.

Optimizing B+ Supply Voltage

I ran the simulation with 250V plate voltage. Also, I checked the CF circuit linearity by increasing the plate voltage from 250 to 350 to 450V. The results (*Fig. 6*) showed that the maximum output voltage stays the same as the plate voltage varies. I chose 450V to save on parts and keep the linearity as high as possible.

However, the plate idling power consumption must not exceed the maximum limit of 12W, so the plate current of SV83 must not be more than 40mA (=12W/300V). Thus, the overall parameter setting on SV83 would be: SV83 plate voltage (450V) = SV83 plate to cathode voltage (300V) + cathode voltage (150V).

Biasing SV83

I chose a self-bias circuit for SV83 to link with the previous stage without any restriction. Somehow, I prefer a direct connection between the first stage and this CF stage, though the direct connection narrows a flexibility of parameter set-

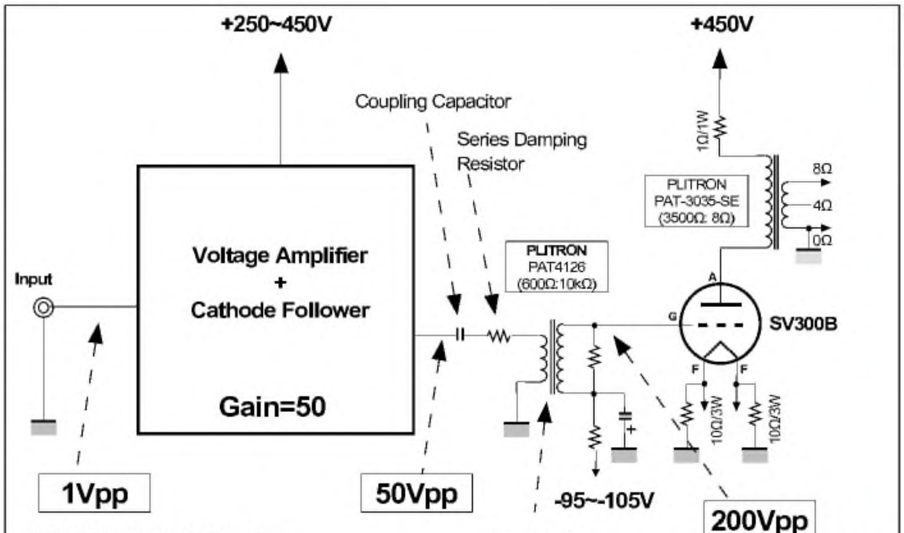

FIGURE 3: Gain distribution.

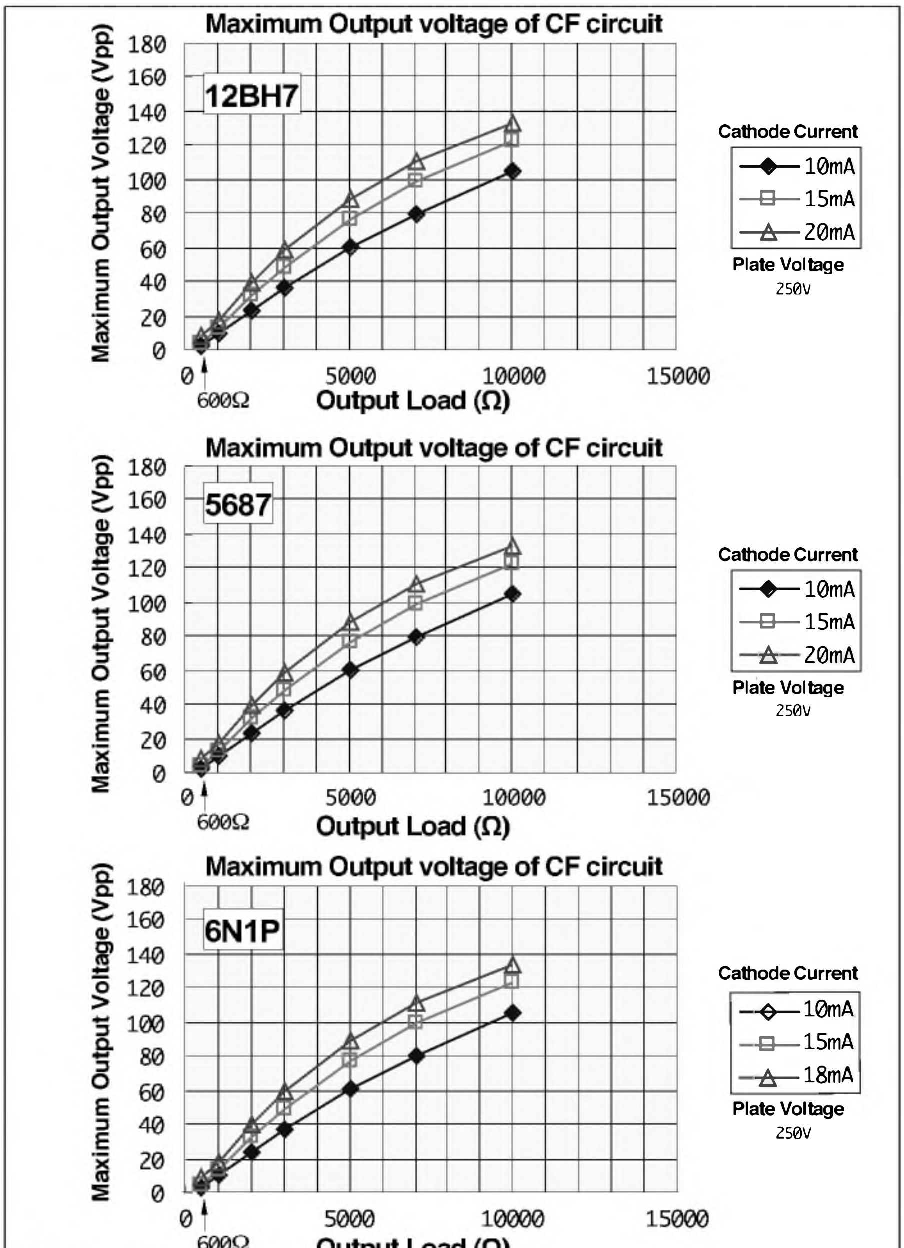

FIGURE 4: Cathode follower.

ting. So the AC connection (e.g., CR connected circuit) eliminates such a restriction and forms a simple circuit.

The TubeCAD simulator indicated that the –10 to –12V grid bias would be adequate for the SV83 self-bias circuit. Tentatively, I used a 10V zener diode from my parts box in the above-mentioned experiment.

Defining Cathode Resistor Value

Given the plate voltage and plate current, you can calculate the SV83 cathode resistor value by the following formula: cathode resistor value (3_4kΩ) = (450V – 300V – 10V) ÷ 40mA.

To determine the optimized value, I built an experimental driver circuit to view the falling edge of the 50kHz square wave while varying the resistor value from 2.5 to 3.5kΩ. The final decision was made by the rise time of the 50kHz square wave, which would be 2.5 to 3μs at 200V pp output level.

The experimental result defined the value of 3.3kΩ. At the same time, I confirmed the circuit produced over 200V pp, up to 256V pp, implying that at least 64V pp was applied to the 600Ω tap of this transformer (*Fig. 7*).

The following shows the tentative parameter settings of the driver circuit:

Plate voltage: 450V
SV83 plate-cathode voltage: ≈300V
SV83 idle current: ≈42mA
Plate power consumption: 12.6W
Grid-bias voltage: –10V
Cathode resistor value: 3.3kΩ

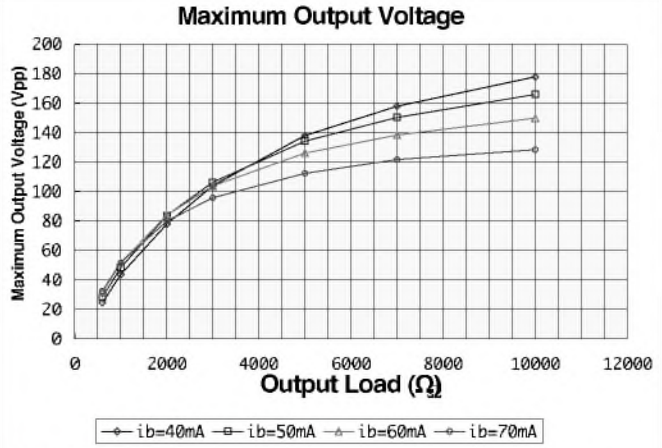

FIGURE 5: SV83 cathode follower.

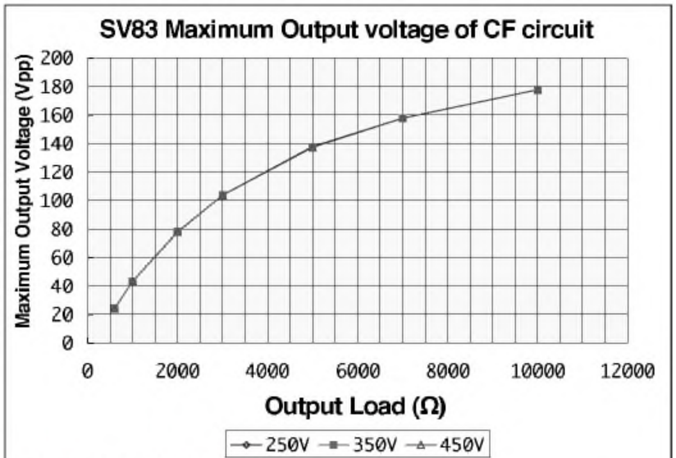

FIGURE 6: SV83 optimized power supply.

Coupling Capacitor

The key to handling this particular toroidal transformer is not to apply any DC current into it when driving. You must insert a coupling capacitor between the cathode pin of CF valve and the 600Ω tap of the transformer to block the DC current path. The coupling capacitor value determines the low cut-off frequency, as well. The larger the capacitance, the lower the cut-off frequency. I chose a 10µF, which produces approximately 23Hz, according to the simulation (*Fig. 8*).

Series Damping Resistor

When a square wave drives such an inductive transformer, the transformer reacts with its inductance and generates an overshoot at the rising edge of square wave, influencing its frequency-response curve. It is commonly said that a series resistor between a capacitor and a transformer works efficiently to suppress its overshoot. I viewed a waveform at 10kΩ tap with 47kΩ load using a digital oscilloscope and took a multiple-exposure photo of several waveforms with various resistors—82, 120, and 150Ω—from my parts box. *Figure 9* shows a big overshoot at a rising edge without any damping resistor. The 120Ω resistor damped efficiently with negligible overshoot.

Voltage Drive Characteristics

Overall, the experimental circuit brought the maximum output level of 300V pp with 3V pp input. The following shows its major characteristics.

Maximum output voltage: 300V pp
Input voltage: ≈3V pp
Gain: ≈100
Frequency response: 30_200kHz @ 200V pp
Distortion: <1%

Figures 10 and *11* show the overall characteristics and waveform.

The overall gain was approximately 100×, which did not reach the initial goal of 200× gain that I expected. The input sensitivity would be 2V pp, corresponding to a 6dB gain loss. It needs only a two-click adjustment to keep the same sound level when listening to music, if you are using a click-stop type volume control such as DACT CT-1 or equivalent. So there might be a slight difference in results. If you wish to get more gain than this, replace 6N1P with other valves such as 12AT7, 6189, and so on, but this change will require fine-tuning to maximize the gain driving SV83.

Total Current Consumption

The following summarizes the circuit current consumption of each circuit block.

First stage-6N1P ≈1mA/ch

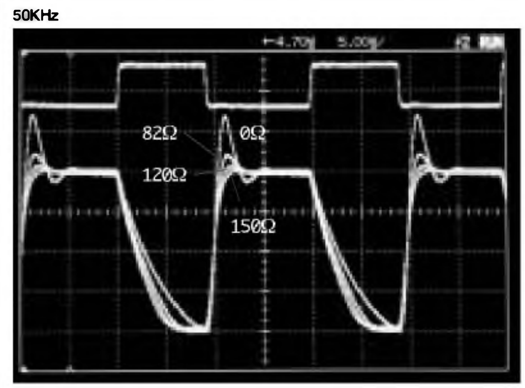

FIGURE 9: Damping resistor.

A-2174-9

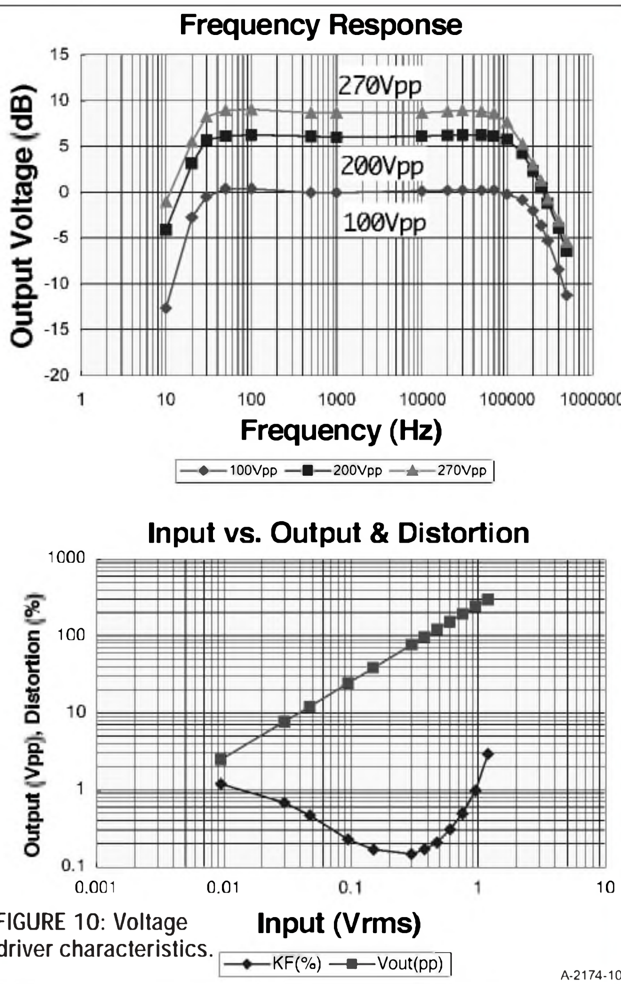

FIGURE 10: Voltage driver characteristics.

A-2174-10

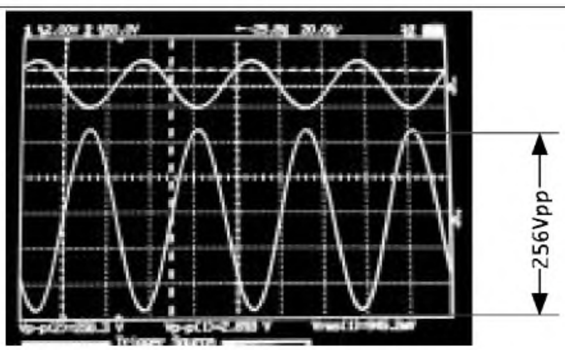

FIGURE 7: 256V pp output.

A-2174-7

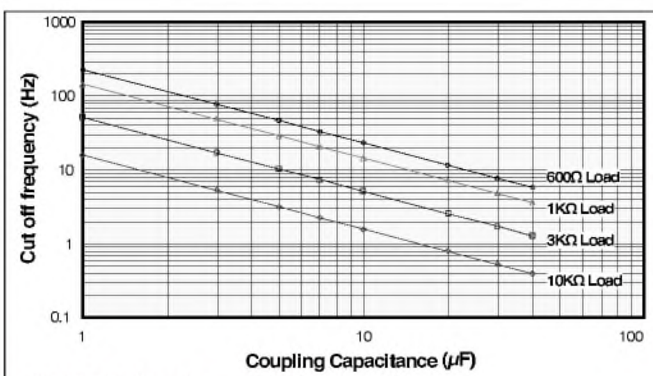

FIGURE 8: Coupling capacitance.

A-2174-8

Cathode follower-SV83 ≈40mA/ch
Final stage-SV300B ≈110mA/ch
Total: ≈151mA/ch

The overall circuit consumes approximately 300mA.

Filament Current Consumption

The filament current of all valves is as follows:

6N1P 0.6A
SV83-2 units 1.52A (=0.76 × 2)

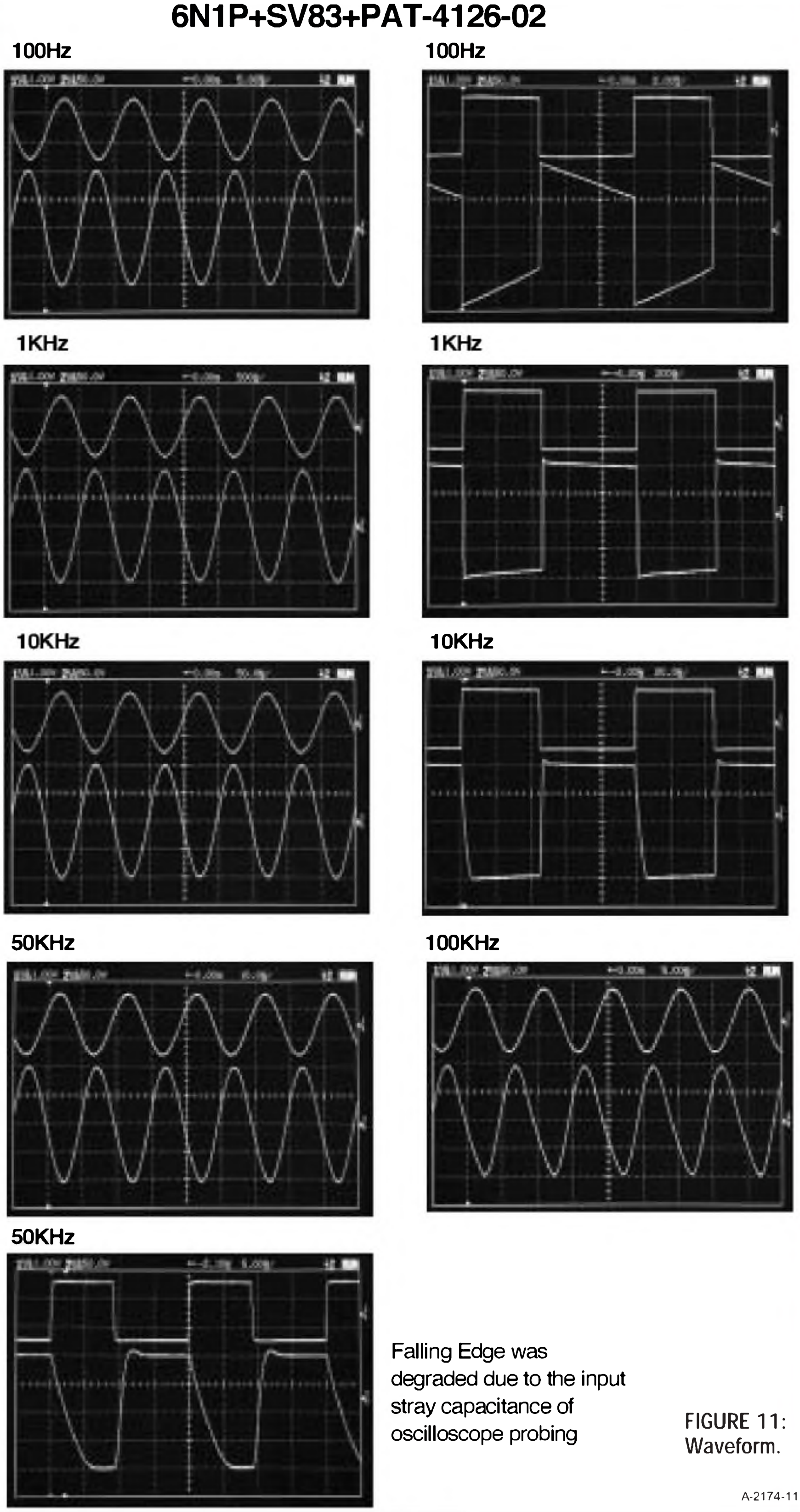

FIGURE 11: Waveform.

