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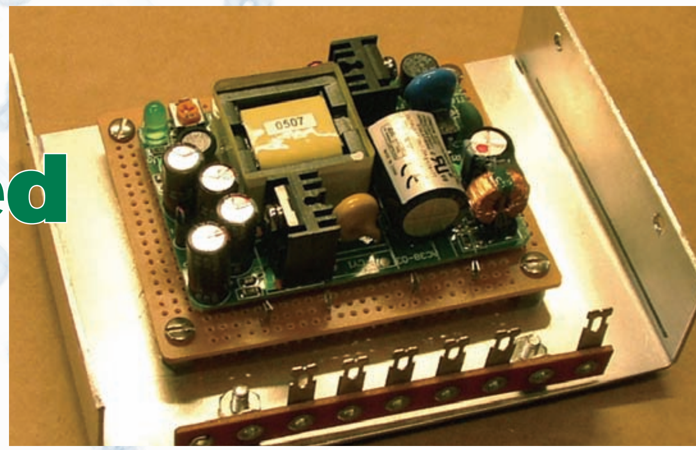

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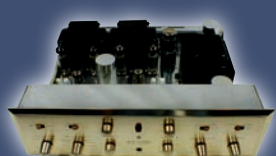

> Parts Upgrade for the CLASSIC SCOTT AMP

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The Hideaway TL Sub

A Danish blueprint for an "invisible" subwoofer.

By Bjorn Johannesen

PHOTO 1: Completed unit, with the line painted black to make it "disappear."

Inspired by Rick Schultz articles, I made a successful transmission line to replace a vented sub. However, I was surprised that the resonance frequency was lower than I had expected—lower than the driver's Fs. I had the impression this was "forbidden," but I could see it was working very well and much better than the same driver in a vented box. Therefore, I started to study Martin J. King's quarterwave design. My own experience and the useful information and supportive help from Martin led to this article.

DISAPPOINTING SUBWOOFER

Some years ago, I made two subwoofers designed by my local speaker shop. The enclosure was a bass reflex with 81 ltr internal volume. The driver was a 10″ Beyma 10B60 (Fs 31Hz, Qts 0.36, Vas 115 ltr, and Sd 380cm^2). You can find all the data for this discontinued driver on www.thielesmall.com.

I was not happy with the result: nothing below 30Hz, port noise, and problems with uneven frequency response due to room resonances. Furthermore, these cabinets are not easy to hide away. Actually, they were hidden away. Far away.

TRANSMISSION LINE SUB

Twenty years ago, I built horns like everybody else at that time. The horns were quarterwave designs with expanding pipe and a driver offset down one-third of the line. Quarterwave designs are not outdated at all, as I found out by reading articles by Rick Schultz. I wanted to design and build a transmission line (TL), which is the term I use for all types of quarterwave design. Because I had two bass reflex speakers, I would be able to compare the TL sub against the vented enclosure.

My idea was to mount the driver in a small box and let the pipe itself be smaller, so I could hide it under the couch. Of course, this would reduce the volume of the cabinet, and you could expect a weak bass. But because both the opening and the driver are close to the floor, I hoped to get a lift in the deep bass. With the opening at the front, you could even place the sub in a corner to get maximum room gain. I ended up with this design, using a small opening, to be able to place the cabinet just about anywhere (**Fig. 1**).

The box itself should not be very visible. The size of the rest of the cabinet was dictated by the space under my couch. I made the opening small, set to about 20% of the cross section (and thereby 10% of the derived Sd).

The total length of the pipe is 230cm, and I assumed, the pipe was tuned to about 37Hz. But I soon found out this was not the case.

I used relatively thin material and included a cross bracket in the box. The front baffle is 22mm MDF. The stuffing is relatively light in the box and for the first line of the pipe only.

FIGURE 1: Simple and practical sub design.

THE RESULT

Photos 1 and **2** show the working unit. In comparison with the vented box, the Hideaway bass is lower, clearer, and

PHOTO 2: The visible part is only 32 × 32 × 32cm.

tighter. There is no noise caused by turbulent air in the opening.

The cone motion is reasonable down to 20Hz, but at 16Hz the movement increases. There is still output at 16Hz, even though the output drops from 20 to 16Hz. This surprised me. It was clear that the tuning frequency of the pipe was lower than I had expected and lower than the driver Fs. I wanted to find out what was happening.

TUNING FREQUENCY

I started to study the work of Martin J. King, who has been very helpful and patient with my questions (*see sidebar*). I could see that the tuning frequency was about 25Hz, and by the use of Martin's worksheets, which can be used for private and non-commercial purposes, I learned what was going on.

If the line had been a straight line, the tuning would be around 37Hz. Actually, the Hideaway enclosure behaves like a mass-loaded TL with a port, which lowers the tuning frequency of the cabinet.

The plot from Martin's worksheets, Acoustic impedance of the cabinet, showed the tuning frequency was about 25Hz (**Fig. 2**). The first peak is at 25Hz, which is the tuning frequency of the cabinet alone.

SOUND PRESSURE LEVEL (SPL)

The dotted line in **Fig. 3** is the driver in an infinite baffle (IB). The SPL might look poor, but because the driver and the opening are at the floor, there is a

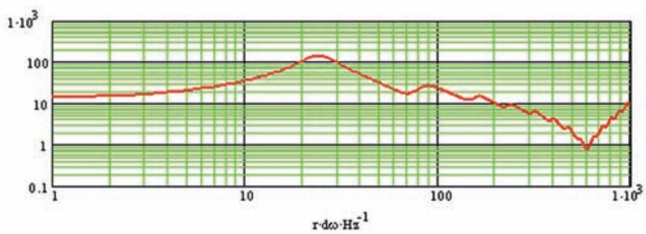


FIGURE 2: Cabinet tuning is 25Hz.

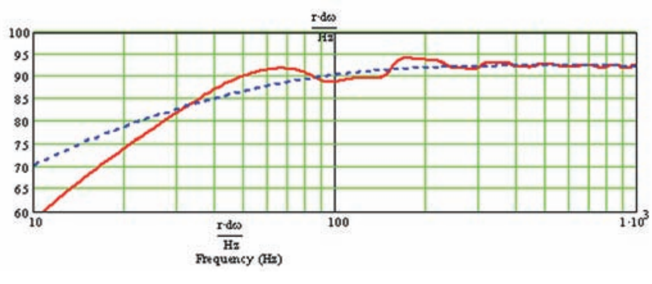


FIGURE 3: Room gain makes it sound better than it looks.

good deep bass performance in the listening room. Because this is a sub, I do not worry about the uneven frequency response above 100Hz, which will be eliminated by the crossover filter.

In **Fig. 4**, the solid line is the driver, and the dotted line the opening. As you can see, the opening is adding at lift to the bass where the driver is rolling off. The output from the aperture is broad and flat. Also notice the fall output from the opening, with a rolloff starting at about 85Hz. The relatively small volume and the mass loading by the small opening causes this unique behavior of the output from the opening.

PHOTO 3: The cabinet is relatively thin, but there are little vibrations.

IMPEDANCE

In the **Fig. 5** plot, the enclosure is not stuffed, to make the impedance peaks more visible. With stuffing the impedance curve becomes very flat, and some of the peaks hard to see.

The dotted line is the driver in an IB. The solid line is the combined enclosure and driver, forming the new resonance system. The first resonance (impedance peak) just below 15Hz is a combination of the air mass in the pipe combining with the moving mass of the driver cone and voice coil oscillating on the driver's suspension. Very much the same as the old trick of lowering the driver Fs by adding some physical mass to the cone.

The second resonance peak about 60Hz is the original cabinet resonance, but with the influence of the driver. The rolloff from the opening is about 85Hz, which corresponds to the dip in the impedance curve between the second peak at about 60Hz and the third peak at about 98Hz.

A GENERIC DESIGN

I have modeled several other drivers in the Hideaway enclosure, and while this versatile design does not necessarily get the very best of each individual driver, the drivers I have simulated are all doing fine. The tuning frequency of 25Hz is a reasonable value for drivers with a Fs of 35Hz and below. A driver with the same Fs, but higher Qts and lower Vas has a little improvement in the deep bass, due to the later rolloff.

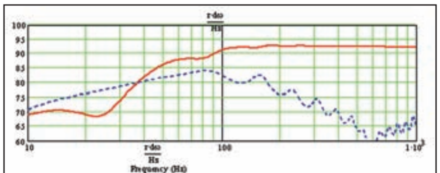


FIGURE 4: The contribution from the opening is shown by the dotted line.

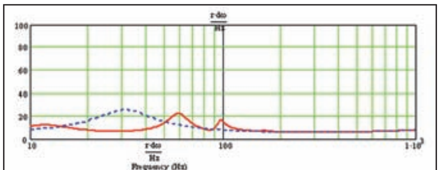


FIGURE 5: Impedance shown with no stuffing.

The Peerless 830452, with the very low Qts of 0.17, is also doing well. In this case the cabinet tuning is higher than the driver Fs of 18.9Hz, which is a good combination for a low Qts driver, which has an early rolloff.

CONSTRUCTION

Figure 6 gives the unit's external dimensions. In addition, the line length is 230cm, internal dimensions of the box are 29 × 29 × 29cm, and cross section in the pipe is 12.8 cm (width) × 15 (height). The footprint is 30 × 140cm using 10mm particleboard. Front baffle is 22mm MDF, which then extends the footprint in this end.

The box uses 10mm particleboard, with 10mm MDF added inside to reduce vibration. The box also has a cross bracket inside. The "labyrinth" walls measure 150 × 14mm. I added triangular pieces of wood in all corners. The labyrinth is closed on the top with 10mm particleboard (**Photo 3**).

The box measures 32 × 32 × 32cm externally, and by placing a piece of glass on the top, the box becomes a nice little table. **Photo 4** gives you an idea of the cabinet size. Stuffing in the box and the first half of the labyrinth is 200g of polyester wool.

DESIGN CONSIDERATIONS

Designing a TL is a challenge, because you have many parameters to work with. But this is also the reason TL design is great fun. I use a design idea as a starting point; I do not choose a driver, and try to find the best design possible. Trying to achieve maximum performance of a driver most likely would result in a cabinet size and shape you probably would not like to place in your listening room. Designing a TL is a recursive process, with the goal of finding the best compromise.

TUNING FREQUENCY

The most important design parameter is defining the tuning frequency of the cabinet. If you increase the tuning frequency, the output level from the opening will increase. And when you lower the tuning frequency, the output level will, not surprisingly, decrease.

The output from the opening is a desired property of the TL, adding a lift to bass. If your driver has a high Qts, you

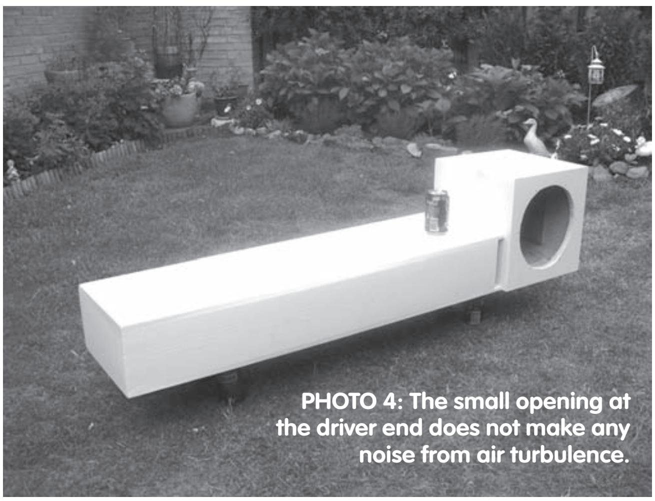


PHOTO 4: The small opening at the driver end does not make any noise from air turbulence.

could set the tuning frequency 5-10Hz below driver Fs. Low Qts drivers have an earlier rolloff; therefore, you could set the tuning frequency higher than driver Fs.

I use a Qts of 0.35 to distinguish between high and low Qts. This is not the "only truth," and your definition might be different. Therefore, the tuning frequency is below the Fs of my Beyma driver.

GEOMETRY

Once you have a tuning frequency in mind, you must decide which geometry to use. Length and geometry sets the tuning frequency of the cabinet. Increasing the volume will increase the bass. Increasing volume also influences the tuning frequency a little.

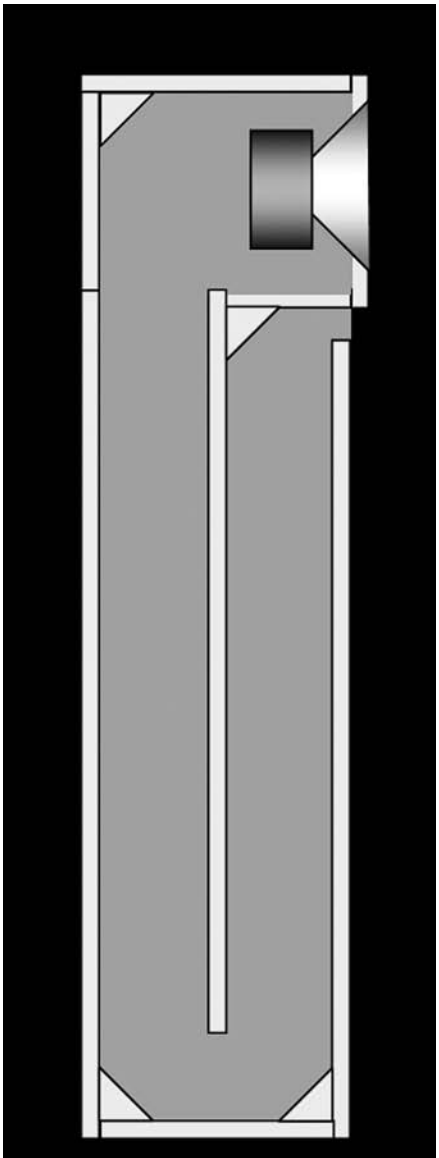


FIGURE 6: Footprint is 140 × 30cm, with 2cm added by the front baffle.

The Hideaway would have had a tuning frequency of 37Hz if it were a straight pipe. But the reduced volume in the pipe and the reduced opening result in a lower frequency. You could call it a ported, mass loaded, tapered quarter-wave. But the label on the design is not important to me.

Tuning frequency is set by the length of the line and the geometry. For a straight line, it is simply a quarterwave of the frequency. A tapered line (opening has a smaller area than the closed end of the pipe) is the smallest possible enclosure for the same frequency. A tapered line is used to tame unwanted upper harmonics. The output from the opening is broader and with slightly less output, compared to the straight line.

An expanding line is the longest line for the same tuning frequency. It has a high output level; unfortunately, the

MARTIN J. KING'S MATHCAD WORKSHEETS

From Martin J. King's website (www.quarter-wave.com), you can download the free version of MathCad 8 and Martin's worksheets, available for your private, non-commercial use. I always use the "Sections" worksheet, and you can model the Hideaway with your favorite driver. The geometry definition I use is somewhat simplified. I have not defined the corner in the pipe as a separate section, but this is not important at low frequencies.

The plots in this article use the following input, and it is very useful to benchmark different drivers. Type in the Thiele/Small parameters (TSP) for your driver, including BL. You can use one of the worksheets to convert from Vas to BL, if you do not know the BL value.

I recommend you measure your driver's Thiele/Small parameters. You should not expect that your driver's parameters will necessarily match the ones documented by the manufacturer. If you do not have the ability to measure, and you do not know the TSP for the driver, try this site managed by Jan Fredriksson: www.thielesmall.com.

Note: Martin uses "d" (driver) instead of "s" (speaker), so Vad = Vas, and so on. I find Martin's terminology more logical than the original TSP abbreviations. Set n_closed := 0 and n_open:= 3 in the worksheet. Then define the geometry.

Driver to closed end:

Length	Cross Section Start	Cross Section End	Stuffing
5.5″	2.21 Sd	2.21 Sd	Y

Driver to open end:

Length	Cross Section Start	Cross Section End	Stuffing
5.5″	2.21 Sd	2.21 Sd	Y
39.5″	0.51 Sd	0.51 Sd	Y
39.5″	0.51 Sd	0.51 Sd	N
0.6″	0.11 Sd	0.11 Sd	N

The last section is the opening, with the length = dimension of the wood used. Total line length is 90.6″. Stuffing is 0.3 lb per ft^3. The pipe is divided into two sections to distribute the stuffing. I used 200g of polyester wool in the box and the first half of the line.