6D22S 1.5A
Total: ≈3.6A

The two units of SV300B filament (5V, 1.5A) are powered by a couple of 5V–2A DC switching power modules.

Power Supply, MOSFET Ripple Filter

This section gives an overview of the power supply, which produces enough power to drive SV300Bs and other valves. Plitron power transformer #754709 provides 340V–0.7A AC taps, so it produces approximately 480V DC unregulated and 390mA DC (=0.7A ÷ 1.8) with a bridged diode rectifying. After rectifying, unregulated 480V DC goes to a MOSFET ripple filter, providing a very simple stabilized circuit with a few zener diodes (450V = 150V × 3) that converts unregulated to the 450V DC regulated, with a good ripple rejection. This ripple filter eliminates a choke coil, reducing amplifier weight and allowing space for other major components.

The Svetlana 6D22S rectifier valve installed at the output terminal of the ripple filter works as a B+ supply slow-start circuit to protect 300Bs, allowing a grid bias to apply earlier than B+ 450V supply voltage. The 6D22S, due to its 30-second heat-up time, gets rid of a timer relay and a damping power resistor lined up in a B+ supply.

SV300B Grid Bias Supply

Plitron #754709 does not provide any appropriate taps for SV300B grid bias requiring approximately –100V DC . . . only a couple of 40V AC taps. So I needed to implement a voltage tripler that steps up from 40V AC to 130–140V DC (unregulated). A simple zener diode voltage regulator with a filter followed by the voltage tripler produces a stabilized and a rippleless –120V to –90V with a 25kΩ, 20-turn potentiometer.

The functions of this potentiometer are 1) varying a grid bias precisely at every 10mV by one turn, 2) providing enough power-handling capability of 0.5W—even though it is a small 1cm²

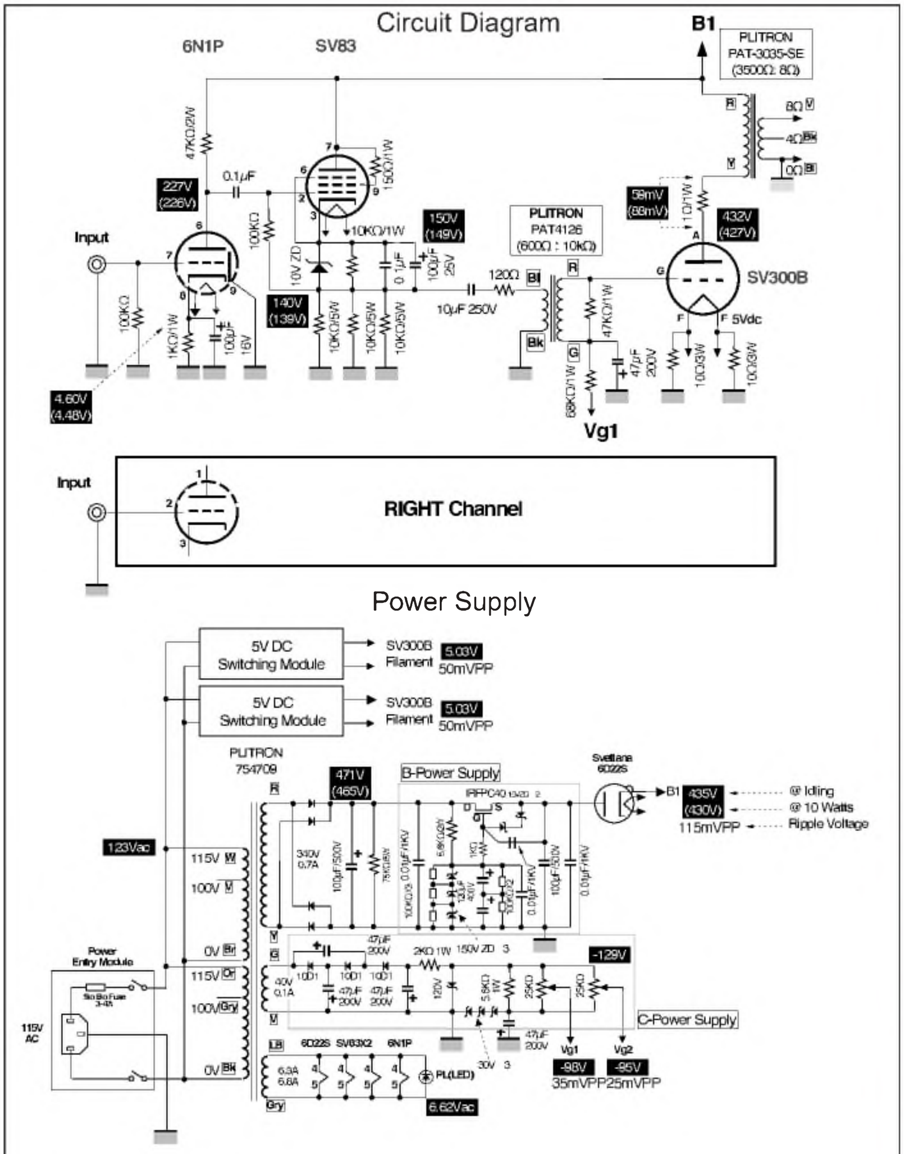

PHOTO 2: PTP board.

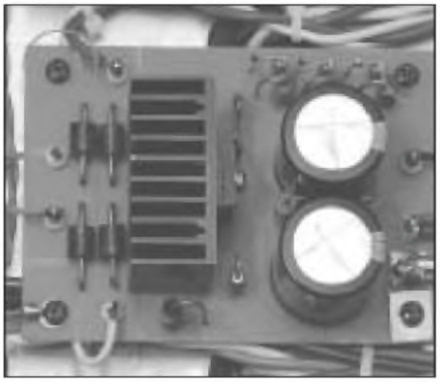

PHOTO 3: B+ power supply.

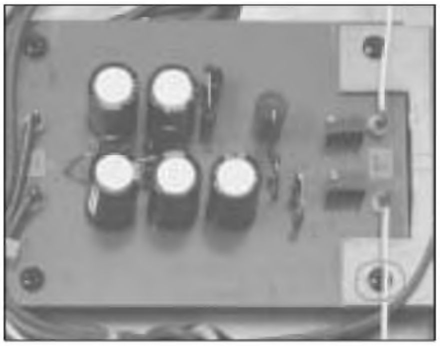

PHOTO 4: C– power supply.

PHOTO 5: 5V power supply.

FIGURE 12: Circuit diagram.

A-2174-12

and a 5mm-thick cube, and 3) high reliability due to its metal-oxide thin layered structure.

SV300B Operating Condition

I have replicated the same operating conditions[2] as shown in my design as follows:

Plate voltage:	435V*
Idling current:	60mA
Grid bias:	–95V - –100V

*Note: Internal forward voltage drop of 6D22S is assumed to be approximately 15V.

Choosing Output Load

The Plitron PAT-30XX-SE series transformer offers three models with various output impedances—such as 2.5kΩ, 3.5kΩ, and 5kΩ. I chose 3.5kΩ, though you could choose any model, depending on whether you prefer a larger output power or a lower distortion than 3.5kΩ.

PHOTO 6: Power entry module.

PHOTO 7: MOSFET.

PHOTO 8: TRITEC speaker terminal.

PHOTO 9: Blue LED pilot lamp.

Circuit Diagram

Figure 12 shows the final circuit diagram. Through the simulation and breadboard experiment, I managed to finalize the parameters. However, I was concerned with the power consumption of SV83, which exceeds the maximum limit of 12W, due to the optimization of the 50kHz square wave fall time. This violation might shorten the lifetime of the SV83, or degrade the tube itself.

So I have relaxed the operating condition a little by lowering plate voltage from 450V to 435V. However, the parameters on SV83 are still marginal, so you might replace either a cathode resistor or a zener diode with a 3.5kΩ or higher, or 11V to 12V. The change will decrease the power consumption of SV83, increasing the operating margin against the maximum limit of SV83 plate power consumption.

PARTS

Custom-Made Case

I placed an order with N-technology in Nagano Ken (the site of the previous Winter Olympic games) for a custom-designed case, made of a stainless-steel plate and a wooden side plate (*Fig. 13*). The case itself could be called "shell-structure," so the case without any welding is durable to accommodate over 20kg to 30kg of heavy transformers and other major components placed on the case top plate.

On the other hand, since this structure needs to attach a couple of steel-

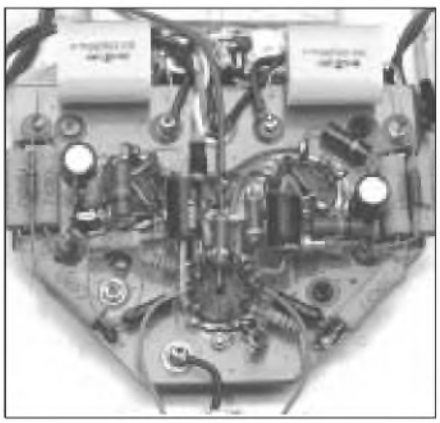

PHOTO 10: PTP board assembled.

made side-walls, exposing the seam between the case cover and its walls, the wooden side walls are attached to both sides.

Wooden Side Board

I used a walnut, or equivalent, wood, sandpapered and oil-stained, and measuring 430 × 57 × 12mm (17 × 2¼ × ½″).

Metal Feet

This is my favorite component—made by International Audio Group (IAG) in Texas—that you put over the bottom plate.

PTP Board

Also made by IAG, this PTP board (*Photo 2, Fig. 14*) allows me to assemble the circuit over a PCB neatly and nicely without any extra wire, with only the lead wires of components. This structure secures a firm star grounding, is simple and easy to assemble, and offers easy maintenance.

Power Supply PC Board

Power supply circuit boards of B+, C–, and 5V for the 300B are assembled on both etched and mechanically hand-stripped PCB (*Photos 3, 4,* and *5*). They offer the following advantages: 1) minimizing hum generation, 2) simplifying the assembly and wiring, 3) enhancing the reliability, 4) improving maintenance, and 5) simplifying a PCB module change.

Power Entry Module

The advantage of this module (*Photo 6*) is to allow me to wire a couple of leads from a power transformer directly without any extra wires, thus enhancing the unit's look. The built-in AC power switch makes the entire AC wiring compact as well.

MOSFET

I chose the IR-made IRFPC40 (*Photo 7*) because of its price—only 100 yen per piece, (≈80¢) in Akihabara, Tokyo. I strongly suggest you get several extra spare pieces, since MOSFETs are quick to die due to mishandling and electrostatic shock.

Speaker Terminal

I chose the TRITEC (Japan) gold-plated speaker terminal because of its quality, reliability, and its commercial-quality look (*Photo 8*).

Pilot Lamp

For some reason, I like a bright blue LED pilot lamp (*Photo 9*). A 12V DC model was driven by a 6.3V AC, allowing me to ignore the polarity when wiring.

ASSEMBLY

The Power Macintosh G4/500MHz with Claris Draw software, under Mac OS

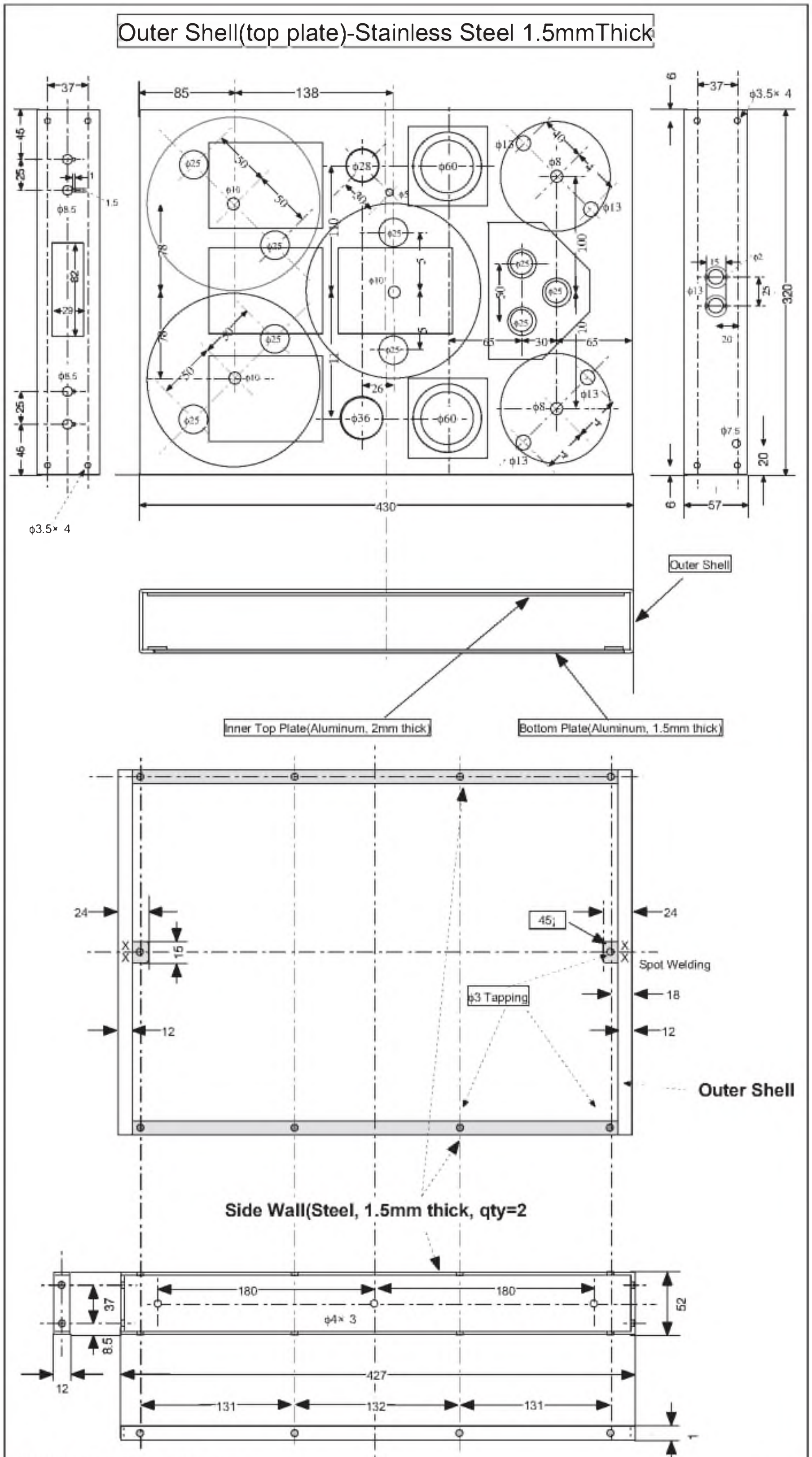

FIGURE 13: Case drawing.

9.2J, produced a good-looking floor plan. (See reference 1 for more information.)

Overall Assembly Sequence

1. Fix a couple of wooden side plates over the inner side plate with a few screws and wood adhesive.
2. Fix all metal spacers, 6D22S socket, and fixing frame of electrolytic capacitors over the inner top plate.
3. Slide the inner top plate with side plates into a stainless top plate. Fix

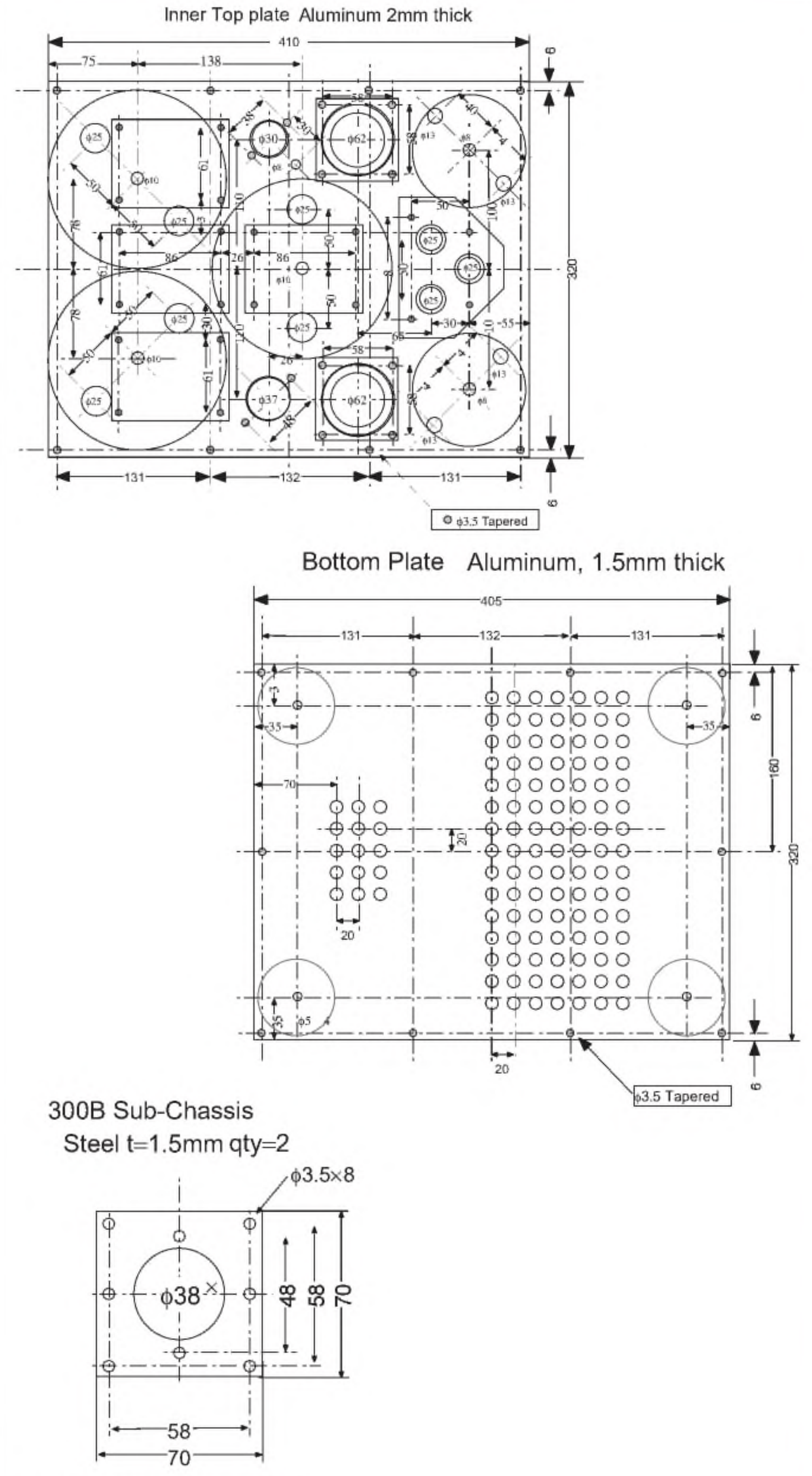

FIGURE 13: Case drawing (cont.) .

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these components with hex stainless bolts.

4. Fix RCA jacks, speaker terminals, 300B sub-chassis, power entry module, and pilot lamp.
5. Fix an electrolytic capacitor over the top plate.

Assembly of Voltage Driver Circuit Over a PTP Board

The PTP board requires less labor and a shorter assembly time of the circuit for a beginner than a conventional assembly scheme, and also eliminates extra wiring cable, shortening the wiring length as much as possible. Pin 9 of the 6N1P internal shield was grounded to minimize the crosstalk in the over 100kHz high-frequency range between the two channels.

The extra wires needed are 1) 300B filaments, 2) B&C power supplies, 3) pilot lamp, 4) electrolytic capacitor, and 5) an internal wiring between RCA jack and a grid pin of 6N1P. I believe this structure will bring some benefits even to beginners, since the board could be separately placed on the table when assembling the components simply and easily (*Fig. 15*, *Photo 10*).

Filament Wirings

Place a voltage driver and a couple of CF valves over the PCB so that 6.3V filament pins (4 and 5 pin) wirings of each socket face inward, and its wire length should be as short as possible. The wirings of each 4 and 5 pins are tied to each other and a couple of 6.3V taps from a power transformer (*Fig. 15*).