Cross section is defined by Sd because it is easier to type in and also gives me a good feeling of the size of the cabinet. In my example, Sd = 380cm^2. You can adjust the input to your drivers. The pipe tuning is 25Hz. You can check your input in the first plot in MathCad: In the acoustic impedance the first, and highest, peak is at 25Hz.

Internal geometry of the cross section is:

2.21 Sd	29cm × 29cm	11.417″ × 11.417″
0.51 Sd	12.8cm × 15cm	5.039″ × 5.906″
0.11 Sd	2.8cm × 15cm	1.102″ × 5.906″

frequency curve is very uneven. If you mass-load the straight line—by making the area of the opening smaller than the cross section—the line will be shorter than the straight line for the same tuning frequency. Volume will, in general, increase the bass and also lower the tuning frequency a bit.

PROBLEMS WITH TL

The resonance of the pipe is what you want. The other, higher resonances cause dips and peaks in the SPL and are definitely unwanted. A common way to tame the upper harmonics is to use a driver offset down one-third of the pipe. This was not an option in the Hideaway design.

Stuffing is also used to damp the uneven SPL and to smooth the impedance curve. On the other hand, I do not want to attenuate the opening. Therefore, I use light damping in the box and only half of the line. In a sub the crossover will get rid of the ripples, anyway.

Please note that damping has very little to do with the acoustic impedance (tuning frequency) of the cabinet. In fact, your local altitude above sea level and the temperature of the air has more impact on the speed of sound, and thereby the resulting tuning frequency, than stuffing.

BEYMA IN BASS REFLEX

I had the opportunity to compare the Hideaway with a vented design with the same driver. The bass reflex has a higher SPL, but output dropped from 35Hz to 30Hz, and below 30Hz there was no sound. The cone excursion increased under 35Hz, and this construction should have had a highpass filter. Compared to this bass reflex, the cone motion of the Hideaway TL is very well damped.

This sound pressure level might look impressive (**Fig. 7**), but the frequency is very elevated due to room gain, and this "boomy" sound is not nice at all. There is no deep bass, and the rolloff is like falling off a cliff. Furthermore, this construction suffered from port noise.

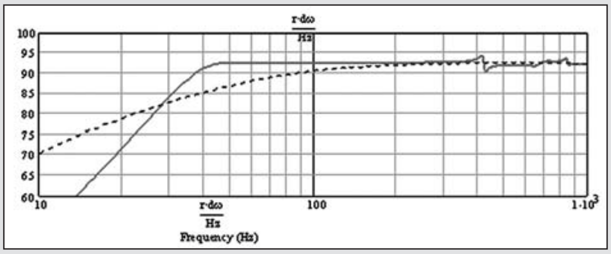


FIGURE 7: The old bass reflex had problems with elevated response caused by room gain.

CONCLUSION

The Hideaway TL has some very useful properties:

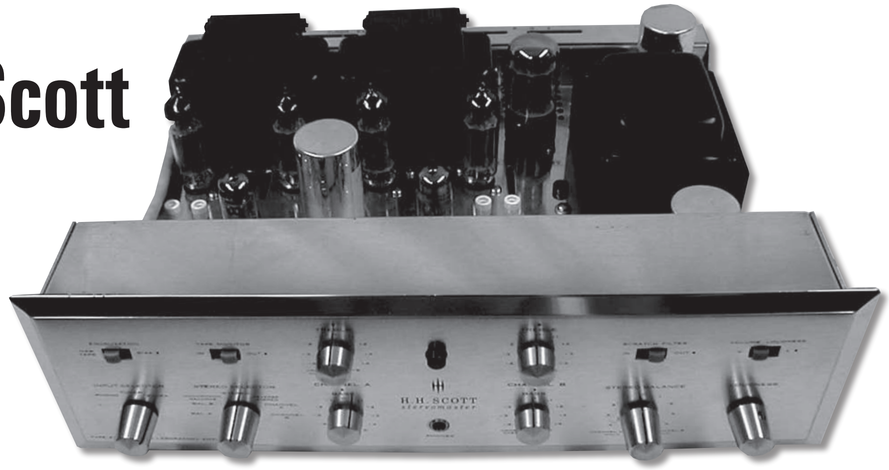
- It is very practical, and does not take up much space.
- The location of the opening allows for the pipe to be hidden.
- Allows for placement against the wall or in a corner.
- With the opening and the driver close to the floor, the deep bass SPL is very good for an enclosure of this size.
- Can be used with a variety of drivers with Fs at 35Hz and below.
- Very good damping of the cone excursion. aX

Bjorn Johannesen is an avid DIY hobbyist involved in building amplifiers and loudspeakers. His speaker designs have included horns, full-range, and transmission lines. The interest in quarterwave designs has accelerated in the last few years, thanks to audioXpress, www.t-linespeakers.org, *and* www.quarter-wave.com. *Bjorn is living in Denmark, just outside Copenhagen. His e-mail address is* bjohannesen@post.cybercity.dk.

ACKNOWLEDGMENT

I would like to thank Martin J. King for the help in understanding his quarterwave design (*see sidebar*).

Renovating the Scott 222C, Part 3



By Charles Hansen

My replacement parts list, keyed to the schematics in **Figs. 33–36**, is shown in **Table 4.** Note that I changed the component designations from those used by Scott. They assigned left channel parts to digits 1-99 (example: C1), then the identical right channel parts by adding 100 (C101). The parts common to both channels had 200 series numbers. Mullard also used this system in their stereo amplifier designs.

I tried to work with this for a while, but found it totally (for me) counterintuitive. So I numbered my schematics 1-99 for the common parts, 100-199 for the left A channel, and 200-299 for the right B channel. Because there are no mechanical assembly drawings in the Operating Manual or lettered component numbers other than the tube ID label on the chassis bottom (which was going to need to be removed anyway), this was not a problem for me.

Parts cost for this renovation ran about $270, plus the cost of the original amplifier. The cost of a new Scott 222C in 1961 was $165 (if it appears this renovation was not exactly cost effective, think of the cost of a new 222C as $1031.25 in 2004 dollars).

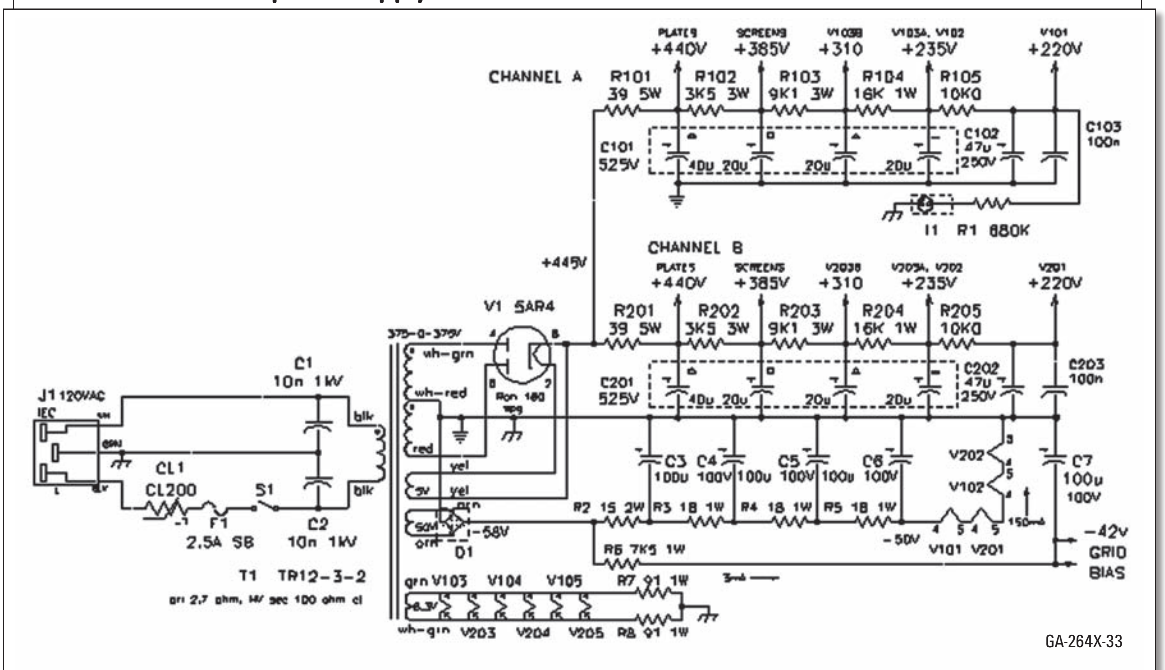
Scott engineers did a nice job in making mostly star connections for the grounds and power supplies. Decoupling caps are used from circuit ground to the chassis at the rear panel phono jacks and speaker terminals to help limit noise. Twisted pairs are used for AC and signal wiring, and the phono signals are shielded from the rear jacks to the front panel. The circuit ground connection to

TABLE 5: SUPPLY VOLTAGES FOR 7189 OUTPUT TUBE AMPLIFIERS.