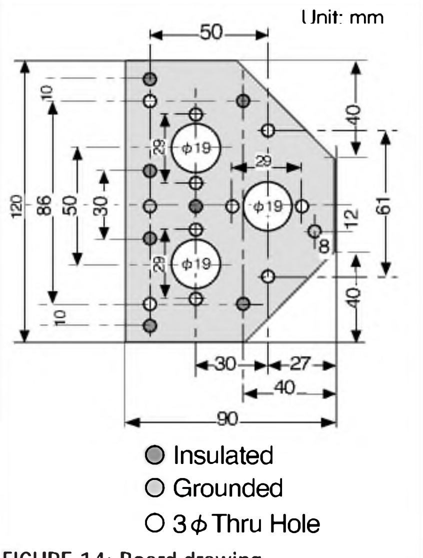


FIGURE 14: Board drawing. A-2174-14

PARTS LIST
AMPLIFIER

PARTS	SPECIFICATION (MANUFACTURERS)	QTY.
Vacuum tube	SV300B matched pair (Svetlana)	2
Vacuum tube	6N1P (Svetlana)	1
Vacuum tube	SV83 (Svetlana), or EL84EG (Ei Elites)	2
Socket	4 pin UX	2
Socket	9 pin	3
Output transformer	PAT-3035-SE (Plitron) or PAT-3025-SE, PAT-3050-SE	2
Zener diode	RD10B NEC or 10~12V 1~3W equivalent	2
Resistor	1Ω 2W 1%, or 1~10Ω 2W equivalent	2
Resistor (300B filament)	10Ω 2~5Ω metal oxide, or 11~30Ω equivalent	4
Resistor (damping)	120Ω ½W carbon	2
Resistor (SV83 plate)	150Ω 1W carbon	2
Resistor (6N1P cathode)	1kΩ 1~2Ω carbon, or 1.2kΩ	2
Resistor (SV83 cathode)	10kΩ 1W carbon, metal oxide	2
Resistor (SV83 cathode)	10kΩ 5W metal oxide	6
Resistor (6N1P plate)	47kΩ 2W carbon	2
Resistor (300B grid)	47kΩ 1W carbon	4
Resistor (300B grid)	47kΩ 1W carbon	2
Resistor (6N1P grid)	100kΩ ½W carbon	2
Resistor (SV83 grid)	100kΩ ½W carbon	2
Capacitor	0.1µF 50WV film, Nittsuko	2
Capacitor	0.1µF 630WV film, Angela	2
Capacitor	10µF 250WV film, Shizuki	2
Electrolytic capacitor	100µF 16V, Nichicon Muse	2
Electrolytic capacitor	100µF 25V, Nichicon Muse	2
Electrolytic capacitor	47µF 200V	5
POWER SUPPLY		
Power transformer	754709, Plitron	1
Switching module	Vout 5V 2A, AC input: 85–132V AC,	2
	Stanley, PowerBoy PB10-0500 or, COSEL, LCA10S-5	
Power supply PCB	100 × 75mm 1.6mm thick, epoxy glass PCB with photo resist	2
Power supply PCB	100 × 75mm 1.6mm thick, epoxy glass PCB	2
Vacuum tube	6D22S, (Svetlana)	1
Socket	9 pin, (Svetlana)	1
MOSFET	IRFPC40, IR or Vdss=>600V, Ids=>5A,	1
	Rds=<1~2Ω equivalent	
Diode	2NU41 Toshiba or 1000~1200V >2A equivalent	4
Diode	200V 1A equivalent	3
Zener diode	15V 0.5W (Hitachi)	2
Zener diode	30V 1W (Hitachi)	3
Zener diode	120V 2W (Ishizuka Denshi), or 120V 3W 3Z120 (Toshiba)	1
Zener diode	150V 2W (Ishizuka Denshi), or 150V 3W 3Z120 (Toshiba)	3
Resistor	1kΩ ¼~½W carbon	1
Resistor	2kΩ 1~2W carbon	1
Resistor	5.6kΩ ~1Ω carbon	2
Resistor	75kΩ 5W metal oxide	1
Resistor	100kΩ ½W carbon	5
Capacitor	0.1~0.01µF 500WV or 1kW ceramic	3
Electrolytic capacitor	47µF 200V	5
Electrolytic capacitor	120µF 400V	2
Electrolytic capacitor	100+100µF 500V Nichicon Cerafine	1
Pin terminal	Teflon insulated	
Heatsink	LEX 20PB55-30, L = 55mm, H = 30mm, 8.9°C/W	1
Potentiometer	25kΩ 0.5W, (Burns)	2
MISCELLANEOUS		
Custom made case	430 × 320 × 57mm, N-technology	1
Metal feet	IAG made	4
Power entry module	120V, 6A, 47–440Hz, UL, CSA, VDE,	1
	Corcom, 250V 3~4A (Slo-Blo)	
AC power cable	Hospital grade, 3m long	1
Metal spacer	3 φ 13mm long	20
Metal spacer	3 φ 30mm long	8
Hex stainless volt	3mm × 6mm long	
Pilot lamp	12V ultra bright LED lamp, DUL-7HJT	1
	(Sakazume Seisakusyo)	
RCA jack		2
Speaker terminal	CU-T80 TRITEC	4
Wires	Teflon insulated	
Isolation tube	Teflon, I.D.=1mm	
Wooden side board	430 × 57mm, 12mm thick, oak or equivalent	2
PTP Board	120 × 90mm IAG custom made	1

B, C, and 5V DC Power Supply

All power supplies were in a PCB base module, requiring only an installation over the inner chassis through spacers. *Figures 16* and *17* show PCB patterns of each board.

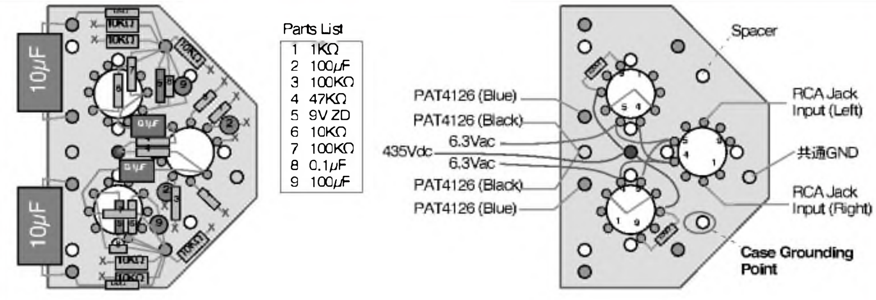
Final Assembly

After installing PCB modules—PTP (voltage driver), B+, C–, and 5V supply board—then wire all connections between these modules and major components—transformers, electrolytic capacitors, power entry module, inputs, and speaker outputs to finish up. *Photos 11–13* show various perspectives of the finished unit.

INSPECTION AND ADJUSTMENT

Wiring Error Check

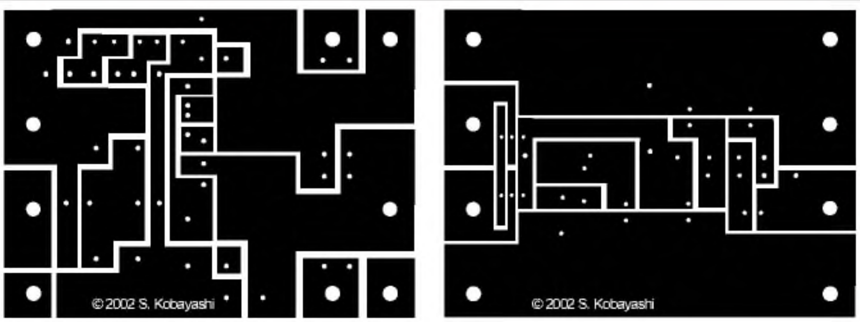
Prior to turning the AC power switch on, double-check all the wiring in the



Assembly Sequence
1) Filaments
2) Grounding Components (Cathode resistors, etc.)
3) Remaining Parts

FIGURE 15: PTP board wiring.

A-2174-15



B+ Supply **C- Supply**

FIGURE 16: PCB pattern drawing (B&C).

A-2174-16

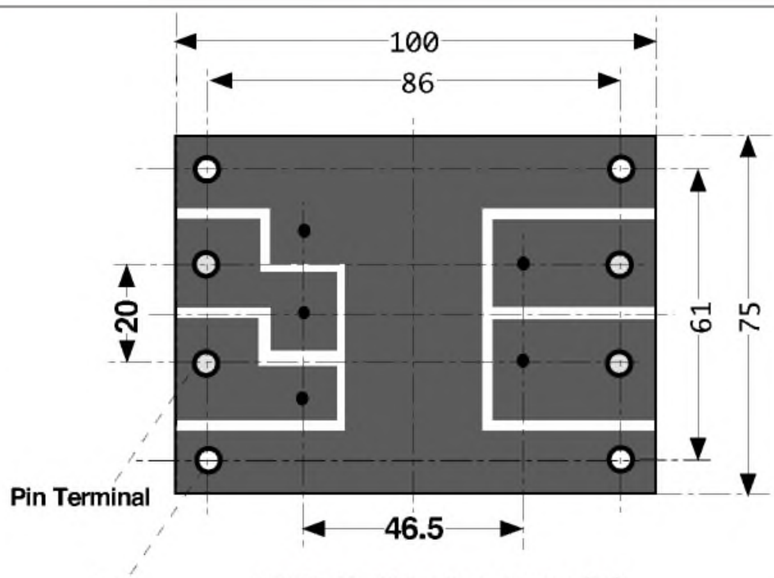


FIGURE 17: PCB pattern drawing (5V).

A-2174-17

circuit. I strongly suggest you use a digital multimeter in the following inspection.

Filament Voltage Check

Measure filament voltage without any valves. The meter reading might be 6.3 to 6.8V AC when probing at filament pins of 6N1P and SV83s. If OK, then go to the next step.

Grid Bias Voltage Check

Measure the grid bias voltage at 300B grid pins, after setting the potentiometer counterclockwise. If the reading is approximately –120V DC, then continue to the next step.

B+ Power Supply Voltage Check

Once you turn off the AC power switch, insert only 6D22S into the socket, and then turn the AC power on again. Measure B+ supply voltage after a couple of minutes, and ensure that the voltage is approximately 450V DC. If good, then continue.

300B Idle Current Adjustment

Once again, turn the AC power off; insert all valves into the sockets. Measure the voltage drop across a 1Ω resistor placed at 300B plate, so that the meter reading is approximately 60mV, while turning a potentiometer with a screwdriver very slowly. The voltage reading might vary as time goes by, so check the value again in 30 or 60 minutes. That concludes the adjustments.

PHOTO 11: Rear of completed unit.

PHOTO 12: Overhead view.

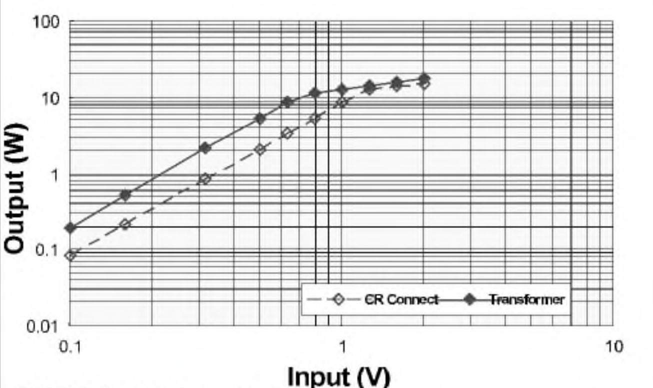


FIGURE 18: Input versus output.

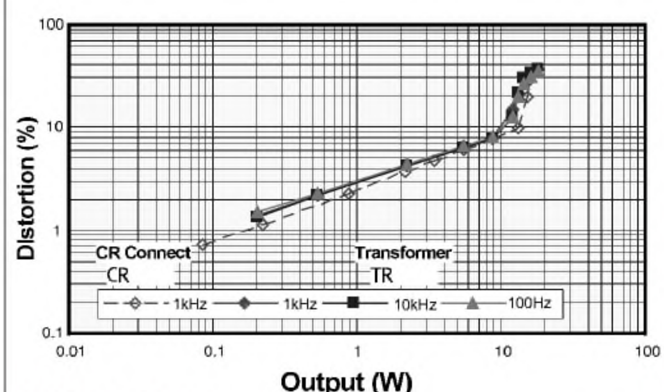


FIGURE 20: Distortion.

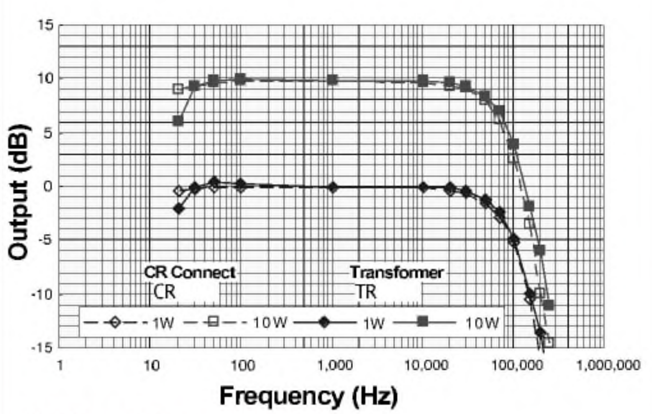


FIGURE 19: Frequency response.

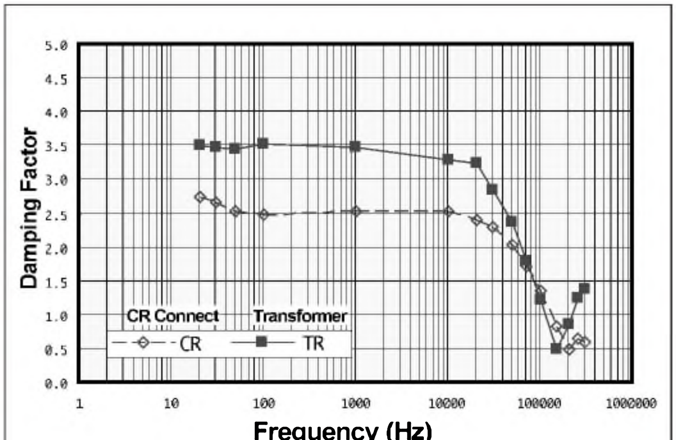


FIGURE 21: Damping factor.

PERFORMANCE MEASUREMENT

I measured the basic characteristics, using the equipment listed at the end of the article.

Input Versus Output Characteristic

The input sensitivity was approximately 0.7V RMS, producing the maximum output power of 10W, while its linearity extends up to 8W (*Fig. 18*).

Frequency Response

The high-end cut-off frequency was approximately 70kHz at both 1W and 10W, while a low-end cut-off frequency was approximately 25Hz (*Fig. 19*). Since the AC coupling capacitor value regulates the low-end frequency, if you wish to lower the cut-off frequency below 20Hz, for example, then replace 15 to 22µF with 10µF.

The frequency response of this amplifier exceeds a CR-connected amplifier[2] by approximately 1dB in the high-end frequency area, though overall characteristics are mostly identical.

Distortion

The distortion (*Fig. 20*) increases linearly as the output power increases toward the clipping level over 10W, and it does not depend upon the frequency. The tendency is also identical to the CR-connected amplifier.

Damping Factor

I calculated the damping factor (*Fig. 21*) under the so-called "on-off method" at 8Ω resistor loading and 1W (2.83V RMS). The factor was 3.5, which is relatively high for a non-NFB amplifier.

Output Waveform

I took oscilloscope views of both square and sine waveforms of 100Hz, 1kHz, 10kHz, and 50kHz, even at 1W and 10W, under an 8Ω resistor load (*Fig. 22*). It appeared that Plitron toroidal transformer shows non-ringing waveform even at 10kHz. At 100Hz, it showed a rather higher level of sag than that of a CR-connected amplifier. Overall, the performance difference between a CR-connected amplifier and this transformer-driven amplifier is negligibly small, though it would be interesting to check the sound difference later.

Residual Output Voltage

I measured a residual noise voltage at an output terminal when the input terminal was grounded. The values of both channels were 1.3mV RMS. I heard a barely audible noise from right in front of the speaker, but it disappeared when I backed out 50cm away from the speakers.

LISTENING IMPRESSIONS

I have added a new B&W Nautilus 802 speaker system in my home in Texas, and placed it over the solid concrete-based carpet in my living room, reproducing a firm and solid bass sound. I used the SV300B PP amplifier and SV300B SE amplifier (CR-connected) for comparison with this amplifier.

Vocal

It seemed the amplifier gave me a depth and a width of presence in vocal categories among the Japanese pop (J-pop), jazz, and classical music as compared to a CR-connected amplifier. Also it gave me a relaxed feeling. Particularly, J-pop, such as "Midaregami" by Gotoh Ryoko (CD#4), originally sung by a well-known Japanese singer, Ms. Misora Hibari, gave me a nice, real, and strong presence due to her song with a piano.

Drums, Electric Bass

I am surprised that even 10W output could drive Nautilus 802 (92dB/1W/1m) efficiently at the level for family listening. The sound volume seems mostly identical to that of 300B pp amplifier. It also surprised me that a drum and an electric bass guitar sounded so nice with this amplifier driving a couple of speakers (#20).

Classical Strings

Electric violins (#9) are interesting too. I was impressed with the high-end tones in the strings.

Classical Vocal

The amplifier produced an ultra low sound sufficiently (as if I were punched by the bass sound of pipe organs from speakers), and the voice of the alto sounded relaxing. I thought the sound quality was much better than that of the CR-connected amplifier. CD #10 also greatly impressed me, as though a big choir were spread widely and deeply over the big stage in a hall.

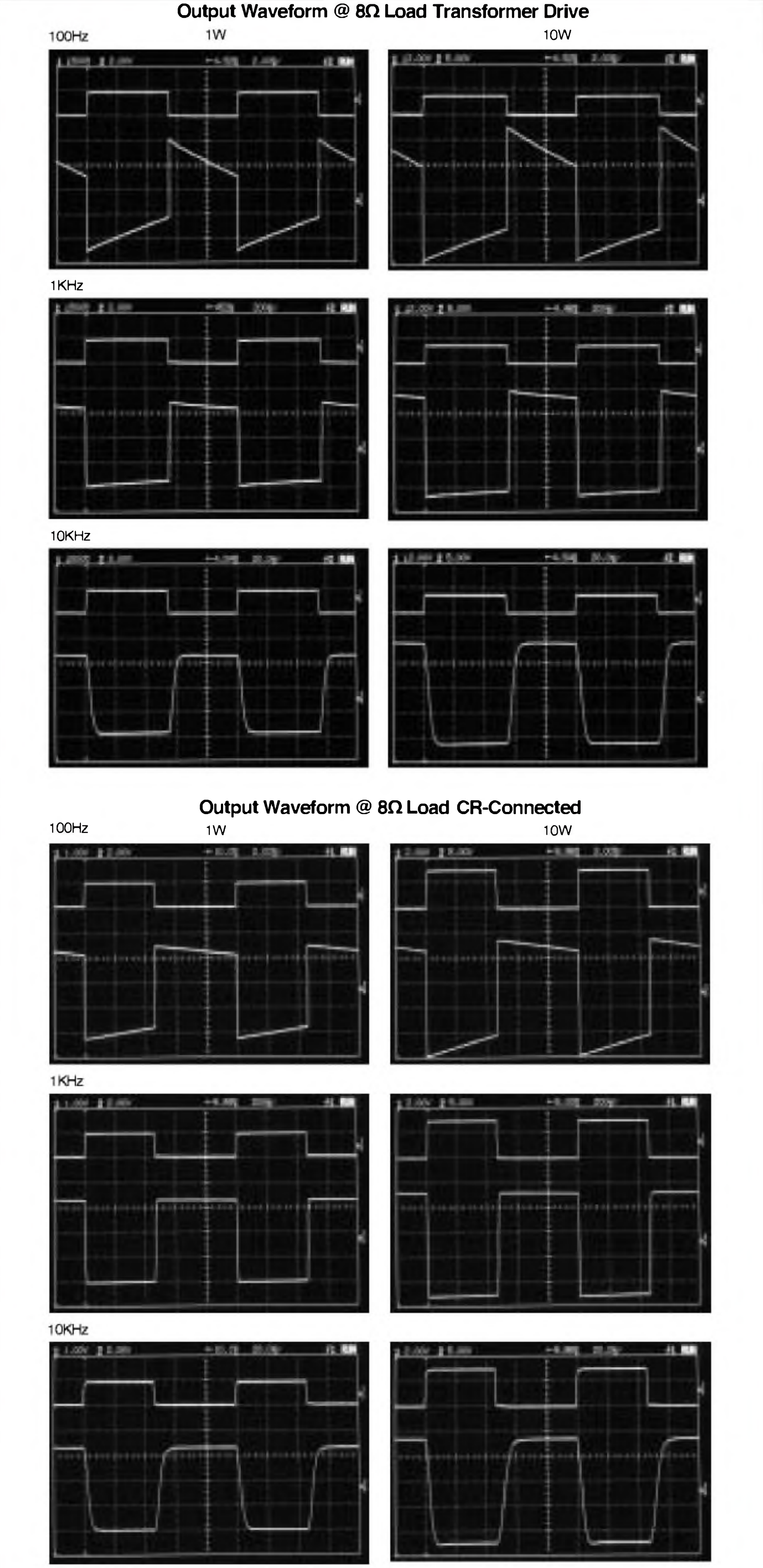


FIGURE 22: Waveform.