Model	Power transformer	Power transformer HV winding	Vplate	Vscreen	Vphase inv	Vamp	Vpreamp	Vphono
222C	TR12-3-2	365-0-365	+420	+345*	+335	+300	+285	+220
222D	TR12-3-4	360-0-360	+410	+398	+315	+238	+238	+205
299A	TR12-3-1	365-0-365	+365	+295*	+280	+280	+230	+175
299B	TR12-3-1	350-0-350	+430	+360*	+320	+320	+225	+170

*a power resistor divider is used to derive the screen voltages in these three models.

FIGURE 33: Modified power supply.



the chassis is located at the phono preamp tubes.

The next order of business was to improve the power supply. There were four models of the Scott integrated amplifiers that used 7189 output tubes: the 222C, 222D, 299A, and 299B. The 222A and 222B used 6BQ5/EL84 tubes, while the later 299C and 299D used 7591 octal output tubes (schematics for the 299 series are also on the H. H. Scott website). Scott used four different power supply circuits with the four 7189 tube amplifiers. The differences are listed in **Table 5** (this is the DC voltage listed on the schematics at the filter caps, not the voltages measured at the tube sockets).

It appears that Scott was constantly revising the power supply for what was essentially the same amplifier design. Only the 299A appears to have kept the 7189 plate and screen voltages within the tube manual maximum limits. The 299B has even higher voltages than the 222C and 222D, although some sources list the 299B output tubes as 7189A. The Sams PhotoFact schematic for the 299B clearly calls for the 7189.

In the end I decided to design a fully dual-mono supply except for the 5AR4 rectifier tube (**Fig. 33**). Tube circuits have rather poor power-supply rejection unless you use constant-current sources or balanced differential stages, which was rare in the H.H. Scott era. I removed the old high voltage can capacitors (C201-C206, also including low voltage bypass caps C14 and C114), replacing them with a pair of modern CE Manufacturing 40µ/20µ/20µ/20µ twist-lock replacement cans with a higher 525V DC voltage rating. I had already replaced the selenium bridge with a 6A 200V silicon bridge rectifier during the initial power-up.

Because the 4 × 75µ/75V bias supply can cap was also worn out, I replaced it with four 100µ/100V discrete high-temperature (105° C) aluminum caps. I also used high-temperature aluminum caps for C105 and C205, the phono stage supply filters. I retained the stock neon power indicator lamp (the 222D and later amplifiers used a 6.3V filament lamp).

In light of the high inrush current (**Fig. 2**), I added a CL200 inrush current limiter, which is a negative temperature coefficient (NTC) thermistor whose resistance is initially high but decreases as its temperature rises due to the current flow through it.

The CL200 is rated for 25Ω (cold) and 1.7 Arms. At 5W/channel output this device dissipates almost 1.2W, so it requires good ventilation to extract all the heat it produces. I positioned it directly under the large hole in the chassis vacated by tubular resistor R212.

The 222C output tube screen supply used 1K2 10W series resistor R213 from the HV filter caps to the four screen grids, and shunt 4K 7W tubular resistor R212 to ground. These two resistors dissipate almost 20W by themselves. The tubular shunt resistor needed to be mounted above the chassis. The 299A and 299B also used shunt resistors for the screen supply, although with higher resistances that didn't waste as many watts. The screen supply for the 222D is more energy efficient, using a higher resistance series-dropping resistor between the plate and screen filter caps with no shunt resistor, and 1K series resistors to each pair of screen grids.

With my dual mono supply, I am using individual screen resistors for each of the four screen grids. I suspect the high voltage winding on the 222D transformer actually had less than 360-0-360V AC. Without R212 the plate voltage in my modified supply rose to +457V DC, while the 222D schematic lists it as +410.

One side of the 6.3V AC filament winding was grounded in the original 222C. I used two 91Ω resistors, R7 and R8, with their center-point grounded for improved hum rejection in the power amplifier section. This also restored the heater supply to 6.3V with the higher 120V line voltage. The Scott amplifiers were designed back when the nominal AC line voltage was 117V AC. I also selected R2 so the heater voltage for the four series 12AX7s was the correct value (50.4V DC) at 2W/channel output and 120V AC line voltage.

I decided to completely change the phono preamp circuit (**Fig. 34**) because it was not very accurate with respect to the specified RIAA equalization curve. A SPICE simulation of the original circuit showed this to be the case, even with the original specified values. I removed the 100K resistors in series with the MAG HI inputs and used these two jack positions for a fourth input selector position, this one directly to the top of the power amplifier loudness control through a coupling capacitor. I used a new Electroswitch D4C0406N four-pole six-position rotary switch for this purpose. The added position is clockwise from the Extra input and is unmarked on my faceplate.

I removed the RIAA/NAB EQ modules from the Equalization switch and abandoned the slide switch in place. Ditto for the Scratch Filter switch. The Tape Monitor switch uses the original slide switch.

The modified phono stage uses standard high voltage polypropylene and ceramic NP0 caps. I needed to maintain flat response at low frequencies because of the steep rolloff of the preamp input high-pass filter. I settled on a higher +220V DC for the phono stage after playing around with the SPICE circuit simulation program to ensure I had sufficient headroom using this value of B+ voltage and adequate power supply isolation from the rest of the amplifier.

Incidentally, after measuring the actual no-load current draw for all the various stages, SPICE proved to be a quick way to determine the optimum series step-down resistors for the whole power-supply design. The unknown is the ripple current in the high voltage filter caps, and SPICE accounts for this quite nicely. It also showed that the ESR and ESL of those older electrolytic capacitors must have been horribly high if the original schematic voltages were correct.

FIGURE 34: Modified phono stage.

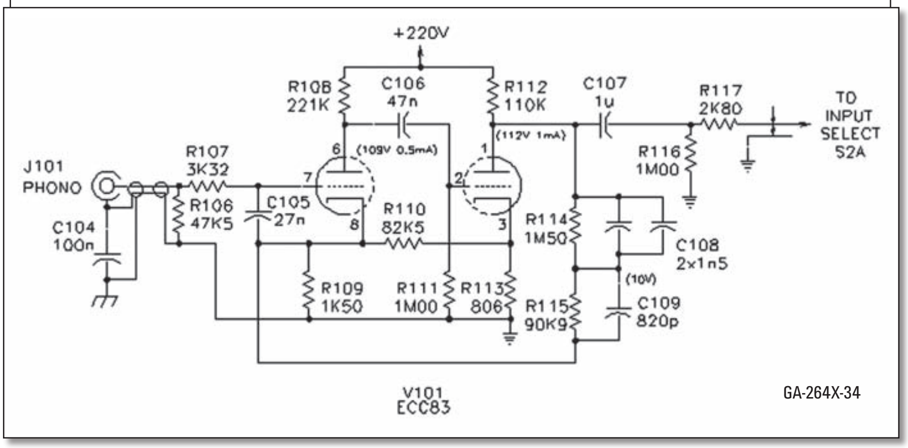
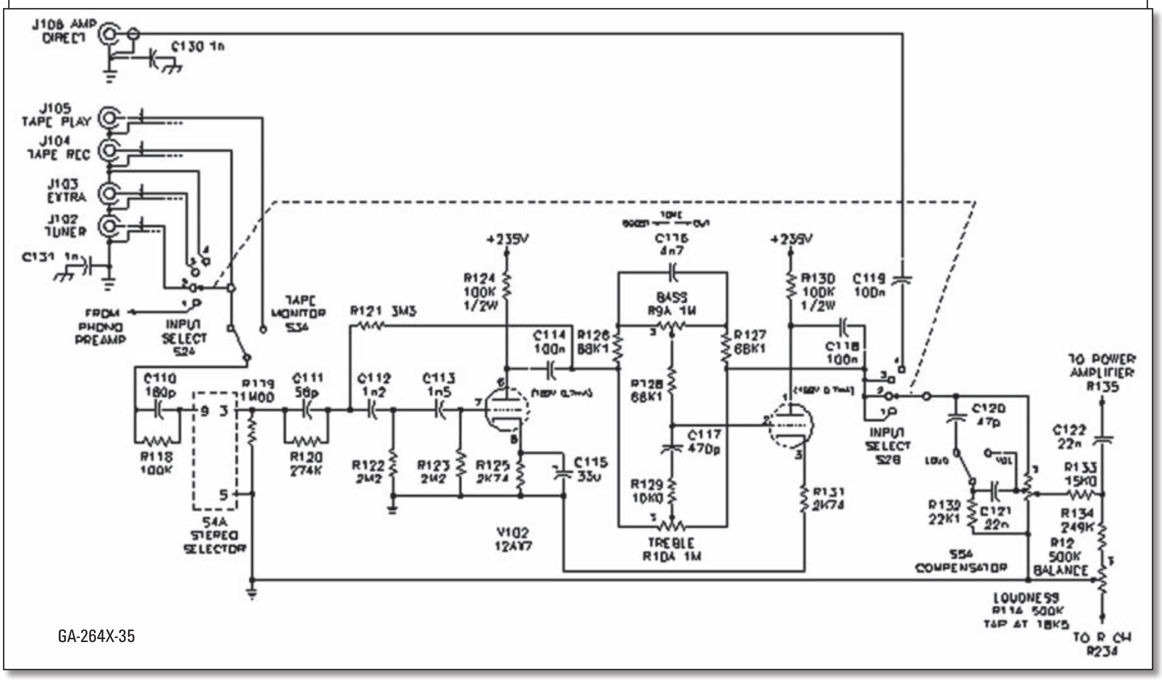


FIGURE 35: Modified preamp and tone controls.



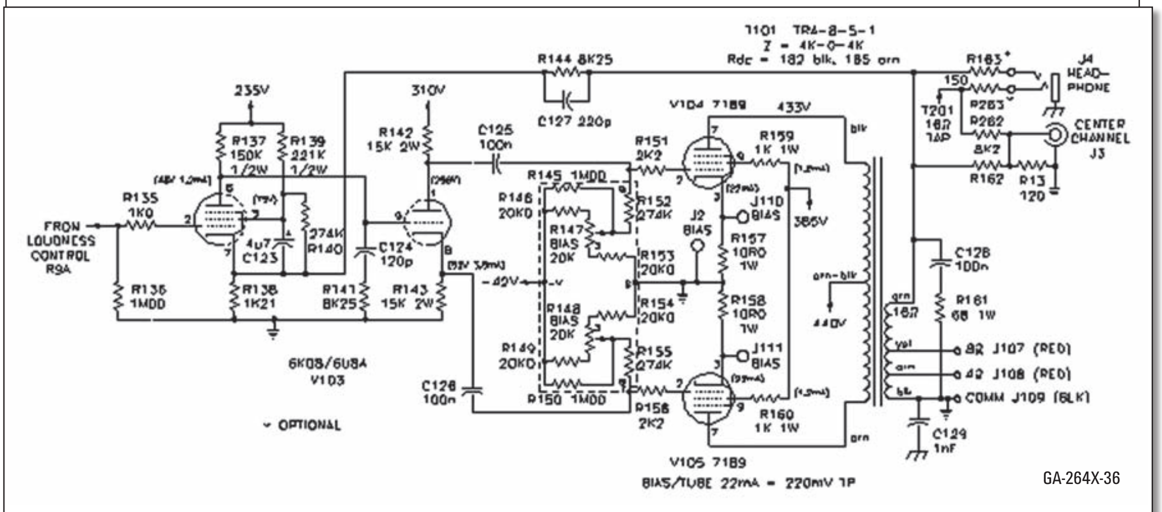
The preamp/tone control circuit (**Fig. 35**) is derived from the Scott 222D and the 299D integrated amplifiers. It may appear that you can remove quite a few of the components near the 222C Scratch Filter, but this is not the case. You can combine C9 and C10 into a single value (I used 56pF), but C8, R14, C9, C10, R15, C11, R16, R17, C12, and C13, along with V2-A, form the sharp cutoff (-12dB) HP filter. The 222C Service Bulletin in my Operating Manual says it "is designed to prevent overload of the output stage and the loudspeaker due to subsonic rumble frequencies and record eccentricities."

Because the filter remains engaged during Tuner or Extra input operation, it may also be designed to prevent saturation of the output transformers at low frequencies, although my tests of the output transformers make this less likely. A test of the modified unit at the direct input to the amplifier, which bypasses the HP filter, will determine whether this is true. I decided to incorporate the input filter for now. It is easier to remove parts later than try to make room for additional parts after all the wiring is finished.

The tapped tone control pots were intermittent and very noisy and needed replacing, so I redesigned the circuit to increase the capacitor values and decrease the resistances. I reduced the Balance control to 500k and added two 249k resistors to each leg to reduce its authority. The Stereo Selector switch allows the selection of either A or B input mono, so having that capability in the balance control was redundant. This allowed me to reduce the value of R133 (old R31) to 15k for less noise.

The power amplifier section is largely unchanged (**Fig. 36**). The part values around V103 are from the 222D.

FIGURE 36: Modified power amplifier.



I added 10Ω cathode resistors to each output tube, which now has its own individual 20k bias adjust pot, and tip jacks to easily measure the output tube bias current. Because the idle current for each output tube is 22mA, each 10Ω resistor dissipates only about 5mW. I used carbon composition resistors for the grid stoppers for their higher surge ratings in case of a grid failure. There is little current through them, so noise is not really an issue.

I did not reconnect the headphone jack, only left it in place to fill the hole in the front panel. If you want a functioning headphone jack, you can use the original 222C circuit, or the simpler one I show on the schematic that uses R163 and R263 to supply the signals from the 16Ω taps. In the latter case, you will need to disconnect your speakers and substitute load resistors when you want to use headphones. I did retain the center channel output for use with my subwoofer amp (the 222D also has a center channel speaker-level output).

Because I have the capability of matching transconductance (Gm) and plate resistance (Rp) at any given value of bias current using an Audiomatica Sofia tube curve tracer, I did not include an AC bias adjust pot in my modified 222C. The Sofia is no longer in production, but Jim Hagerman makes a fine tube curve tracer, the VacuTrace[8].

The output stage and output transformer are mainly responsible for the distortion characteristic. An ideal push-pull amplifier will cancel all even harmonics. You already know from **Fig. 30** that the output transformers have a small even harmonic content due to the winding imbalances.

The second harmonic can also occur if there is an imbalance in the bias current between the push-pull pair. A large imbalance can cause momentary saturation of the output transformer on each swing of the AC output voltage. Higher orders of even harmonics are produced if the dynamic characteristics of the output tubes vary significantly from each other.

If you want to use unmatched tubes, setting the DC bias to 22mA in each tube will still leave you with the dynamic unbalance caused mainly by differences in tube transconductance. This can lead to higher levels of even-order harmonic distortion (some may even find this sonically desirable). You can compensate for this by adding an AC bias adjust pot in series or in parallel with R142/R242. With the pot centered, the total resistance should still equal 15k. I don't know how much adjustment range will be needed, but if the pot resistance is not optimal you risk drawing too much current in the phase inverter with the pot at its minimum setting.

In order to provide a center channel output signal that is compatible with my subwoofer power amplifier, I adjusted the value of resistor R13 to provide a center channel output signal level equal to the Direct input signal level when the loudness control is set for unity gain.