A-2174-22

Brass Instruments

It is said that the amplifier using an interstage E-I-cored transformer is poor at reproducing over 20kHz tone of brass instruments. However, I did not note any such weakness. In reality, most of the amplifiers using a conventional E-I-cored interstage transformer cut off their frequency response up to 20kHz to 30kHz.

Summarizing my impression of this amplifier sound, I did not detect any inferiority of bass sound presentation when compared with a CR-connected amplifier, as I might have expected. I believe this amplifier might exceed the sound quality of a CR-connected amplifier a little bit in terms of presentation of width and depth of the sound quality in general. Overall, I am quite satisfied with this interesting performance and its sound quality. ❖

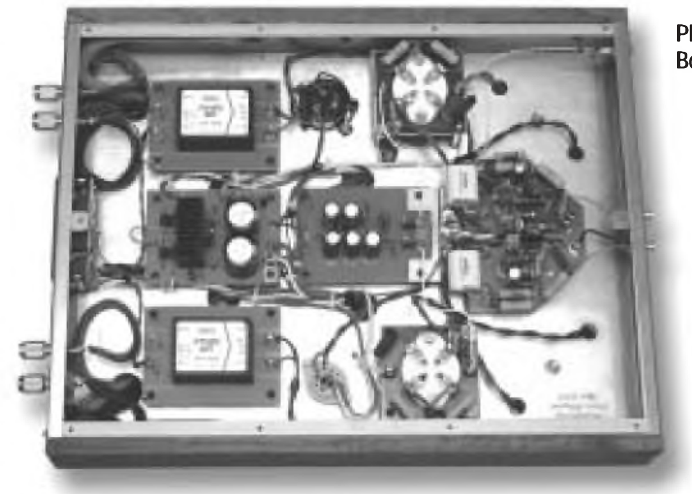


PHOTO 13: Bottom view.

SOURCES

Japanese-Pops

1. Katoh Tokiko, *My Best Album*, UICZ-4001
2. Inoue Yosui, *UNITED COVER*, FLCF-3863
3. Nagafuchi Tsuyoshi, *Ore no Taiyo*, FLCF-3780
4. Tanimura Shinji, et. al, *Misora Hibari Tribute*, COCP 31003

Classic

5. Ozawa Seiji, *2002 New Year's Concert*, UCCP-9413
6. Watanabe Reiko, *J.S. Bach*, WPCS-11101
7. ARVO PART, *I AM THE TRUE VINE*, HMU 907242
8. Yo Yo Ma, *The Best of Yo-Yo Ma*, SRCR 2294
9. Bond, *Born*, Decca 289 467 091-22
10. Robert Shaw, *Amazing Grace*, Telarc CD-80325
11. Maria Callas, *The Very Best of Maria Callas*, EMI 7243 5 57230 2 4

Rock, R&B, Jazz

12. Sade, *The Best of Sade*, EK85287
13. John Lennon, *Imagine*, CDP 7243 5 24858 2 6
14. Elvis Presley, *Moody Blue*, RCA 07863 67931-2
15. Rita Coolidge, *Out of The Blues*, BRL BEA-51572
16. Telarc's *Got the Blues*, CD-83468
17. Telarc, *The Blues White Album*, CD-83553
18. Eric Clapton, *Unplugged*, Reprise 9 45024-2
19. Holy Cole Trip, *Don't Smoke in Bed*, CDP 0777 81198 2 1
20. Roger Waters, *Amused to Death*, CK 47127

RESOURCES

Tec-Sol, Inc.
435-0016
Schizuoka-ken, Hamamatsu-shi, Wada-cho 514
053-468-1201
Fax: 053-468-1202
Tec-Sol Inc. website, http://www.tec-sol.com/

PM Components website HP,
http://www.svetlana-tube.com/

Plitron website, http://www.plitron.com/
Svetlana tubes, Plitron toroidal transformer

International Audio Group
2790 Brighton Road,
Killeen, TX 76544
Phone/Fax: 905-829-5858
PTP board, metal feet

N-Technology
389-0515
Nagano-ken, Chiisagata-gun, Tobu-cho, Tokita 1275-
0268-63-7991
Fax: 0268-63-7990
Custom-made case

MEASUREMENT EQUIPMENT

Audio Analyzer HP-339A
Audio Generator Kenwood AG-204D
600Ω ATT HP-4437A
Homebrew 8Ω 50W 2ch. dummy load
Digital Multimeter Fluke 8020B
Oscilloscope HP-54600B

LISTENING REFERENCES

CD Player, TEAC VRDS-50
Line amplifier, 6N1P+ PAT-4126-02 line transformer
Speaker, B&W Nautilus 802

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2. Kobayashi, Satoru, "SV300B SE Stereo Amp Using Plitron Toroidal Transformers," *Glass Audio* 6/00, p. 1.
3. Kobayashi, Satoru, "6N1P Line Amplifier no Seisaku," *MJ* 3/99.
4. "Using the Svetlana 6N1P Dual Triode in New or Existing Designs," Technical Bulletin No. 43, http://www/svetlana.com/ (closed as of 5/2002).

Faster, Straighter, Nicer

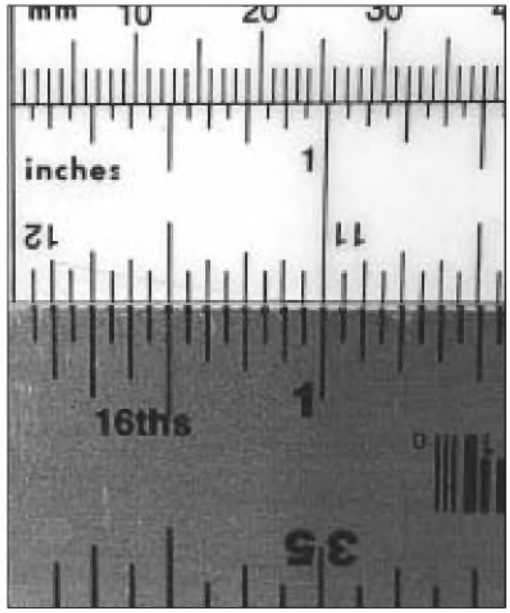


PHOTO 1: The shadow makes this look worse than it is, but the first sixteenth on the bottom scale is more like 3⁄32″, making the scale unusable from the end.

Here are some helpful, common-sense building tips to ensure that your construction projects appear true and nice-looking.

By Hilary Paprocki

Outside of developing good-sounding circuitry, probably the biggest challenge for the home constructor is making a device that looks sharp, straight, and professional. It might be a little easier if we all had computer-controlled machining stations. But there is plenty that we can do to leverage the accuracy of our handwork to a higher level.

Since this is a magazine that is basically about building things, it's probably pretty reasonable to presume that most of you have workshops of one sort or another. The elaboration of these shops could range from a trunkful of parts and tools under the kitchen table to three walls of workbenches and shelves in a clean, dry suburban basement.

The shop is not the object, apart from the enjoyment you get from being there. But we're making things, and the quality and general niceness of what we make will vary depending on our experience and on the tools we use. I believe that it is a mistake to spend time lusting after that $900 table saw unless you're really at the top of your amateur class and something like that would make a difference to you. At the duffer level we can do some pretty good work as long as our measurements are precise and we hit our marks.

I can't help you hit your marks, but I do have an accumulation of ideas regarding making marks efficiently. Forget the table saw for the time being—and maybe give yourself a little $100 present to set yourself up with some decent shop gauging.

THE FIRST STEPS

What are you measuring with? A ruler? A tape measure? OK, fine, you also need a really good long metal scale. Mine's a 24″ stainless job; there are painted steel ones at the hardware store that might be fine as long as they're not so thick that the line you're reading is 1⁄8″ away from the line you're trying to mark.

Make sure you pay enough (*Photo 1*). The poor guy with the cheap scale will go crazy until he discovers that his first sixteenth is more like a sixteenth-and-a-half on one end.

Then hook your tape measure onto the end of the scale and check it, making sure that the innermost edge of the hook is touching, as it would on the edge of a board (*Photo 2*). Do the numbers match? Unless you've done this before, they probably don't. You're supposed to bend the little hook on your tape in or out every once in a while to keep it accurate—especially if you ever drop the tape measure on the hook.

Check your yardsticks, too. They're wood; they'll vary a bit. You probably shouldn't use a wooden measuring device, except in one instance I'll describe later.

As long as you're going to be reading a scale and scratching little marks onto your material, why not have a nice pair of those drugstore reading glasses by your bench? Extra-strong ones will magnify the scale and the pencil tip nicely, which improves your accuracy. If you already wear glasses, just put these on over your regular ones for the moment. Wearing two pairs of glasses when you're finding the line with your

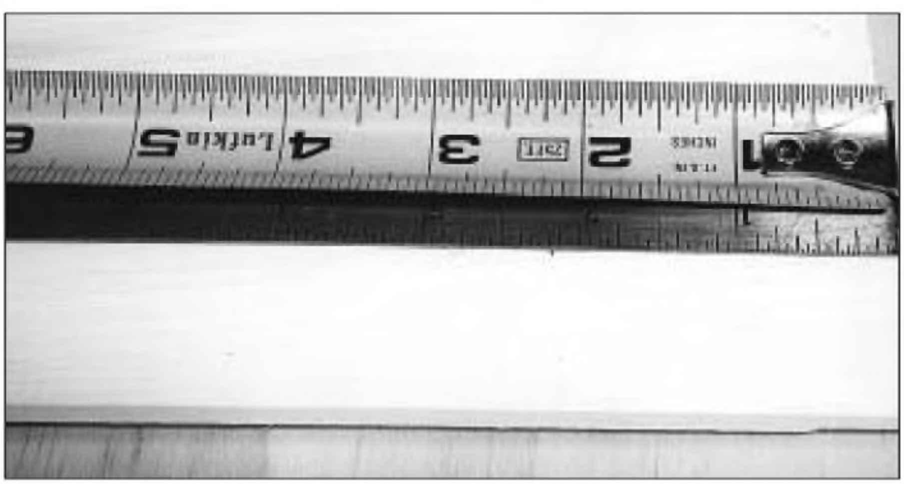


PHOTO 2: The hook on your tape measure is adjustable—have you ever checked it?

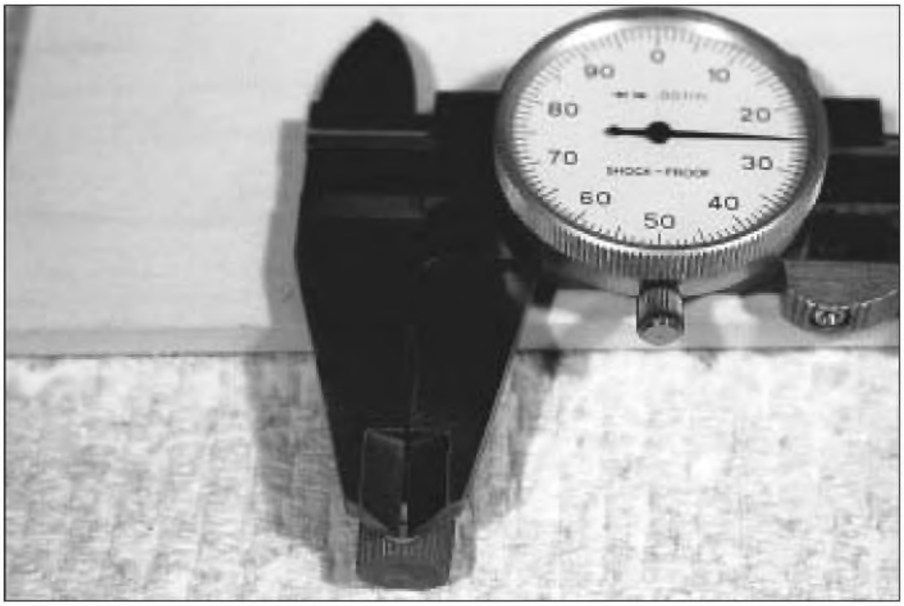


PHOTO 3: You don't need to actually grab an object to measure it. With a magnifying glass you can see that there is a 0.025″ hole in this spray can actuator.

PHOTO 4: Steel bars. Pretty simple.

center punch is an enlightening and very rewarding experience.

By the way, if you're really just beginning and don't have a center punch yet, put this magazine down right now and go get one. We don't mark a spot and try to drill a hole there; we mark a spot, whack a crater there with a punch and a hammer, and drop a small drill into the crater.

Shop carefully for those drugstore glasses. Not all the expensive ones are the good ones. Look for fine glass lenses and real hinges that snap open and shut; surprisingly they don't cost much more than the cheesy ones. You just need to find a store that has the right kind.

And speaking of light, you can't have too much of it when you're measuring and marking. You know where to find lights . . . get some!

NEXT STEP

Buy yourself a present: a dial caliper (*Photo 3*). You can obtain a 6″ capacity tool for maybe $80 for a good brand name, or $30 or less for one from China. I bought the China one.

It was a long time before I broke down and got a caliper, and now I use it all the time: checking screws for the hole size they'll need; checking panel thickness; checking connector pin diameters; checking grommet sizes; unlabeled wire gauges, guitar string gauges, capacitor diameters; even (get this) checking spray can aperture sizes (see the "Totally Unrelated" sidebar).

AVOID MEASUREMENTS

Of all my discoveries in all the years of building, this was the most profound: My work has improved 100% when I stopped measuring.

What?

Well, suppose you're going to drill holes for rubber feet in a panel, or to mount a circuit board, or a line of jacks or switches. You line the scale up along the left edge and count ¾″ from the bottom of the panel, make a mark, line up the scale on the right edge, make another mark at ¾″, draw a line, and hope that the line hits both marks and that they are accurate.

Well, to heck with that. One of the best practices is to avoid measurements whenever you can. Suppose you're cutting 40 pieces of 3′ long 2 × 4. Measure and cut the first one, then mark it as a master and use it to trace a line for each of the others. A lot of potential variability will disappear, and you will spend less time and effort.

Just make sure you use the same master every time, instead of the successive parts—otherwise, the last piece will be 40 pencil-lines'-width longer than the first. Even if you do use one master, they'll all be one pencil line longer than the first one unless you sand a tiny bevel on the edge you use to mark the others.

Getting back to that panel, you must have a bar in your workshop, and preferably several bars. If your town has an industrial supply shop, ask them whether they sell steel bar stock. And if they do, ask whether they have inexpensive ends and cutoffs (they do).

I have a bunch of little bars, generally about 14″ long, for which I spent a dollar or two each (*Photo 4*). They are a mix (1″ × ⅛″, ½″ × ¼″, ¾″ × ⅛″, ½″ × ½″, and so on). You could make your own out of a nice, stable wood such as maple if you can hit the dimensions right on and exactly flat on your $900 table saw. But I say just treat yourself to a nice piece of steel.

Now that you have these bars, you don't need to measure anything under an inch or so. Put the enclosure on the table. The line of switches should be 1¼″ above the bottom edge; just lay the 1″ and ¼″ bars against the box and draw the line (*Photo 5*). It's straight, it's level, it's in the right place, and it took you ten seconds and zero effort.

Corner holes on a panel? Place a chunk of wood on the table and hold the panel standing up against it. Put the bar against the panel and draw.

You'll need to wipe off the metal bars with some ordinary paint thinner, and probably clean them up a little on all faces with a flat file, because they'll have a few scratches and dents. Lay the file flat on the surface and pull it back and forth with your fingertips until you can tell that the bar face is more or less smooth. Easy does it. Then write the dimension on each face and you're equipped.

The people who sell you the bars can also sell you nice aluminum plate for panels—flat, shiny, beautiful, any thickness you wish, and very reasonable.*

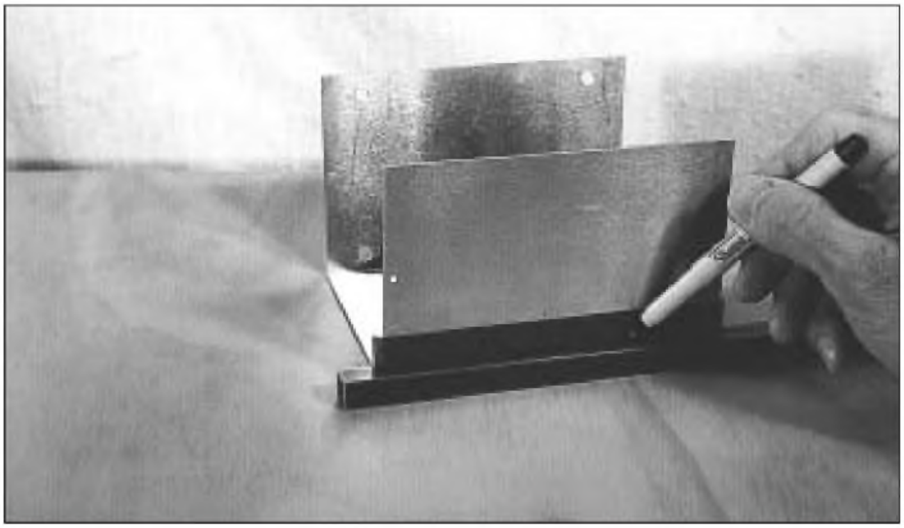


PHOTO 5: Put some tape on the box, put the box on the table, put the bar on the table, draw the line on the tape on the box on the table with the bar.

I also have a bar for my router. I made it out of ¼″ clear acrylic, which is a nice, stable material and looks cool. Its width is the distance from a ¼″ bit to the flat edge of the router's base (*Photo 6*).

It was a piece of cake to make; just put the plastic against a stop, run the router along the stop, and the bar makes itself (*Photo 7*). So now when I need to rout a pocket for a shelf, say, inside a speaker cabinet, I draw a pair of lines where the pocket should go, hold the plastic bar to one of the lines (not covering the other one), and clamp a stop board to the panel against the other side of the bar (*Photo 8*). Take away the plastic, run the router along the stop, and one side of your channel is done, no effort.

Place a ½″ metal bar between the stop and the router and there's your guide for the other side of the pocket: ½″ bar plus ¼″ router bit equals ¾″—unless the bit is undersize. Check the pocket with the piece of wood you intend to stick in it before you unclamp everything. You might need to make another pass with some index cards stuck in the stack to widen the pocket a little.

Speaking of routers, the pros use them to cut their speaker holes, you know. Go to the local Woodcraft store and obtain a Jasper circle guide; both the store and the guide are quite nice. Or make your own, if you prefer. They even have calibrated stick-on faces at the store, so the guide that you build will look like a commercial product, with labeled holes and everything. You'll cut rounder holes with less effort than ever.

Speaking of measuring with nontraditional objects, do you want to measure a hole or a gap? Your drills make fine gauges for general work. Use the end that isn't pointy.

HIGH SCHOOL MATH

You'll need to cut an angle once in a while. I needed to make a dozen speaker stands once, overhead, angled down at 5°. A little protractor won't do—it's impossible to draw an accurate 8″ long line with a 3″ radius pencil mark. You can probably buy a fancy gadget for this task, but you might not need to.