REBUILDING

Before I fired up the soldering iron, I made a new set of wiring drawings using the LK-48 ScottKit manual as a guide. Because of all my circuit changes, I needed to add 11 more terminal strips on the main chassis (indicated by the arrows in **Fig. 37**) and one on the rear chassis near the IEC receptacle. I installed the new terminal strips and tube sockets using 6-32 stainless steel hardware (available from Digi-Key). Stainless will not have a galvanic reaction with the aluminum chassis if moisture is present, as could occur with zinc-plated hardware. The original hardware was cadmium plated and after all those years the plating had started to turn to a yellow-green powder (cadmium is one of the environmentally undesirable heavy metals, and cadmium plating is being banned under the European Cadmium Directive 91/338/EEC).

I wanted to place the RIAA EQ network as close to the phono preamp tubes as possible, so I lost the use of the front panel RIAA/NAB switch as terminal points. I also decided not to use the scratch filter switch for some of the low-pass input filter components.

I must marvel at the wiring layout of the original 222C. The Scott engineers and designers crammed quite a lot of circuitry into their products, and most of the H. H. Scott line used just one chassis size. I left the old can capacitor C207-C210 in place but did not make connections to the positive terminals. I did use its ground terminals as grounding points

FIGURE 37: Main chassis and added terminal strips.

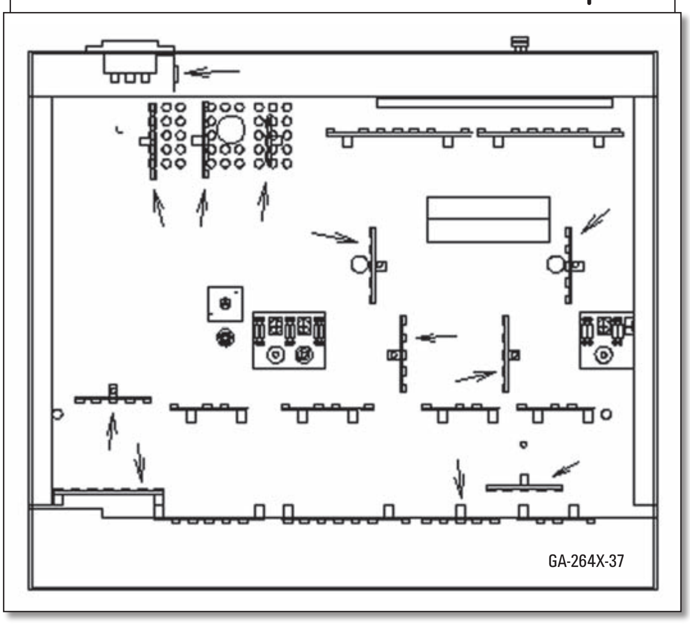
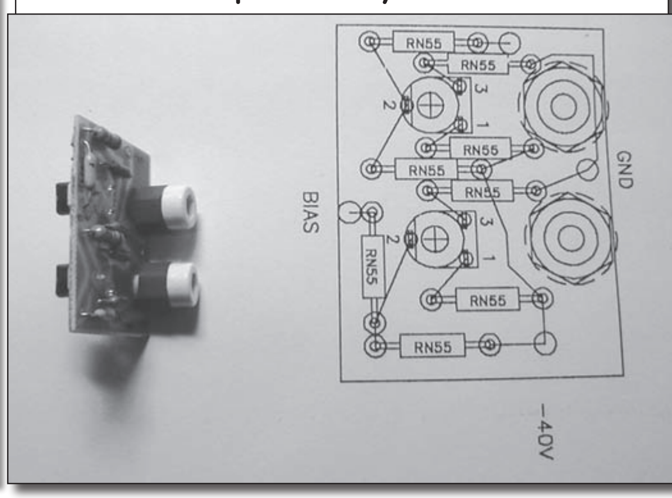


PHOTO 8: Bias adjust board layout detail.



for discrete replacement caps C3-C7.

Do yourself a favor and pre-tin any new old stock (NOS) items such as tube sockets, terminal strips, and so on. Make sure they take solder properly before you install them in the chassis. The original wire was solid core PVC wire, which is susceptible to melting with a solder iron. I used Teflon insulated wire, which is readily available on eBay in convenient 50′ lengths, for all my re-wiring, and I followed the original 222C wire colors as closely as possible.

Before making the bias adjust boards, I made a preliminary CAD parts layout to find the minimum size I needed. Then I located a space near the output tubes where the boards could fit. Since the bias adjust pots and pin jacks needed to fit through new holes in the chassis, I made a chassis drill jig from ⅛″ aluminum so the hole locations would be accurate. This is important because the pot adjusters cannot bind in the holes.

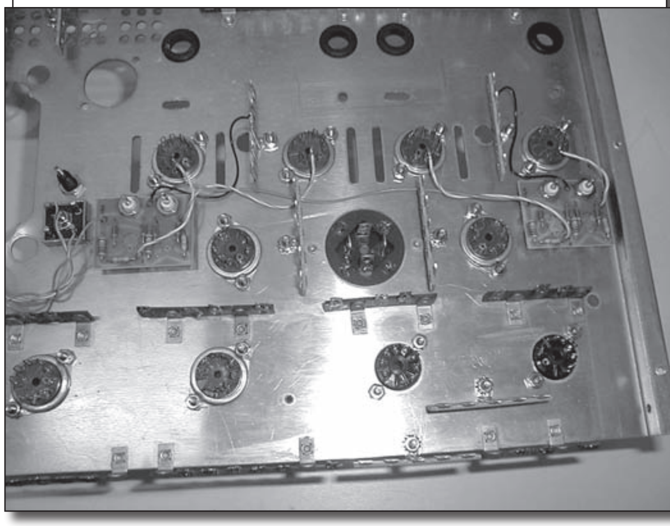
I routed the tracks for the two bias adjust boards with my Dremel router. The track layout and one of the finished boards is shown in **Photo 8**. Each board has two adjust pots with pin jacks that allow the mV drop across the 10Ω output tube cathode resistors to be monitored and adjusted from the top.

I added plastic spacers that allow the pin jacks to do double duty as circuit board mounting studs. These spacers bring the two bias adjust pots to the ideal height in their holes in the chassis. I first used the chassis drill jig to exactly locate the pots on the boards and soldered them in place. The installation of the boards in the main chassis is shown in **Photo 9**, while a close-up of a board installation is shown in **Photo 10**.

The new tone control and balance pots have slightly smaller metric bushings so they no longer self-center in the old mounting holes. I had to fit the front plate on and adjust each pot by trial and error until it was centered. The anti-rotation tabs are also on the opposite side, so the pot terminals are upside down from the originals. This doesn't affect the wiring at all.

I drilled a new anti-rotation hole for the input selector rotary switch because it didn't line up with the old one. The

PHOTO 9: Bias adjust boards installed in chassis.



second deck of this switch and the deck mounting screws came very close to one of the original terminal strips on the main chassis when I test-fitted the front chassis before I began wiring. I used a belt sander to grind down the length of the two rear deck spacer bushings to 7/32″ (same as the front deck spacers) and cut eight threads from the two 1-3/8 × 4-40 deck mounting screws to allow the switch to fit properly without interference. The rewired front panel is shown in **Photo 11**, with the original wiring drawing from the LK-48 manual. I didn't wire the headphone jack, as I mentioned earlier.

Once I removed the riveted input jacks and their insulated spacers, I reused the outer phenolic insulators for the new rear panel phono jacks. The original jacks were more closely spaced than in today's units, as was pretty much standard for the time. RCA plugs were quite small and didn't have threaded outer shells. The shield braid was fitted over and soldered to the ground shell.

As a result, most modern phono plugs won't fit the close spacing without the A and B channel plug shells hitting each other. I considered a custom fiberglass jack insulator, but the cutouts in the rear panel sheet metal preclude any closer spacing. I had to use small metric Deltron gold jacks (available from Mouser) to fit the available space. The holes in the Scott insulators are actually too big for the Deltron jack body, but the included gold flat washer works fine.

There is a problem inside the chassis because the included ground lug is too small and pulls through the insulator hole. I solved this by using a fiber shoulder washer at the rear of each jack. These washers are only available in inch sizes, so I had to bore the ¼″ ID of each washer to 7mm with a reamer to allow the threaded jack body to pass through the shoulder washer.

The good news is that holes in the original phenolic insulators were large enough to allow me to space the jacks for each channel further apart than the originals. Now modern sleeved interconnects will just fit without the plug shells touching. The phenolic insulators and fiber washers provide the only mechanical support for the new jacks, so they flex a bit when I plug in a pair of interconnects.

I also test-fitted the rear panel before doing any wiring to be sure the new gold speaker binding posts would not hit the main chassis. The rewired rear panel (**Photo 12**) is again shown with the original LK-48 drawing for that panel.

The original 222C used "mike lapel" wire (an insulated wire and a bare wire running in parallel inside black vinyl

PHOTO 10: Bias adjust board installation closeup.

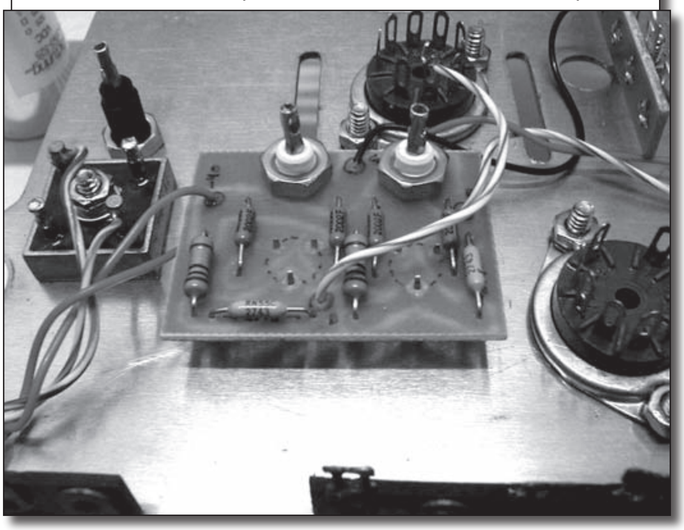


PHOTO 12: Rewired rear panel.

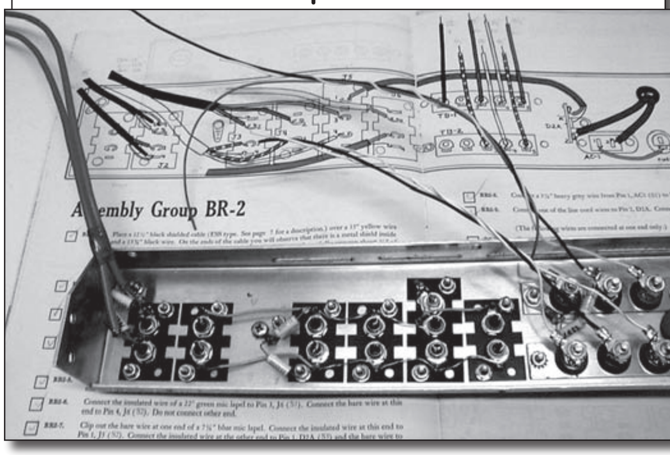


PHOTO 11: Rewired front panel.

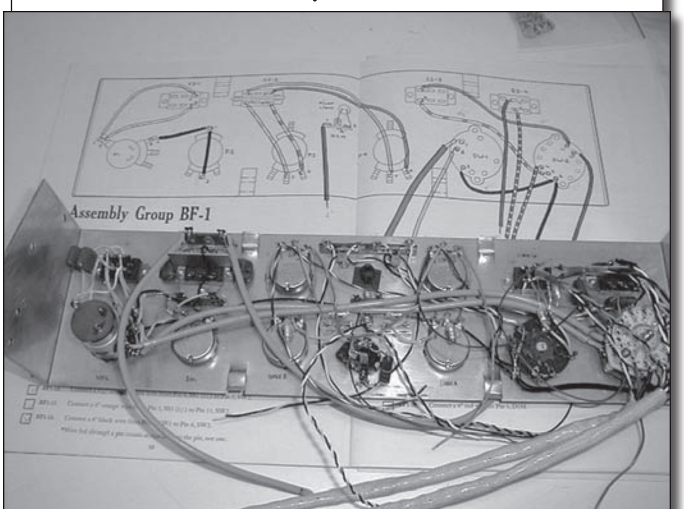
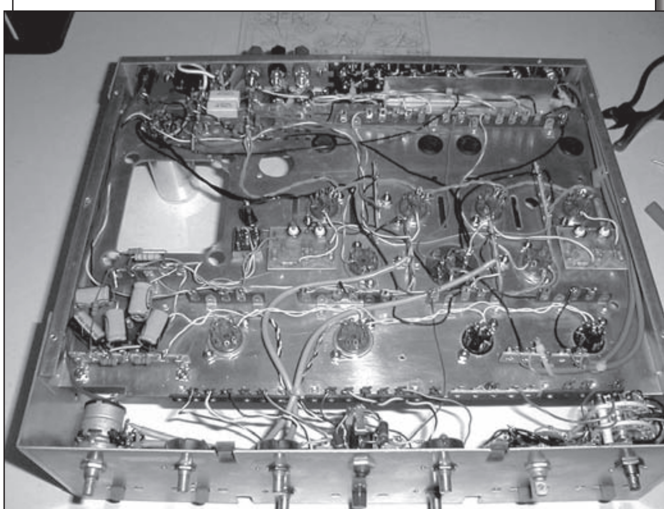


PHOTO 13: Front and rear panels installed to main chassis.



sleeving) for some of the line-level signals on the front panel. The wiring from the line-level input jacks on the back routed all the line-level signals for each channel together inside one piece of "conductive vinyl," which is sleeving with a conductive material inside which surrounded the hookup wire. I replaced these with twisted pairs running inside high temperature Varflex sleeving. The mag phono inputs originally used "ESS conductive cable," a hollow vinyl tube with a spiral metal shield inside. I used conventional twisted-shielded pairs here.

Photo 13 shows the modified front and rear panels re-installed and interconnected to the main chassis. The new resistors and five green filter caps for the output tube bias supply and the DC supply for the four 12AX7 tubes are installed at the lower right.

Photo 14 shows a bottom view of the chassis with all the wiring completed, just prior to the transformers being installed on the main chassis. I dressed the wiring with nylon wire ties where necessary.

Photo 15 shows the bottom of the modified 222C with the transformers installed. Although there are more parts

PHOTO 14: Main chassis bottom view with all wiring completed.

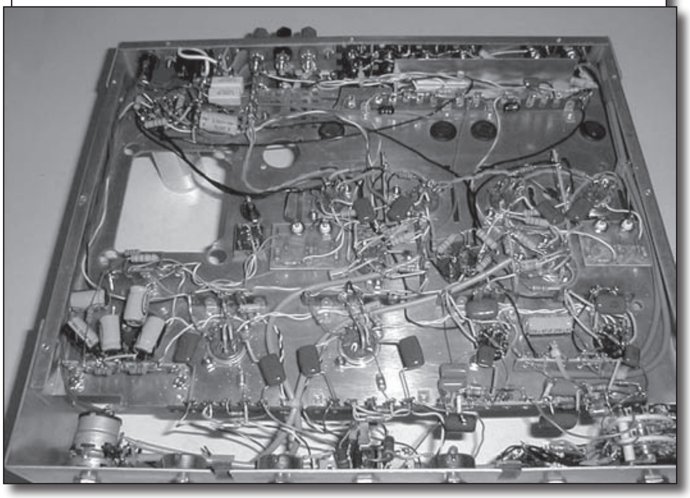
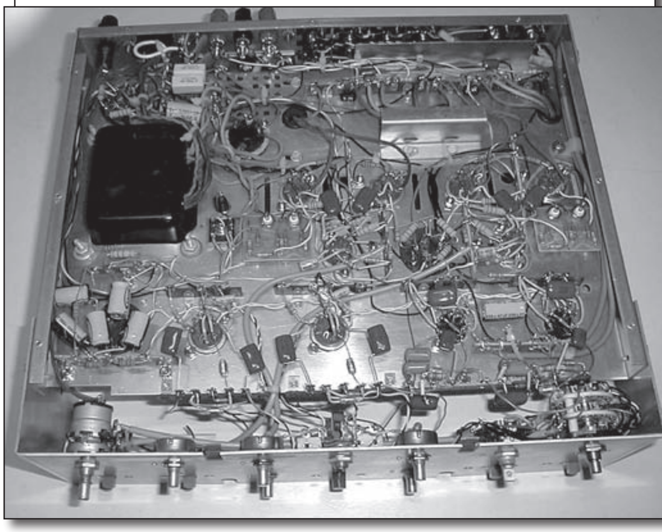


PHOTO 15: Bottom view with transformers installed.



than in the original unit, modern passive components are smaller and have thinner leads and higher maximum temperature ratings. This leads to a lower parts density, which will improve the cooling.