Your local home-improvement store has commercial vinyl floor tiles that are a very square 12″, for less than a dollar each. Lay one atop another, sides even and ends offset, to draw a straight parallel line an inch or so in from the edge on the lower tile. Then figure out a triangle:

(tan of angle)(adjacent side) =
opposite side

So you look up "tan," and the trig chart, calculator, slide rule, or whatever says that "tan" of 5° is 0.0875. The side adjacent to the angle is the 12″ width of the tile. Twelve inches times 0.0875 is 1.05″; that's the height of the triangle. Call it $1\frac{1}{16}$″, which is only ten thousandths off at 12″ out from the angle. Measure up the opposite edge and draw the line across the tile.

Cut along the angle with a utility knife and a clamped-down straightedge, gently clean up the cut edge with a sanding block, and you have a permanent 5° template that's the same every time. It's well worth 68¢ to run off a dozen exactly consistent parts without

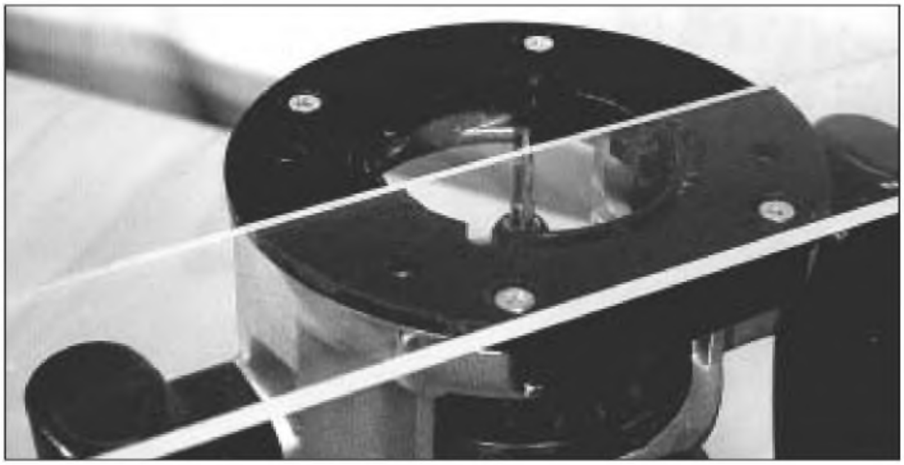


PHOTO 6: The router bar provides the distance from the bit to the edge of the router base.

having to measure anything more than once. Of course, if you're doing just one or two wide panels, you can draw the triangles right on the panel.

Here's a 17¾″ panel (*Photo 9*). Tan 5° is still 0.0875, times 17.75 is 1.55″. One and 17/32″ is real close, better than a tenth of a degree.

You can take the idea of reference measurements out with you when you're shopping for materials. The world is full of objects that you can use to measure with when you're out and about. Those 12″ floor tiles (the ones on the floor; you usually don't need to carry your own) are useful when you're looking at cables, drapes, or cutoff pieces of material in the store. Just look down, there's a gauge.

You know the measurement of your

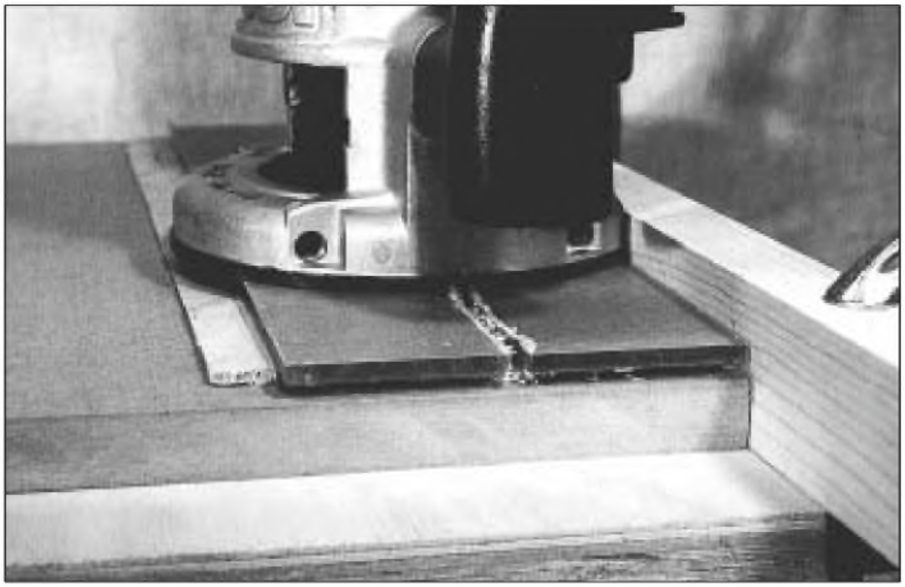


PHOTO 7: It almost makes itself. Just cut the plastic against any improvised fence.

TOTALLY UNRELATED

We all use cans of spray paint once in a while, and you may have noticed that older half-used can contents tend to thicken up in storage and clog the actuator when you come back to the latter half of the can.

You sometimes need to use a pin vise (tiny finger-powered chuck) to bore the actuators out a little so that you can use the remainder of the can. Use fresh, sharp drills in the 0.025″, 0.030″, 0.035″ range, not more, and a light touch.

And when you're done painting, the long nozzle on your spray contact cleaner solvent tucks nicely into the can end of the actuator. Just put on your safety glasses and give it a short zap. It'll be clean, clean, clean.

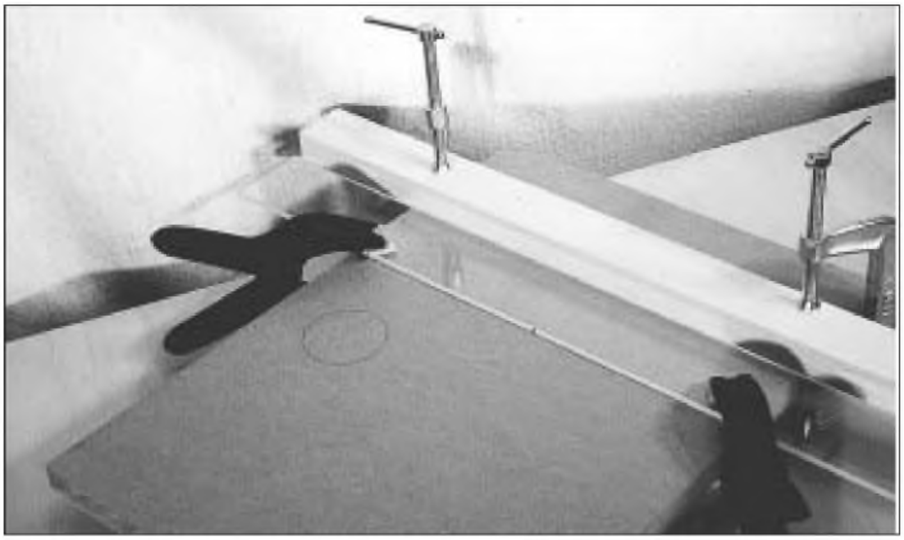


PHOTO 8: And now you can use the plastic to locate a fence exactly the right distance from the line you want to rout along.

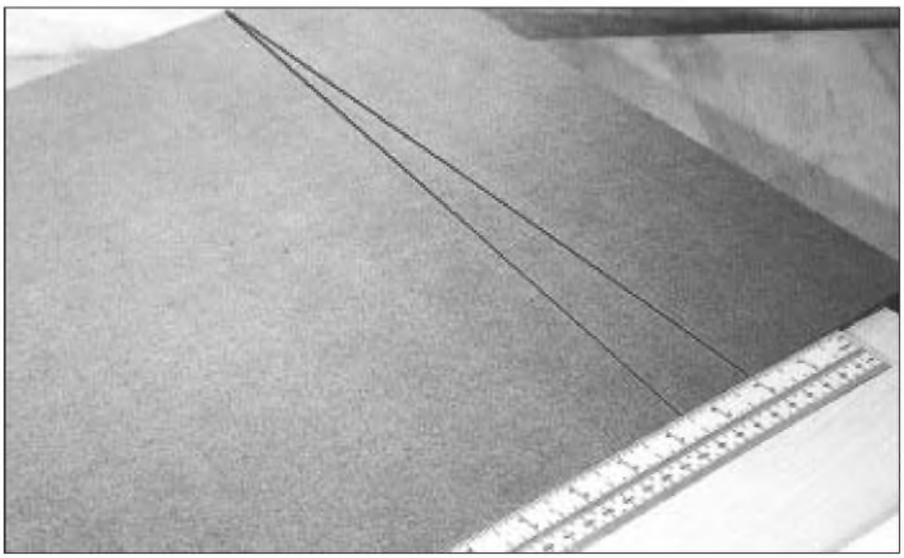


PHOTO 9: A scale and a calculator will give you more accurate angles than a line protractor.

shirt sleeve; it's a gauge that you always have with you. A penny is pretty darn near ¾″ in diameter. A nickel is 2mm thick on the rim.

These kinds of ideas can travel with you all the time. But the basic point here is that a reference piece is a quicker, easier, and more accurate way to mark out your material than scales, rulers, and tapes. Happy Building!

*So your picturesque town doesn't have much in the way of industrial supply houses? Try Nolan's in Syracuse (800-736-2204). They've recently consolidated their stores into a master location in Syracuse, and the service is terrific. I just ordered four 1.5′ cold rolled steel flats (the bars mentioned in this article)—the whole thing was maybe eight bucks, and not only were they happy to do it, it arrived in bombproof packaging. Of course, if you need that nice stainless scale and some 0.025″ drills too, they can fix you up. If you like, try the site: www.nolansupply.com. Ask them to send you their catalog—you'll find all kinds of materials, including audio favorites like thin sheet lead and brass rod for cross-chassis control shafts. ❖

Amplification Without Distortion, Recreating a 1920s Amplifier

Return with us now, to those thrilling days of early audio amps and the music of the '20s. **By Larry Lisle**

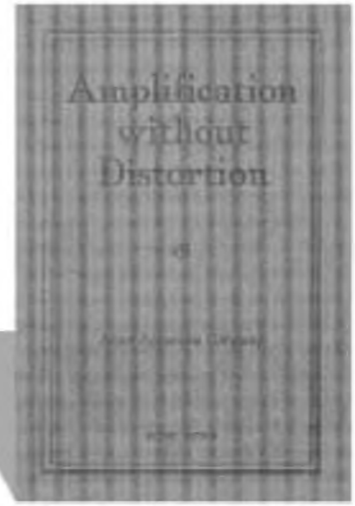


PHOTO 1: (copy of pamphlet cover) What a great title! It got me thinking about what the amplifiers of the '20s sounded like.

Audio has, in recent years, taken a long look at its past and found new sounds in old circuits, such as the single-ended triode tube amplifier.

When I came across a booklet titled *Amplification Without Distortion* (*Photo 1*) from 1923, I couldn't resist buying it. It was by the Acme Apparatus Company of Cambridge, Mass., describing radio and amplifier circuits using their products. I later found another edition from 1926.

I decided to build one of the amplifiers (*Photo 2*) using tubes, transformers, and other parts from the period, just to see how it would sound. It was an interesting experience.

HISTORY

Before proceeding, though, a word on what electronic audio was like during the '20s. Notice that I said electronic audio; mechanical audio was still dominant in reproduced sound during the decade. The first electronically cut phonograph records didn't appear until about 1925, and the first all-electronic phonograph players didn't enter the market until a year later; the Brunswick Panatrope and the Victor Electrola, which were very expensive high-end items. The wind-up phonograph was the common recorded sound reproducer well into the 1930s.

The main use for electronic audio was radio. There are rapid changes in the computer world today, but they're nothing compared to the breakneck, revolution-a-year pace of radio during the 1920s. Broadcasting didn't really get started on a large scale until about 1922 with the regenerative receiver being dominant. Just a couple of years later the regen was obsolete for broadcast work and a welter of different TRF (tuned radio frequency) circuits ruled until the superheterodyne took over in the 1930s.

To most listeners, audio was often an afterthought during the first days of broadcasting. The emphasis was on long-distance reception—or just hearing a station at all! But as the decade progressed more emphasis was placed on fidelity, hence the Acme booklets.

The early audio amplifiers used in receivers were transformer-coupled. Transformer-coupling had a couple of advantages. First, the transformers could be used to give gain. It took at least three resistance-capacitance coupled tubes to equal a two-tube transformer amplifier. Transformer ratios varied between two-to-one and five-to-one.

Second, the circuits function using relatively low voltages. Ninety volts was plenty for most circuits, except perhaps for the output stage, because you didn't have the voltage drop associated with resistance-capacitance coupling. This was important when most radios were powered by batteries. Remember that in 1920 only 35% of American homes had electricity, and by 1930 the figure had risen to only 68%.

For the homes that did have electricity there were "A" and "B" battery elimi-

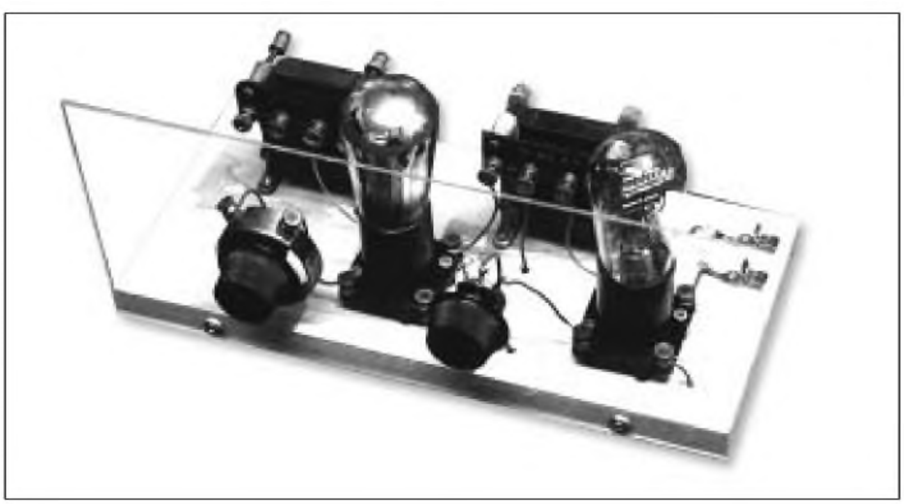


PHOTO 2: A re-creation of a 1920s style audio amplifier that might have been found in most American homes. A 201-A and a 171-A are mounted on a 12 × 5″ board, with a Plexiglas panel. The transformers and most other parts are from the '20s.

nators, which would plug into the wall outlet. However, these were expensive, about $40 for an Acme plate battery eliminator, and about the same for what we would call a trickle charger for the storage battery that lit the filaments. You could buy a lot of dry cells or charge up a storage battery many times for that money. Therefore battery power ruled in many homes.

AMP DESIGN

For the amplifier that's the subject of this article, I chose to use a 201-A voltage amplifier and a 171-A output tube to recreate an amplifier typical of the mid to late '20s. The 201-A was ubiquitous and some radios used them in every stage. When higher output tubes were developed, such as the 171-A in 1926, owners would frequently upgrade their radios by putting one of these tubes in the output stage.

Wiring was easy to get to and it was no problem to change the plate voltage and bias for the output tube without disturbing the rest of the radio. Neither the 1923 nor the 1926 edition of *Amplification Without Distortion* specifies tubes. There were only a few possibilities and builders would consult a chart to determine the correct bias battery for the tube and plate voltage to be used.

I chose a plate voltage of 90 for each tube. As mentioned previously, battery power was more commonly used than AC for most of the '20s, and I suspect that 90V was the value most frequently employed.

I used Acme A-2 transformers, which were in both editions of the booklets. The only other parts in the amplifier are the rheostats and tube sockets. I had one old rheostat and used it along with a newer potentiometer of 25Ω. To hold the tubes I used old Kelford brand sockets, which had an internal spring assembly to reduce microphonics.

Figure 1 shows the circuit. I redrew the original from the 1923 booklet into modern form. The original circuit, *Fig. 2*, operated at 45V, but the reader was cautioned that if higher voltage was used, the bias arrangement from an accompanying three tube circuit should be used.

CIRCUIT FEATURES

The circuit is a little more subtle than it first appears. First, there's a separate rheostat for each tube. If a single rheostat is used in the filament wiring for both tubes, feedback, either positive or negative depending on the phase relationship of input and output, could occur.

Another feature in the arrangement of the filament circuit is that the positive side of the bias battery is returned to the negative side of the A battery, while the positive filament terminal is connected to the negative plate supply. This arrangement was fairly common in the '20s. It adds the voltage drop across the filament rheostat to the negative grid bias.

If the plate voltage on a 201-A was 45V to 67V, no separate bias battery at all would be needed. With 90V a bias battery has to be used but the filament arrangement reduces the size, and also may help save the tube if you make a mistake. Notice also that the filament rheostats are unbypassed by a capacitor. A very small local negative feedback voltage is thus generated which improves the fidelity. Negative feedback wasn't "invented" until 1927, but it was used just the same.

The Acme booklet specified an Acme "Kleerspeaker" for the output. This was something like a high impedance earphone attached to a horn. Since I didn't have one I used a Hammond 1628SE to drive my regular speaker system. A Hammond 125-E also worked well.

The graph in *Fig. 3* shows the frequency response of the amplifier. It looks awful until you realize that the audio broadcast by most radio stations of the day wasn't much better. There were few stations that transmitted any bass and hardly any speakers that would reproduce it.

For comparison, a second curve shows what the response curve looks like with a 68,000Ω resistor across the secondary of each transformer. This shows that by playing around with different values of resistors on the secondary of the two transformers, the audio output could be made considerably flatter, but that's just my modern urge to tweak showing itself.

The instructions in the Acme book said that should the circuit howl or squeak (oscillate) it might be necessary to reverse the connections to one of the transformer primaries. It did and reversing the connections solved the problem.

HOW DOES IT SOUND?

The amplifier sounds gorgeous! It brings the music of the '20s to life (*Photo 3*). I never really appreciated Bix until I heard him on this amplifier; likewise many other '20s artists from Caruso to Jimmie Rodgers. There is something absolutely compelling about the combination.

On the other hand, I put on one of my favorite test CDs, *Sinatra's Sinatra*, arranged and conducted by Nelson Riddle, and it sounds awful!

I have a theory that music of a period sounds best on the equipment of that period. An amplifier using beam tetrodes without feedback, typical of the 1940s, is at its best playing the big band swing of the same period, and often doesn't sound good at all on modern music. This amplifier makes '20s music come alive.

EVERYTHING FITS!

For the record, I did measure the total harmonic distortion and it was very consistent at 1000Hz as being about one and one-half percent at a reasonable listening level with one-tenth of a volt input. The curvy response curve makes IM and individual harmonic

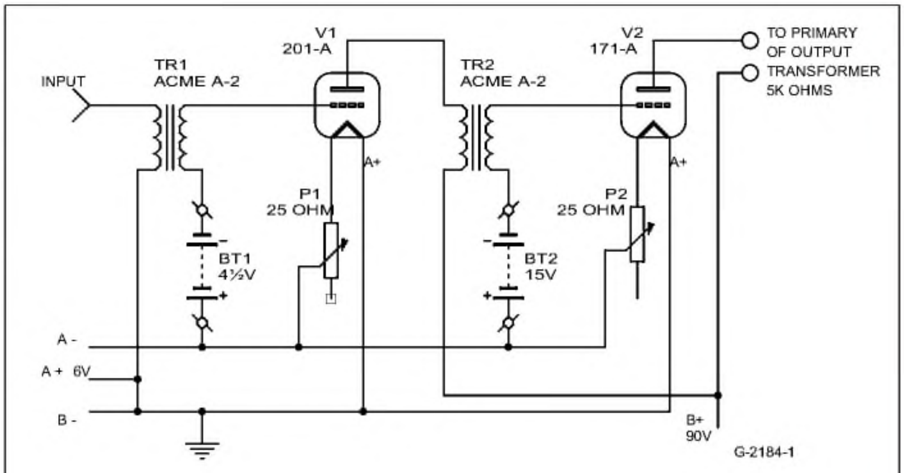


FIGURE 1: This is the circuit of the 1920s style amplifier. It sounds great playing '20s music. Note the filament and bias connections: the positive side of the bias batteries go to A– while A+ is connected to B–. I explain the reasons for this in the text.

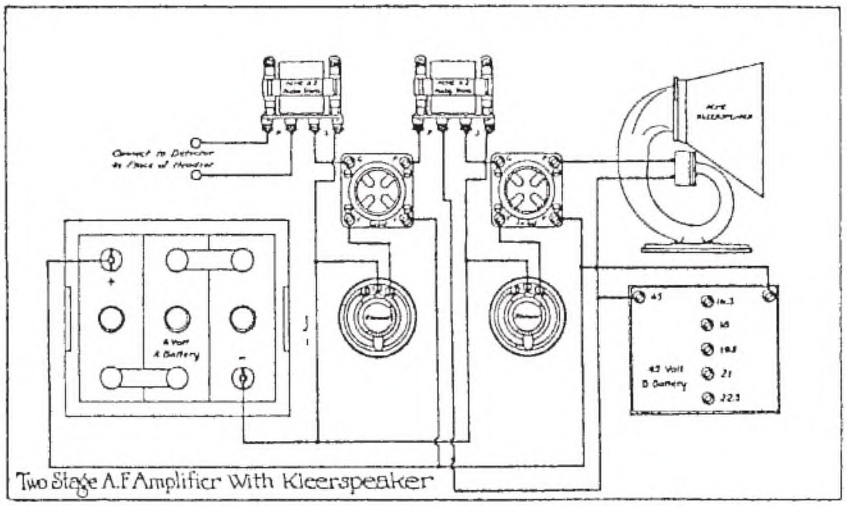


FIGURE 2: This is the original circuit from the Acme booklet. The plate voltage is only 45V so a bias supply had to be added, as explained in the article. G-2184-2

measurements meaningless. Even the THD numbers have to be taken with a grain of salt, though I think in this case they reflect what I hear.

Measurements and this amplifier are an anachronism, though. The people who designed, built, and listened to ones like it in the '20s thought they sounded good, and so do I.

If you would like to build an amplifier like this one, I suggest that there's no need to try and find rare tubes and transformers. Try interstage audio transformers intended for the restoration of antique radios that have a ratio of 3 to 1. You can use inexpensive triode-connected filamentary tetrodes or pentodes for the tubes and modern potentiometers as rheostats. I built my amp on a 12 × 5″ board and used a Plexiglas panel of the same dimensions.

Remember that some modern CDs of '20s music had to be made from whatever sources were available. In some cases these weren't very good and some tracks have distortion built in. Most are clean, though, so before you tear everything apart like I did trying to find out why the amplifier suddenly went sour, try a couple of recordings made after about 1925 first. If you like '20s music and don't have a big collection of original 78s and the equipment to play them, I think you'll like this amplifier! ❖

Editor's Note: A new release of cylinder recordings made in the first two decades of the 20th century is available from Old Colony (PO Box 876, Peterborough, NH 03458, 603-924-9464, Fax 603-924-9467, custserv@audioXpress.com, www.audioXpress.com) with the title Sounds Cylindrical.

PHOTO 3: The amplifier with my selection of '20s music.

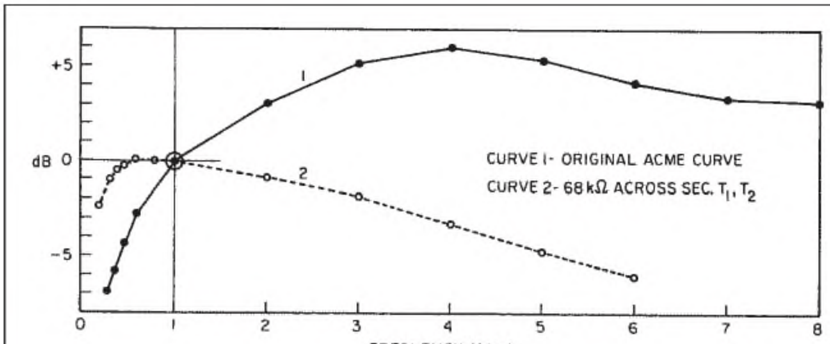
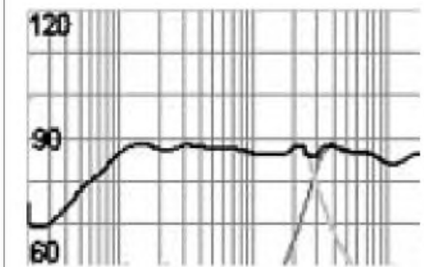


FIGURE 3: The response curve of the amplifier isn't up to today's standards—but the curve fit the music of the '20s just fine.



Product Review

Marchand PM48 Computer Amp

Reviewed by Charles Hansen and Duncan and Nancy MacArthur

Marchand Electronics, Inc., PO Box 473, Webster, NY 14580, (585) 872-0980, FAX (585) 872-1960, www.marchandelec.com. Price: $100. Warranty: 2 years.

PHOTO 1: PM48 card.

The Marchand PM48 is a 12W per channel stereo amplifier card for a personal computer. The card fits into—but does not make any connections to—a standard PCI expansion slot connector. Power is obtained from one of the unused power-supply disk drive connectors, where the 12V DC yellow and black wires are used to power the amplifier card.

INSIDE THE AMPLIFIER

Photo 1 shows the Marchand PM48 amplifier card, and *Photo 2* shows how the connections are made to the PC and to the external audio source and speakers. Audio signals are connected to the PM48 through a 3.5mm stereo phone jack. Two pairs of plastic speaker binding posts on ¾″ centers allow you to connect 4Ω or 8Ω speakers with standard banana plugs. The review amplifier came with a 1m shielded interconnect with 3.5mm stereo phone plugs.

TOPOLOGY

A schematic was not furnished with the unit. The PM48 is based on the Philips TDA8566Q stereo bridge-tied-load (BTL) car radio power amplifier IC with differential inputs. Of course, with the three-conductor 3.5mm stereo jack connection, there is no way to implement fully differential inputs. The two negative differential inputs at the amplifier chip are connected in common through isolation caps and returned to the jack ground sleeve via a resistor.

Just a few discrete parts make up the remainder of the card. The power amp IC is mounted to a 4″ × 3⅞″ × 1/16″ black painted aluminum plate that serves as a heatsink. The plate and the IC are mounted on the PC card. The audio path uses metal film resistors, and what appear to be polypropylene coupling caps. The card uses a mute delay turn-on circuit to avoid speaker pops during power-up. The IC itself has thermal, short-circuit, and reverse polarity protection.

PHOTO 2: PM48 connections.

Inside the PM48, the computer +12V DC input is filtered by a total of 660μF. The TDA8566Q has a rather low supply voltage ripple rejection (SVRR) of 50dB[1], as compared with a power-supply rejection ratio (PSRR) of 120dB[2] for a dual supply power amp IC such as the LM3875. It will be interesting to see how much computer power-supply noise gets through to the speakers (see sidebar, "ATX Computer Supply Noise and Voltage Regulation").

MEASUREMENTS

Making measurements on the PM48 presented a number of challenges. The BTL output stage does not have a common connection to ground, so I needed to work out a ground isolation scheme that allowed me to measure the audio output without shorting one of the speaker connections to ground. Along with that was the concomitant requirement not to introduce excessive hum or noise.

My distortion test set has an input ground lift switch, and I ended up using my laptop for the computer-based ADC-216 DSO because the desktop computer I usually use eventually makes its way to power line ground. My IMD test signal generator has a floating output, but its metering circuit is ground-referenced. In order to monitor results with my analog scope, I made a "phantom" ground for the scope probe with two series resistors across the +12V DC supply.

Making measurement connections

in the confined space inside my PC was also out of the question. For this reason, I used a separate ATX computer switching power supply with the following ratings:

+5V DC at 22A
+12V DC at 8A
-12V DC at 0.5A
-5V DC at 0.5A
+3.3V DC at 14A
+5V SB (standby voltage) at 0.8A

The ATX motherboard connector wants to "see" a motherboard before the power supply will start. This requires jumping the green ATX connector wire to the adjacent black wire (thanks to computer guru Bill Schatzow for solving that little problem).

Before any audio testing, I made some noise and regulation measurements on the supply (see sidebar). I loaded the +5V DC (the red and black wires in the hard drive power plug) to 3A to simulate a modest PC motherboard load without drive activity. This probably represents an ideal case for the audio board. Inside the PC, the PM48 will be subject to much high-frequency EMI and the constantly varying +12V DC supply voltage as the computer performs its various tasks.

I operated the PM48 amplifier at 2W into 8Ω for one hour. The aluminum heatsink (sitting flat on the test bench without benefit of the computer cooling fan) was quite hot to the touch. The amplifier dissipates a minimum of 1.8W at idle, and up to 18W at two-thirds power into 4Ω. You must exercise care in selecting the motherboard slot into which you install the PM48, since surrounding boards could be subject to the high temperature at the heatsink, or block its cooling air flow.

The initial 0.27% THD reading remained the same throughout this run-in period. The distortion was essentially the same for each channel, so I present the bottom output channel here.

TABLE 1
CROSSTALK

FREQUENCY	R TO L	L TO R
100Hz	–52dB	–53dB
1kHz	–52dB	–52dB
10kHz	–51dB	–50dB
20kHz	–48dB	–49dB

By Duncan and Nancy MacArthur

Computer amplifiers such as the Marchand PM48 are aimed at two separate but overlapping markets. The first consists of listeners desiring better or louder sound effects for computer games: we'll call this group Type 1. The second consists of people using their computers for listening to music: we'll call them Type 2. The latter category includes those who like to listen to background music while working on the computer (Type 2a) as well as those who intend to use their computer as a dedicated audio reproduction system (Type 2b).

Given the wide variety of expectations, any single product is unlikely to meet every user's requirements. Most computer systems place little emphasis on sound. Often any additional amplifier and speakers will sound much better than those supplied with the computer. For a user ranging from Type 1 to Type 2a, any separate amplifier, including the Marchand, will provide a more satisfying output than the very inexpensive amplifier already wired into his or her computer.

But the audio quality required for Type 2b use is another matter. When an amplifier is part of a dedicated audio reproduction system, it must be compared with other audio reproduction equipment as well as with live music. Depending on its intended use, a computer amplifier should be judged by two or more completely different standards.

REVIEWING THIS AMPLIFIER

In addition to the multiple-possible-uses issue, computer amplifiers such as the Marchand don't fit easily into traditional audio systems. The amplifier input is through a ⅛″ stereo phone jack, which fits in well with the most common computer output but does not allow use of known RCA cables for evaluation.

Similarly, the outputs of the Marchand are on ¾″ spaced five-way binding posts recessed into the back of the computer. Large speaker cables will not fit into this recess. Bare wires or banana plugs will work.

In fairness, both these connector issues cause problems for Type 2b users (and comparative reviewers) rather than for computer users, who appear to be the Marchand's intended market. For a typical computer user, the ⅛″ phone plug and bare wire speaker connection may be more of a blessing than a concern.

INTRINSIC ISSUES

In general, a very quiet computer is a very noisy audio system. The standards of comparison for each use are wildly different. A computer commonly is used in an office environment where you'd expect a moderate amount of fan noise.

An audio system, on the other hand, usually is located in a quiet living or family room; furthermore, audiophiles often regard even a tiny amount of fan noise with extreme suspicion. Replacing computer fan(s) with quieter versions is certainly possible, but to our ears, unmodified general-purpose computers generate an unacceptable amount of noise for a dedicated audio system. This complaint is not specific to the Marchand, which has no fan, but is a general comment about computer-based audio.

Computers generate abundant RF noise, even outside the shielded case. The Marchand, on the other hand, is located inside the case where the RF environment is much worse. Obviously, any amplifier built into the computer will be in the same high-RF environment, but many audio component designers spend time and money to ensure low levels of RF within their designs. As we see it, while the Marchand is a viable alternative for Type 1 (computer) purposes, its design includes compromises that may be less than optimal for Type 2 (audio) purposes.

Another area of compromise involves the DC power supply. The same switching power supply that supplies the disk drives powers the Marchand amplifier. This power supply was never designed for audio use. In contrast, many amplifier designs place at least as much emphasis on the power-supply design as on the amplifying elements themselves.

A computer accesses its disk drives constantly. With the Marchand this disk access, as well as other computer housekeeping functions, was quite audible. These sounds constantly reminded us that we were listening to a computer "add-on"; their presence distracted us from listening to the music itself. Not only is the power supply not part of the amplifier, but also it's entirely dependent on the host computer. (For this review we used an AOpen computer.)

HUMAN INTERFACE

As mentioned previously, the input connector is quite standard for computer use. A ⅛″ phone to ⅛″ phone patch cord (supplied by Marchand) makes connection to the computer's audio output easy. But, as noted later, although zip cord is widely available, perhaps it isn't the best choice for speaker wire for the Marchand.

Although the Marchand doesn't make any electrical connection to the PCI bus in the host computer (power is supplied through the existing internal disk power cable), the amplifier is intended to be plugged into a bus socket, which provides additional support required by the amplifier and especially by the large heatsink (a flat metal plate). Mounting the Marchand in a non-powered position is possible but requires additional support of the amplifier board. In either case, we recommend removing the amplifier prior to shipping the host computer.

SOUND JUDGMENTS

As with previous listening critiques, we auditioned the Marchand using tracks from the *Hi-Fi News and Record Review Disk III* (track 2: Parry's "Jerusalem," track 4: Vivaldi's trumpet concerto, tracks 5 and 6: excerpts from Prokofiev's "Peter and the Wolf," track 7: Purcell's "Welcome, Welcome Glorious Morn," track 10: a Corkhill percussion piece, and track 14: Rio Napo RSS demo), as well as a wide variety of other music. The PM48's sound was warm and slightly rounded-off. These characteristics were particularly noticeable in the Prokofiev and Corkhill pieces. Its imaging was blurred, and the images tended to "pull" towards the speaker positions on the Prokofiev and RSS tracks. On the Parry and Vivaldi pieces, the PM48 provided little ambience information. Overall, the sound was pleasant but not engaging.

Using 12 GA zip cord for speaker connection exacerbated all these characteristics. With zip cord cables the bass became excessive and the sound warmer and blurrier. We recommend pairing the Marchand with a more detailed (but still inexpensive) cable such as the DH-Labs T-14. We performed these listening tests and the following comparisons with T-14 cable.

We did all our listening using the CD drive of the AOpen computer. The quality of the host computer drive is another variable to consider when evaluating a computer-based amplifier. Since we weren't familiar with the sound of this drive, we compared the performance of the PM48 to several other amplifiers using identical speakers (Genesis 400) and cabling (the supplied patch cord and T-14 speaker cables).

Because we didn't have access to a comparable internally mounted amplifier board, we compared the PM48 with a variety of amplifiers, including a boom box, an Audiosource AMP-1, and (briefly) the Manley Stingray. The Marchand surpassed the boom box's amplifier in every category. Even with its supposedly lower output power, the Marchand was cleaner and more dynamic. It possessed a more extended frequency response.

The AMP-1, on the other hand, bested the Marchand in all categories and provided a more pleasant listening experience. To be fair, the AMP-1 when new cost more than the PM48. We chose the AMP-1 for comparison as being representative of spare amplifiers that you might already own. *(Spousal note: Most audio hobbyists keep an extra amplifier—or two or three or four—stashed away in the garage. Not that I'm bitter, mind you.—NM).* To nobody's surprise, the relatively expensive Manley Stingray sounded better than the AMP-1 and much better than the PM48.

FINAL THOUGHTS

NM: Should you buy the Marchand PM48? Let's ask a second question: What are your circumstances and listening habits? If your computer desk is short on space or you're interested in improving the sound of your computer games, the PM48 will sound better than the cheap output circuit already built into your computer. If you plan to listen seriously to music and you have extra space on your computer desk, you might prefer to experiment with an external amplifier.

DM: If you're looking for the ability to drive speakers directly and reproduce sound effects realistically and are unwilling to add an external amplifier to an already crowded computer table, then the Marchand PM48 is a straightforward answer worth considering. But if having an extra amplifier box is not a concern, an external amplifier—either a used one from the garage or a new one such as the S-5—is probably a better bet.

This would be the left channel using the top-left definition of phone jack polarity. There is no noise at all during start-up or shutdown, and the amplifier brings itself to life after a short time delay.

Then, with your ear to the speaker, you hear a hashy white noise. The output noise (input shorted) measured a high 68mV RMS, with +20mV DC offset. The signal-to-noise (S/N) ratio was 33dB referenced to 1W, 8Ω, 80kHz BW. With A-weighting the S/N ratio increased to 94dB.

The PM48 amplifier does not invert polarity. Input impedance measured 22k5. The gain at 2.83V RMS output into both 4Ω and 8Ω loads was about 23dB. The output impedance at 1kHz was 0.25Ω, decreasing slightly to 0.19Ω at 20kHz. While the power amp IC is rated for 2Ω loads, I limited my testing to 4Ω and 8Ω.

The frequency response (*Fig. 1*) was within ±1dB from 20Hz to 85kHz, at an output of 2.83V RMS into 8Ω. I normalized the response graph at 0dB for 1kHz into 8Ω. There was a slight +0.5dB peak at 39kHz.

When I connected a load of 8Ω paralleled with a 2μF cap, the PM48 appeared to operate without any fuss, but the internal protection disabled the output after about 30 seconds. In order to reset the amplifier, I needed to remove the capacitive load, remove the input signal, and cycle the +12V power.

The IHF load, which simulates a loudspeaker impedance peak at 50Hz, produced a 0.3dB higher response at this frequency than the 8Ω resistive load alone. The PM48 amplifier will be fairly insensitive to variations in speaker impedance with frequency. Crosstalk performance was limited by the 45dB channel separation limit of the TDA8566Q and wasn't helped by the close coupling of the two audio channels within the 3.5mm interconnect (*Table 1*).

THD+N versus frequency is shown in *Fig. 2* for the loads indicated at the right side of the graph. During distor-

TABLE 2
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Frequency response	20Hz–20kHz ±1dB PBW	20Hz–85kHz ±1dB
Gain	24dB	23.1dB 8Ω
Input sensitivity, full output	0.45V RMS typical	0.5V RMS
Total harmonic distortion	0.1%, 1W, 1kHz	0.26%, 1W, 1kHz
Power output (RMS)	12W per channel 4Ω	11.76W 4Ω
IMD-CCIF (19 + 20kHz)	N/S	0.25% CCIF
MIM (9 + 10.05 + 20kHz)		0.09% MIM
Signal to noise ratio	Better than 100dB	94dB "A"-wtg (see text)
Noise	N/S	68mV RMS
Input impedance	10kΩ	22k5
Output load capability	2Ω min (4Ω or 8Ω typical)	
Power requirements	+12V DC	0.155A min to 3.62A max
Output impedance	N/S	0.25Ω 1kHz 0.19Ω 20kHz

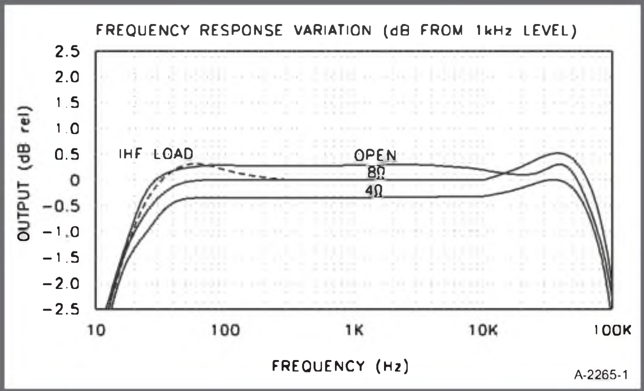


FIGURE 1: Frequency response.

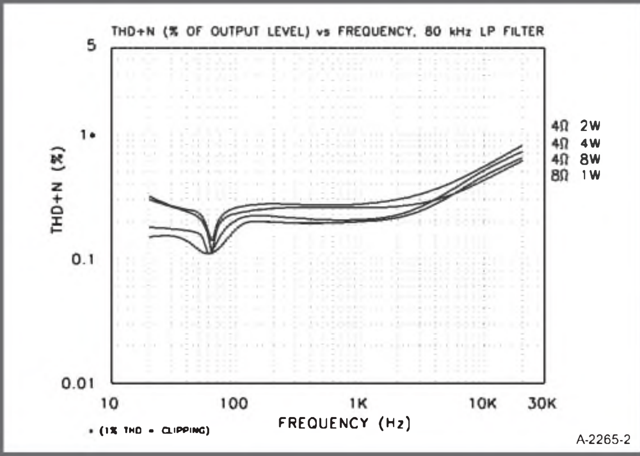


FIGURE 2: THD+N versus frequency.

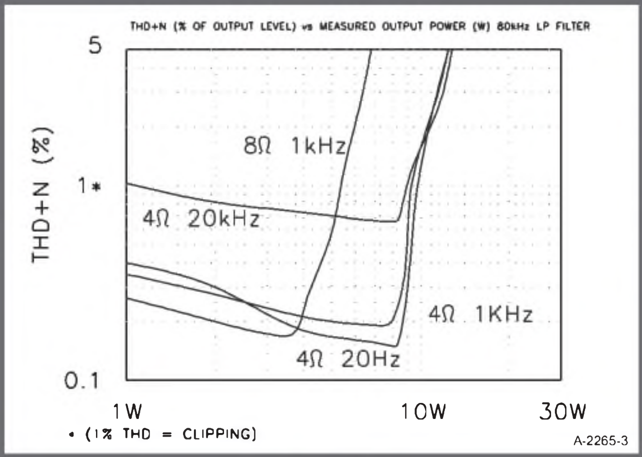


FIGURE 3: THD+N versus output power.

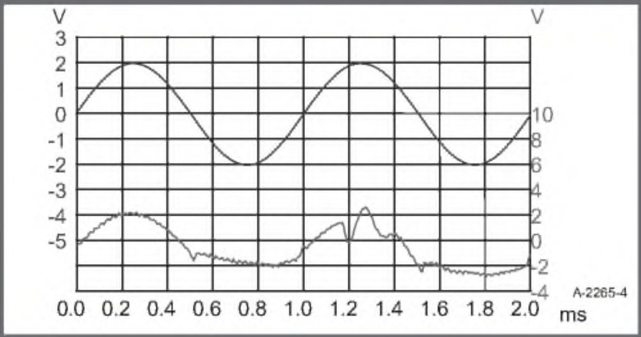


FIGURE 4: Residual distortion.

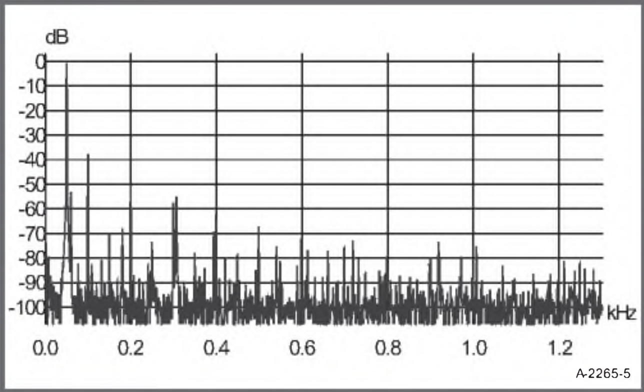


FIGURE 5: Spectrum of 50Hz sine wave.

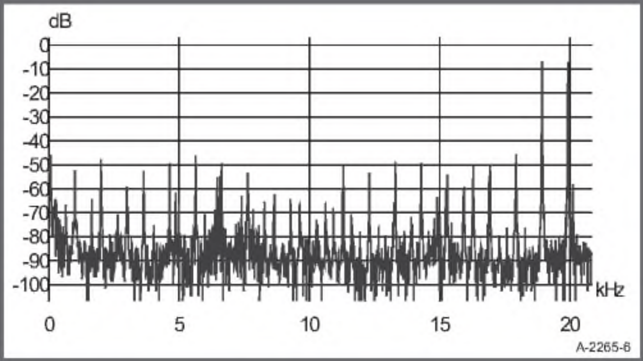


FIGURE 6: Spectrum of 19kHz + 20kHz intermodulation signal.

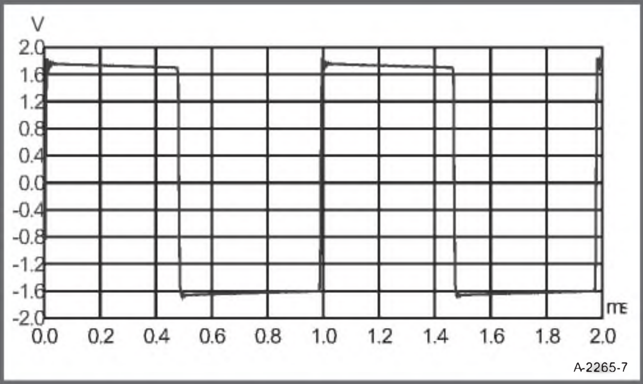


FIGURE 7: 1kHz square-wave response.

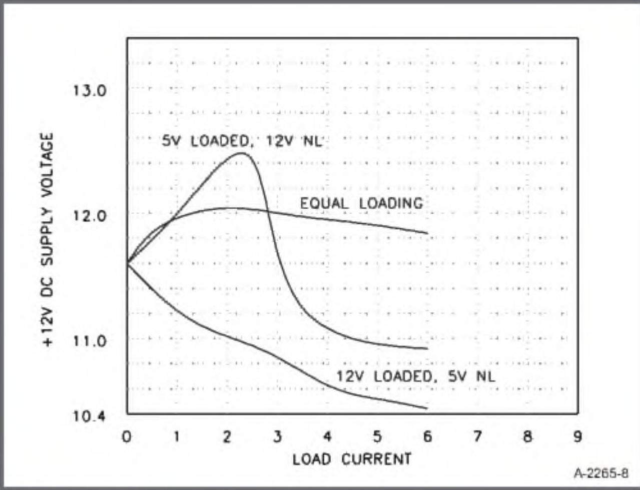


FIGURE 8: ATX power-supply variations.

tion testing, I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. You can see where the distortion test set notch filter has removed some power line hum at 60Hz. Without this hum component, the THD+N would have been lower overall.

Figure 3 shows THD+N versus output power for various loads and frequencies. The PM48 does not exhibit brick-wall clipping, and the sagging +12V DC rail voltage limited its maximum output power. I saw 11.76W driving both channels, and 11.84W with one channel driven, into 4Ω. Assuming its rated power is specified at 1% THD, it fell just short of its 12W per channel 4Ω rating by 0.09dB.

The distortion residual waveform for 1W 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 distortion residual signal shows a fairly high second harmonic due to what appears to be non-symmetry of the BTL

ATX COMPUTER SUPPLY NOISE AND VOLTAGE REGULATION

Most computers use efficient switching power supplies, so audio quality is not one of their design criteria. The ATX-compatible (Rev 2.03) supply I used operates at a switching frequency of about 78kHz. The PM48 taps into the +12V DC supply. This is also used for the hard disk, floppy and CD drives, and the processor cooling fans. At its maximum audio output the PM48 adds about 3.6A to the total 12V DC load.

The +5V DC and +3.3V DC outputs are the critical supply voltages, since the logic has a supply tolerance of only 5% (±0.25V DC on the 5V DC). These two supplies usually have overvoltage protection monitoring. The +12V DC output may or may not be specified for 5% regulation, while the other lower current supplies can vary as much as 10%. *Figure 8* shows the variation in the +12V output for various loading conditions I applied to the two outputs at the hard drive plug. Overall, I found the +12V DC rail varied from +10.4 to +12.5 over the limited 0–6A loading I applied to the two sources, while the +5V rail remained within the ±0.25V logic limit.

The power-supply outputs also carry a fair amount of 78kHz switching supply noise, and may have up to 1% ripple. *Figure 9* shows the output noise from the PM48 with its inputs shorted. The noise is predominantly made up of the 78kHz switching frequency slightly modulated by 60Hz.

This shows one of the problems with using a computer switching supply for audio equipment. The PSRR for op amps and power amplifier ICs is at its maximum at low frequencies and decreases at 20dB per decade of frequency below the PSRR breakpoint. (This is also true of the common-mode rejection ratio, CMRR.)

There is no PSRR curve on the TDA8566Q data sheet, but the data for the better-performing LM38752 shows the +PSRR to be 107dB out to 250Hz, decreasing to 71dB at 78kHz. The –PSRR curve doesn't do as well, with 100dB out to 50Hz and only 40dB at 78kHz. The TDA8566Q has only one supply voltage ripple rejection (SVRR) point specified—at 50dB. I think it's safe to assume this is the low-frequency limit.

Using the same 20dB per decade curve approximation, the TDA8566Q may very well have 0dB SVRR at 78kHz. Thus the quality and type of power-supply filter and bypass capacitors becomes much more critical with switching power supplies used for audio. The ATX supply has one 1000µF filter cap at each of the three high-current outputs (+3.3V,

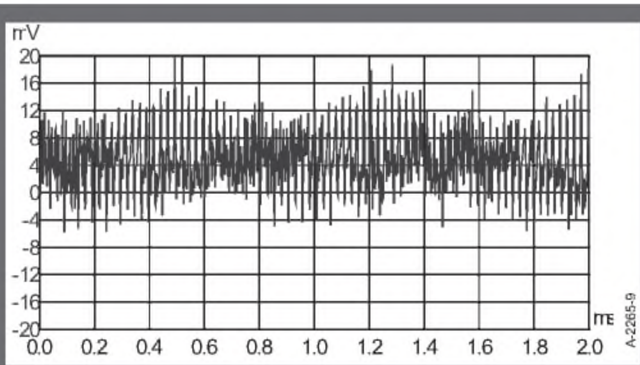


FIGURE 9: Output noise, input shorted.

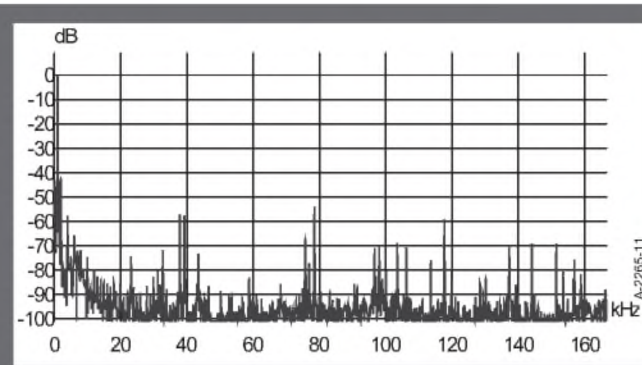


FIGURE 11: Spectrum with ATX switching power supply.

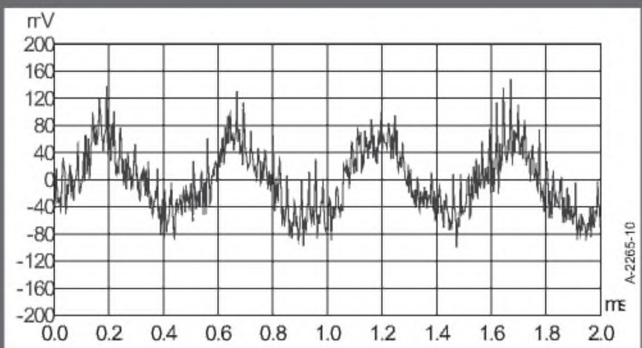


FIGURE 10: ATX 12V power-supply ripple.

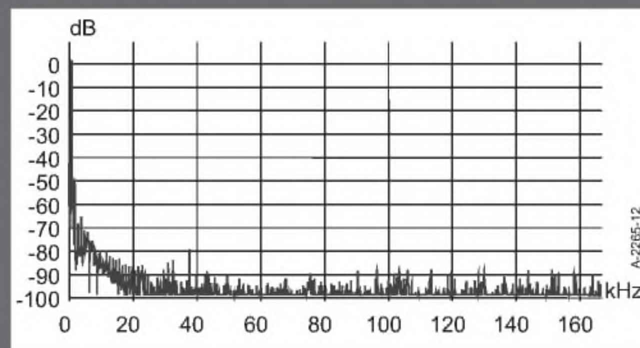


FIGURE 12: Spectrum with regulated linear supply.

output stage, overlaid with switching supply noise. THD+N at this test point is 0.26%.

The spectrum of a 50Hz sine wave at 1W into 8Ω is shown in *Fig. 5*, from zero to 1.3kHz. The THD+N here measures 0.21%. The second, third, fourth, and fifth harmonics measure –39dB, –75dB, –61dB, and –91dB, respectively. Power line artifacts are also present at 60Hz (–54dB) and 180Hz (–68dB). You see the effect of notching out the 60Hz component in *Fig. 2*. Repeating the test with a 1kHz sine wave produced a similar distribution of harmonics.

Figure 6 shows the amplifier output spectrum reproducing a combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 10V p-p into 8Ω. The 1kHz IMD product is a high –52dB (0.25%). Note the even higher amounts of IMD present at approximate multiples of 1kHz throughout the spectrum.

Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, not shown) resulted in a 1kHz product of 0.09%, with the same relative distribution of 1kHz multiples. This test gives a better indication of the PM48's nonlinear response, since it is a closer approximation to music than a sine wave.

The 3.5V p-p square wave into 8Ω at 40Hz (not shown) showed reasonable LF tilt. The leading edge of the 1kHz square wave had just a bit of HF ringing (*Fig. 7*). The 10kHz square wave (also not shown) showed approximately four cycles of the same ringing. This appears to be related to the slight frequency peak at 39kHz. *Table 2* shows my measured performance as compared with the manufacturer's specifications. ❖

+5V, and +12V). The PM48 adds its three 220µF aluminum caps to the +12V input, with one 220nF film bypass cap.

At no load the 78kHz power-supply ripple measured 30mV p-p on the +12V DC output (0.25%, or –52dB). With my 3A preload on the +5V DC supply and an output of 2W into 8Ω at 1kHz, it increased to 200mV p-p (*Fig. 10*). The noise is modulated at twice the 1kHz output frequency as it changes the current demand on the +12V supply at a rate of 2kHz.

I expanded the scale of my spectrum analyzer to 166kHz to show the full effect of the power-supply noise. *Figure 11* shows the output of the PM48 at 1W into 8Ω at 1kHz with the ATX switching power supply. This spectrum of power-supply noise helps explain the big difference between the broadband and A-weighted S/N ratios.

Figure 12 shows the same test condition using a well-filtered 14.4V DC regulated linear supply I use for car audio testing. The THD+N at 1W into 8Ω at 1kHz has dropped to 0.21% from 0.26%. This is not as dramatic a reduction as the comparison between *Figs. 11* and *12* would lead you to expect. The reason is that the band-limiting 80kHz LP filter in the distortion test set has removed most of the HF power-supply hash from the audio output.

However, given the mediocre intermodulation performance (*Fig. 6*), you can assume the 78kHz ripple may interact with audio signals and cause non-harmonic in-band artifacts to appear. Unfortunately, a slight peak in the amplifier frequency response occurs at half the 78kHz power-supply switching frequency, so some of the intermodulation products may be due to an excitation of this response peak. This is only speculation on my part, but note the peak at 38kHz in *Fig. 12*.

The performance of the TDA8566Q is fairly typical of car audio amplifier ICs, and, while not up to high-end standards, it is better than many of the AC-adapter powered computer speaker amplifiers that come standard with most personal computers. The high noise environment in a PC is a real challenge for critical audio applications. Keep that in mind when selecting a sound card to archive your LPs to digital, since the A-D conversion will take place inside your PC, most likely using the ATX supply as a power source.

REFERENCES

1. Philips TDA8566Q Data Sheet, 21 Feb. 2001.
2. National Semiconductor LM3875 Data Sheet, June 1993.

Xpress Mail

OVERLOOKED TUBE

Perhaps I can shed a little light on Mr. Markwalter's question about the 8417's lack of success ("Vintage Glass," Feb. '03 *aX*, p. 72). While the tube does have some very desirable specifications, audio OEMs didn't like the tube because of apparent design flaws, particularly the GEs. The screen grids had a bad habit of failing, and the tubes were also prone to becoming gassy. The two problems exacerbated each other.

While others, such as the Sylvanias, had a better reputation, they also had a higher than acceptable rate of failure than most power tubes. Unfortunately, this failure mode usually resulted in the tube's catastrophic failure and damage to the amplifier's circuitry. Granted the bulk of 8417 tubes ran comparatively well, but the failure rate and resulting damage was too high for OEMs to like the tube, which never made it into the big time like other power tubes.

I believe part of the problem was that OEMs were always playing the game of one-upmanship in the power output department. Output tubes were often run at or very near maximum specs. Some tube types handled it quite well; the 8417 failed miserably when run hard. They are not commonly found today.

Depending on how hard Mr. Markwalter is driving his tubes, he may have fairly good luck with them. The triode connection could be less prone to the failure mode.

Edwin G. Pettis
Pettis Engineering
Grand Junction, Colo.

J.L. Markwalter responds:

I was drawn to the type 8417 power tube primarily because of its outstanding characteristics when used as a triode; i.e., relatively high mu and low plate resistance. The combined plate and screen dissipation rating (40W) puts it in the league of the 300B and a few others. I am unaware of the 8417 being failure prone for the reasons Mr. Pettis gives. I used the tube successfully in an amplifier I designed a number of years ago and operated it for more than 20 years in my home system, now retired. The tubes I used were made by Westinghouse.

My amplifier operated the tubes in Class AB1 biased to 40mA/tube, 400V plate supply, thus 16W dissipation per tube when idling. I do not believe it is good design practice to operate tubes at their limits if you expect reliability. A good design should not cause the failure of other components after a catastrophic tube failure. I will say the envelope temperature of the 8417 seemed rather high even so.

If, as you say, the tube has design flaws, a larger envelope, among other things, might have been in order. I think its electrical characteristics are good enough to warrant fur-

ther improvement if indicated. However, I no longer design using vacuum tubes, but pass along my experience with the 8417 to those who might be interested.

Congratulations on using the 8417 in a Class AB1, push-pull triode output configuration. You are probably the only designer who has used this tube in an output stage, connected as a triode, and you are correct that it is very sensitive and easy to drive. The tube is still available from Antique Electronics Supply for $53.

Your information pertaining to its operating characteristic as a Class AB1 Triode power amplifier should be useful for readers: 400V DC plate supply, 25W output, 5000 plate-to-plate load or 30W output, 3000 plate-to-plate load.

Congratulations on your amplifier design, using this very sensitive power output tube. I must confess, I prepared the data sheet for this tube while working at Sylvania. I do not know the name of the tube engineer or the marketing agent, or I would readily supply this information. You, Mr. Markwalter, are to be complimented for seeing the potential of operating this tube as a Class AB1 Triode output stage.

Again, thank you for your design efforts and for bringing this interesting piece of information to our attention.

Joseph N. Still
Bel Air, Md.

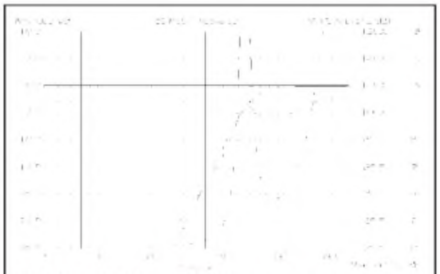


FIGURE 1: Simulated low-frequency response (using FE-164) using Boxplot software. B-2224-1

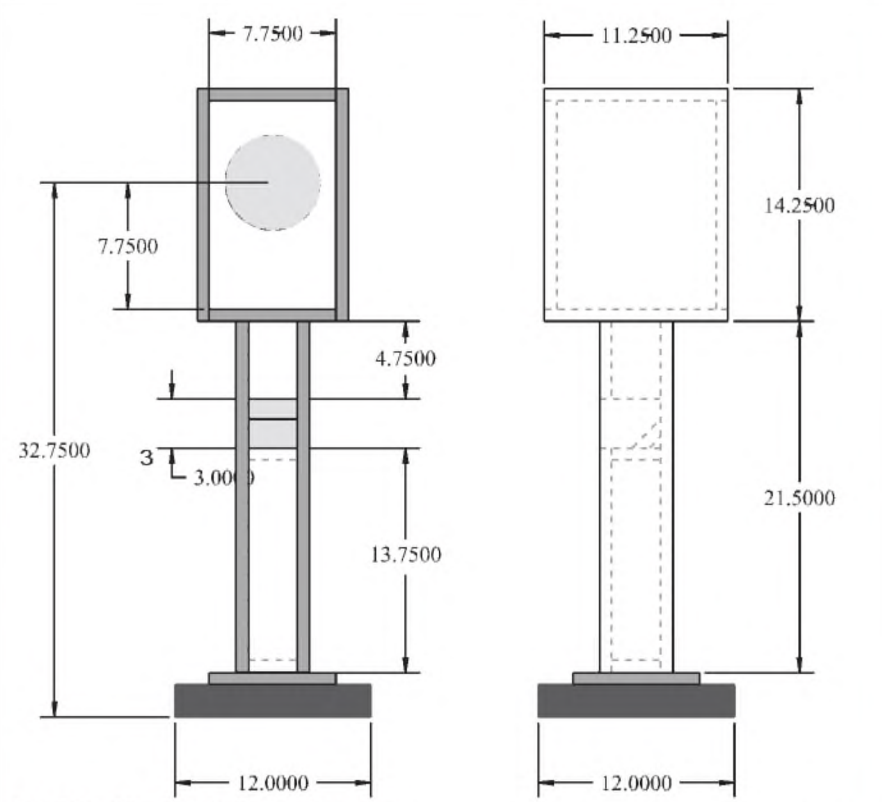


FIGURE 2: The FE-164/FE-167E speaker. B-2224-2

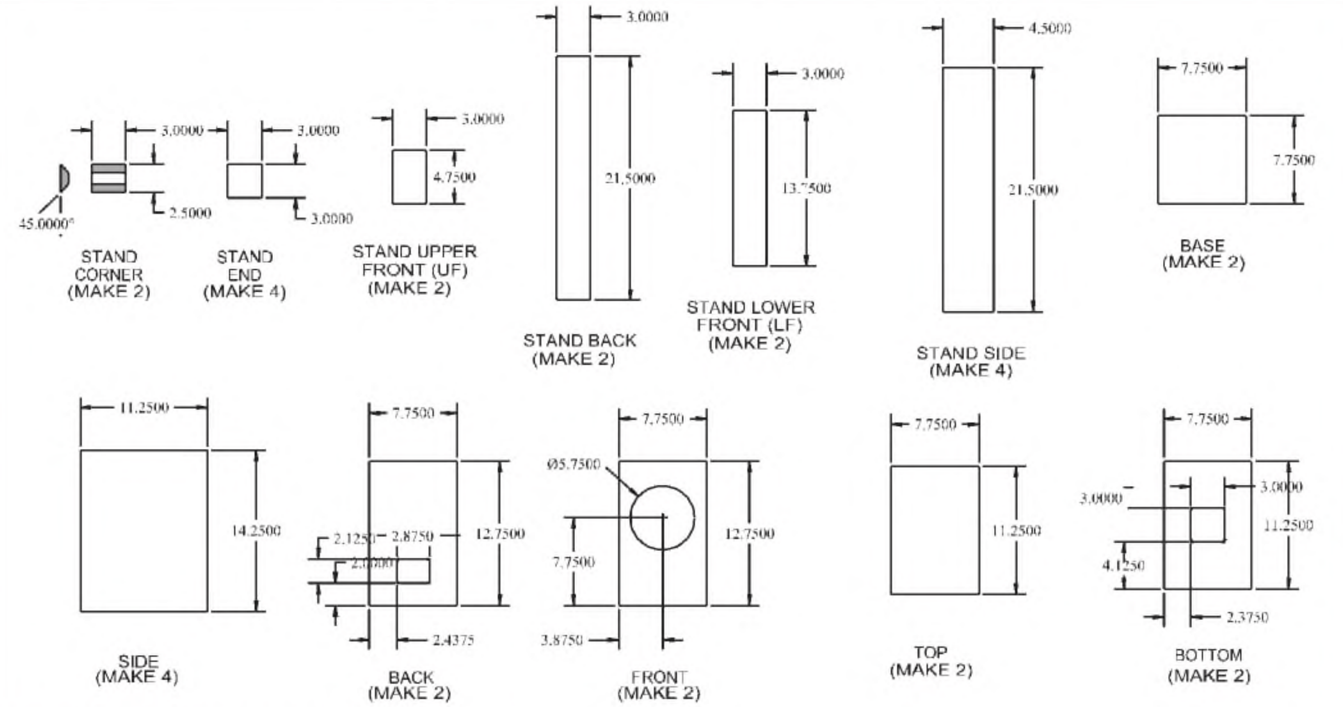


FIGURE 3: Individual parts used to make the speaker. B-2224-3

J.L. Markwalter responds:

I appreciate Mr. Still's comments. With continuing interest among audio hobbyists, including among equipment designers, in vacuum tube power amplifiers, I would think the 8417 would be widely chosen, especially as a triode. Such is not the case. I find rather amazing the current interest in some "ancient" triode tube types having low amplification factors and directly heated (filament) cathodes—neither desirable in themselves—although they usually have low plate resistance, which is desirable.

I used the 8417 as a triode in an amplifier built in the 1960s, which remained in service for more than 20 years. I used a single-ended push-pull output configuration similar to that shown in the Audio Amateur Glass Audio Projects*, first edition, page 39.*

Since the output transformer must have a split primary, the choice is somewhat limited. I used an UTC LS-55, which was popular at the time, having both line and voice-coil secondary windings. The UTC LS-57 is OK, too, having only a voice-coil secondary.

While these transformers were rated at 20W, the 8417 as a triode would provide an honest 20W with them. I suggest using the 5000Ω plate-to-plate primary connection. I bet a number of these transformers are still around today, but not in use.

CORRECTION

Due to a printing error, we are republishing the figures (*Figs. 1–4*) accompanying Peter Millet's full-range speaker project that appeared in last month's issue (p. 6). We regret any inconvenience the poor quality of the reproduction may have caused. – Eds.

CAPACITOR VIOLENCE FOR THE UNINTIMIDATED

Here's how I identified and modified the capacitors in my upgraded consumer-quality video equipment, consisting of a TV set and DVD player[1]. This represents a knockdown, drag-out approach to upgrade, which will appeal only to the most hard-core videophiles, as the changes are very intrusive and may cause your local repair shop to refuse to work on your equipment. Such was the case for me, so I am now an experienced DVD repairman/technician. In a way, that isn't all bad.

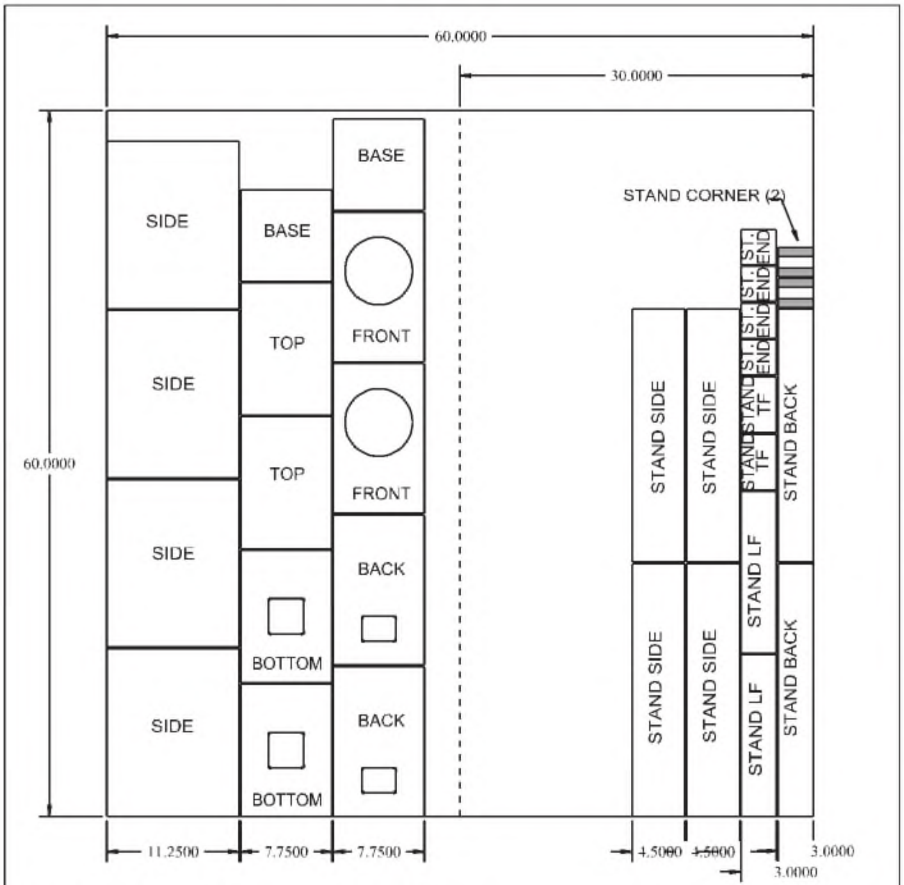


FIGURE 4: Cutting diagram from a 5′ × 5′ sheet.

B-2224-4

Schematics

Scan the schematics thoroughly for aluminum electrolytic and ceramic SMD (surface mount) chip capacitors. I shade the electrolytics red and the ceramics blue to make them easy to find on the page later. The DVD schematics are especially dense, and require a magnifier to make the component symbols legible.

Having a little familiarity with how to interpret the schematics, I attempt to omit those capacitors which serve areas of the circuit which already work fine, such as DVD transport drive circuitry, the parts within a TV tuner, and so on. It is useful to learn the identity and function of the integrated circuits to guide this effort.

Hit List

Next I make up a list of all marked capacitors. I sort them numerically either by part number or by pin number when associated with a large video-processing IC. For digital equipment I have two lists—one for electrolytic, one for ceramic caps.

Each line in the list contains the pin number (where appropriate), part number, value, accuracy, voltage rating, type, function in the circuit (here's where the real work comes in), and modification. I check off each mod after I execute it.

Hits

Both TV and DVD signals occur in the megahertz spectrum, so the use of aluminum electrolytics makes mincemeat of the signals. I remove the electrolytic capacitors and replace them with equal-value tantalum caps, which are also polarized. I bypass these with film and ceramic caps. Fortunately, the PC boards have either through-holes for leaded electrolytics or generous solder pads (for SMD electrolytics), to which feet bent onto the leads are soldered.

Electrolytics are often bypassed with SMD ceramics, but not very successfully, for the following reason: The best ceramic caps are made from the NP0/COG chemistry, which yields the lowest

DIFFRACTION SPREADING LOSS (DSL)

A Follow-up to "Simple Satellites, Part 2" (June '03 *aX*, p. 44).

One of the problems with DSL is that it is called by a variety of names. "Diffraction rise," "diffraction step," "diffraction loss," and "spreading loss" all refer to the property we call DSL, a condition in which a finite freestanding enclosure is omnidirectional at low frequency, but at high frequency converts to radiation restricted to the side containing the drivers. Thus, out in front of an enclosure employing one perfectly flat response driver, the on-axis response at high frequency will, in theory, be 6dB above that at low frequency. This rise in on-axis response with rising frequency is the "diffraction step," and its cause is the property we call DSL.

Surely, most readers have experimented with the placement of large enclosures. Sliding them across the floor to a position closer to the rear wall can greatly increase the upper bass region response. However, the effect on imaging or other characteristics may be adverse.

Part of what allows placement to vary the bass response of a large enclosure is the DSL effect. When you move the enclosure near the rear wall, partial suppression of the DSL increases the upper bass region response. Keep in mind that with small enclosures the effect of DSL can extend all the way up to near 1kHz. You are not playing with just the upper bass region with placement of such small boxes, but with the midrange region, which greatly affects the presence and quality of the sound.

Improper placement of small satellites used with a subwoofer can yield a system with a "hollow" sound. It has the bass along with the treble, but the critical range from maybe 100Hz to 1kHz is suppressed. If you have *Speaker Builder* issues that go back to 1987, check out Ralph Gonzalez's excellent discussion on the DSL topic.[1]

This work indicates that small satellite boxes out in the room on stands require the full theoretical 6dB of DSL compensation. For small boxes on the floor, but out from the rear wall, about 3–4dB of DSL compensation is probably sufficient, but up to 6dB is acceptable. Tall, narrow enclosures with small woofers mounted high off the floor should be considered as "stand mounted" boxes.

How can you handle the DSL problem? Rather than just ignoring it, you have the following options:

1. *Avoid DSL.* You can accomplish this by using the satellite drivers in a very large panel or very large enclosure. This approach surely inhibits building small satellites. Mounting a small driver in a wall avoids DSL, but may cause other problems. Small enclosures used near the floor/rear-wall interface will also limit DSL, again with other possible

distortion. There are many other chemistries which yield far higher capacitance for a given volume, but they make far more distortion. A quick read of the Digi-Key and Mouser catalogs points up the fact that NP0/COG SMD caps aren't offered at much over 300pF, so when your schematic shows a value of 0.010µF, you can bet it isn't NP0.

You can bypass the ceramic SMD caps with NP0 chips of identical physical size, by tacking right on top of them. This isn't easy (they are tiny), but it can be done. My method is to hold the piggyback cap in place with tweezers and tack one end using a sharp, conical soldering iron tip and the solder tinning already on the end caps and iron. With one end tacked, I use solder on the other end, then on the first end. Sometimes I even add a small-value film cap here, too, but this makes the assembly vulnerable to snagging and damage to the foil traces. Magnifying glass inspection of the joints then follows.

In a very few instances SMD ceramic capacitors are parts of filter circuits, so you should replace them with equal-value NP0 whenever possible, to preserve the frequency response of the circuit. Often I remove the SMD caps and replace them with leaded ceramics. If I damage the foil traces, the leads of the new capacitors are long enough to reach other solder pads on the same nodes.

Payoff

Is all this worthwhile? Yes. These circuit components are the reasons the megabuck video demonstrations look so good. Perseverance will pay off. Having made my last mods to my DVD player, the Superbit version of Fifth Element now really does look sharper, clearer, snappier, and more pure than the original letterbox version.

Darcy E. Staggs
Orange, Calif.

REFERENCE

1. "A DVD Rescue," July 2002 *audioXpress*, p. 24.

consequences. A simple way to apply this is to place small satellites in a bookcase or in a wall-mounted bookshelf. In both cases you should surround them with books to form a large front panel.

2. *Compensate for DSL.* You can compensate for DSL in a variety of ways:
   a. Electrically correct the response via equalization. You can do this at line level with a network between the preamp and amplifier, possibly part of an active crossover, and also at speaker level with networks included with the passive crossover. Boxes #1–#4 with the modified Infinity drivers use equalization built into the woofer crossover.
   b. Use a dual voice coil (VC) driver with one VC driven full range and the second VC used just for DSL compensation. Focal and probably others manufacture drivers just for this purpose.
   c. Use multiple drivers in the satellite with some drivers dedicated to producing DSL compensation. This is the approach used in boxes #9 and #10.
   d. For larger enclosures that handle the bass region, you can get some DSL compensation via the woofer alignment used. The IB is great for this, because you can produce a peaked response while maintaining a well-damped system. About 3–4dB of DSL compensation is available with the IB[2].
   e. You can build a closed box with a peaked response by raising the total system QTC; however, most listeners find the sound of such under-damped systems unacceptable. With the vented box you can get a peaked response by using the BB4 alignment. Some listeners find this approach acceptable; others do not like the sound of this alignment. Remember that with drivers handling the bass region the use of DSL compensation will likely increase the cone displacement requirement.
3. *Out-fox DSL by careful driver selection or crossover design.* Sometimes you can find a driver with a response that droops with rising frequency through the region where DSL will affect the response. By properly sizing the enclosure, you can use this slope to help compensate for DSL.

With a two-way satellite it is sometimes possible to select the enclosure size and crossover frequency to permit DSL compensation. In such cases, you compensate for DSL by shaping the crossover responses and adjusting the relative driver levels. A clever application of this is the THOR system[3], which takes advantage of the 6dB sensitivity rise of the dual woofers used in that system. — **G.R. Koonce and R.O. Wright, Jr.**

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1. Gonzalez, Ralph, "Balancing Small Speakers," *SB* 3/87, p. 44.
2. Koonce, G.R., and Wright, Jr., R.O., "The Infinite Box Range Concept," Jan. and Feb. '03 *aX*.
3. D'Appolito, Joseph, "THOR: A D'Appolito Transmission Line," *aX*, May 2002, p. 8.

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AD INDEX

Book Review

Opamps for Everyone

Reviewed by Jan Didden

Ron Mancini, Editor in Chief. Texas Instruments Design Reference, Publication number SLOD006A. US $39 or free download from www.ti.com.

Opamps have been with us for over 60 years, and they have presented problems for circuit designers since day one. You see, the feedback that makes them accurate and versatile also has a tendency to make them unstable.

All IC manufacturers produce publications describing their products and applications on how to use them. After all, if you don't know a company's products or how to use them, they wouldn't sell a lot of them. Some of these publications come real close to design reference manuals. TI's *Opamps for Everyone*, however, is one of a kind—four hundred pages packed with design guidance in 18 chapters plus annexes and references on every conceivable use of opamps.

CONTENTS

The book begins with a review of circuit theory, going into ideal opamp circuits, followed by single supply opamp techniques. Then it becomes serious: feedback and stability, non-ideal opamp properties, and several chapters on voltage feedback and current feedback opamps, their differences and compensations. A very good chapter on opamp noise follows. A chapter on understanding opamp parameters closes the information section.

The following chapters focus on specific application areas such as sensors, interfacing opamps to DACs and ADCs, wireless applications, oscillators, and active filters. Another gem: Circuit board layout techniques discusses issues such as mechanical construction, grounding, passive components, decoupling, and a host of other very valuable tips. The last chapter describes low voltage circuits.

What makes this book unique for me is that it crams into a single volume many widely different topics that are all related to the use of opamps. Although it is not for the absolute beginner, anybody with just a bit of experience in the use of opamps (or discrete amps, for that matter) will appreciate this book.

Look at the sidebar on noise. Yes, there is math in it, but it is so clear and straightforward that you will have no trouble understanding it. And the information is directly applicable to the (audio) use of opamps. To be honest, there is some more involved math, but you will certainly understand the material, if not the math, and gain valuable insights.

This leads to the question: for whom is this book? As Mancini says in his "forward" (is this a typo or intentional?), there should be something in it for anyone interested in practical electronics, while it should not be too boring for the practicing engineer. Practical examples use TI opamp types—which is hardly a surprise—without plugging TI products.

CONCLUSION

For me, this book neatly fits between Walt Jung's *Opamp Cookbook* and Jirí Dostál's *Operational Amplifiers*. The first one is ideal if you choose to use opamps and need guidance on the application, while the latter goes very much into detail and theory with lots of math. TI's book gives you ready-to-use circuits, but also tells you why the circuit is the way it is, and how to roll your own for your specific requirements.

The breadth of the book is also its weakness, if you could call it that. I find myself referring to a couple of chapters again and again, while hardly reviewing others. I suspect that every reader will find his favorite chapters in the book.

The price is very reasonable, and you can even get it free as a PDF download. (Well, free is relative. Printing over 400 pages and getting them bound takes an effort and isn't really worth the savings.) Old Colony would do well to make the book available for its aficionados. ❖

CHAPTER 10, PARA 10.2.5, NOISE UNITS:

Example: A TLE2027 opamp with a noise spec of 2.5nV/RtHz is used over an audio frequency range of 20Hz to 20kHz with a gain of 40dB (100 times). The output voltage is 1V (0dBV). To begin with, calculate the RtHz part: Rt (20.000–20) = 141.35. Multiply by the noise spec gives 2.5 × 141.35 = 353.38nV, which is the equivalent input noise. Since the gain was 100, the equivalent output noise is 35.3µV. The signal to noise ratio can now be calculated: 20 log (1V/35.3µV) = 20 log (28329) = 89dB....