POWER UP THE UNIT–CAREFULLY!

Rather than just turn on the AC line voltage, I decided to take it one deliberate step at a time for safety sake. First I did a point-to-point resistance check and a visual inspection of all the solder joints and the parts and wiring clearances around any high voltage locations. Without any tubes in the unit, I connected a 30V DC power supply to pin 2 of the 5AR4 rectifier socket. Then I connected a fixed resistor from each tube socket plate to cathode and plate to screen pin in turn and made sure there was 30V DC at each plate and screen, a small voltage at each cathode, and no voltage at each control grid.

Next, I repeated these tests with a high voltage supply, slowly increasing the voltage while looking for any evidence of high voltage breakdown. This also made sure the electrolytic filter capacitor charge currents were not excessive the first time they had voltage applied. I installed all the tubes except the HV rectifier and used a Variac to slowly increase the AC line voltage.

I measured the 6.3V AC heater voltage and made sure all the tubes were lit. I measured the 50V DC across the four 12AX7 preamp tube heaters and the -40V DC output tube grid bias voltage. I set the four grid bias pots fully counterclockwise (maximum grid cutoff).

Only when all these tests were successful did I install the HV rectifier tube and begin operational tests. I connected shorting plugs to the EXTRA input, turned the loudness to zero, connected 8Ω load resistors to the 8Ω output binding posts, and plugged in the AC cord. The inrush current limiter prevented the bright heater flare-up that was so noticeable in the original amplifier, especially in the series 12AX7s.

The Scott 222C, 222D, and LK-48 family of amplifiers called for a total output tube pair bias current of 44mA, or 22mA per tube. (The 7189 data sheet calls for an idle current of 15mA per tube.) I set the bias for each output tube to Scott's recommended 22mA (220mV

TABLE 4: PARTS LIST.

Symbol	Value	Description	Vendor	Part no.
C1, C2	10nF 1kV	ceramic Y5P	Mouser	140-102P9-103k
C3-C7	100µ 100V	aluminum 105c	Mouser	140-HTRL100V100
C101, C202	40µ/20µ/20µ/20µ at 525V	aluminum FP	AES	C-EC40-20X3-525
C102, C202	47µF 250V 10%	aluminum 105C	Mouser	140-HTRL250V47
C103, C203	100nF 250V 10%	polypropylene	Mouser	1429-2104
C104, C128, C204, C228	100nF 100V 10%	polypropylene	Mouser	1429-1104
C105, C205	27nF 100V 10%	polypropylene	Mouser	1429-1273
C106, C206	47nF 250V 10%	polypropylene	Mouser	1429-2473
C107, C207	1µF 250V 10%	polypropylene	Mouser	1429-2105
C108A, C108B, C208A, C208B	1n5 600V 5%	polypropylene	Mouser	715P15256JD3
C109, C209	820pF 50V 5%	polystyrene axial	Mouser	23PS182
C110, C210	180pF 50V 5%	polystyrene axial	Mouser	23PS118
C111, C211	56pF 500V 5%	ceramic NBP	Mouser	140-50N5-560J
C112, C212	1200pF 50V 5%	polystyrene axial	Mouser	23PS212
C113, C213	1500pF 50V 5%	polystyrene axial	Mouser	23PS215
C114, C118, C119, C125, C126, C214, C218, C219, C225, C226	100nF 400V 10%	polypropylene	Mouser	1429-4104
C115, C215	33µF 25V 20%	aluminum 105C	Mouser	140-HTRL25V33
C116, C216	4700pF 50V 5%	polystyrene radial	Mouser	23PW247
C117, C217	470pF 50V 5%	polystyrene axial	Mouser	23PS147
C120, C220	47pF 500V 5%	ceramic NPO	Mouser	140-50N5-470J
C121, C122, C221, C222	22n 250V 10%	polypropylene	Mouser	1431-2223k
C123, C223	4µ7 160V	aluminum 105C	Mouser	140-HTRL160V4.7
C124, C224	120pF 500V 5% silver mica		Mouser	5982-15-500V120
C127, C227	220pF 500V 5% silver mica		Mouser	5982-15-500V220
CL1	250 1.7A	CL200 inrush current limiter	Mouser	527-CL200
D1	6A 200V	diode bridge	Mouser	583-RS603
F1	2.5A slo-blo	fuse 3AG		re-use
J1		IEC receptacle		
J2		tip jacks, bias voltage (B1k)	Mouser	530-105-0803-001
J3		phono jack, gold (wht)	Mouser	161-1005
J101-J106		phono jack, gold (blk)	Mouser	432-0100
J107, J108, J207, J208		speaker post, gold (red)	AES	S-H266R
J109, J209		speaker post, gold (blk)	AES	S-H266B
J110, J111,J210, J211		tip jacks, bias voltage (red)	Mouser	530-105-0802-001
J201-J206		phono jack, gold (red)	Mouser	432-0500
R1		680k 1W 10% carbon composition		re-use
R2	15 1W 5%	metal oxide	Mouser	261-15
R3-R5	18 1W 5%	metal oxide	Mouser	261-18
R6	7k5 2W 5%	metal oxide	Mouser	262-7.5k
R7, R8	91 1W 5%	metal oxide	Mouser	261-91
R9A, R9B,R10A, R10B	1M pot, linear (bass, treble)	carbon	Mouser	31VA601
R11	500k pot, dual tapped at 18k5 (loudness) w/sw	carbon	Mouser	re-use
R12	500k pot, linear (balance)	carbon	Mouser	31VA505
R13	120 ¼W 5%	carbon film	Mouser	291-120
R101, R201	39 5W 10%	wirewound	Mouser	280-CR5-39
R102, R202	3k5 3W 5%	wirewound	Mouser	588-23J-3.5k
R103, R203	9k1 3W 5%	wirewound	Mouser	588-23J-9.1k
R104, R204	16k 1W 5%	metal oxide	Mouser	261-16k
R105, R205	10k0 ½W 1%	metal film	Mouser	273-10.0k
R106, R206	47k5 ¼W 1%	metal film	Mouser	271-47.5k
R107, R207	3k32 ¼W 1%	metal film	Mouser	271-3.32k
R108, R208	22k ¼W 1%	metal film	Mouser	271-221k
R109, R209	1k50 ¼W 1%	metal film	Mouser	271-1.5k
R110, R210	82k5 ¼W 1%	metal film	Mouser	271-82.5k
R111, R116, R119, R136, R145, R150, R211, R216, R219, R236, R245, R250	1M00 ¼W 1%	metal film	Mouser	271-1.0M
R112, R212	110k ¼W 1%	metal film	Mouser	271-110k
R113, R213	806 ¼W 1%	metal film	Mouser	271-806
R114, R214	1M50 ¼W 1%	metal film	Mouser	271-1.5M
R115, R215	90k9 ¼W 1%	metal film	Mouser	271-90.9k
R117, R217	2k80 ¼W 1%	metal film	Mouser	271-2.8k
R129, R229	10k0 ¼W 1%	metal film	Mouser	271-10k
R118, R218	100k ¼W 1%	metal film	Mouser	271-100k
R120, R140, R152, R155, R220, R240, R252, R255	274k ¼W 1%	metal film	Mouser	271-274k
R121, R221	3M3 ¼W 5%	carbon film	Mouser	291-3.3M
R122, R123, R222, R223	2M2 ¼W 5%	carbon film	Mouser	291-2.2M
R124, R130, R214, R230	100k ½W 1%	metal film	Mouser	273-100k
R125, R131, R225, R231	2k744 ¼W 1%	metal film	Mouser	271-2.74k

R126, R127, R128, R227, R228	68k1 ¼W 1%	metal film	Mouser	271-68.1k
R132, R232	22k1 ¼W 1%	metal film	Mouser	271-22.1k
R133, R233	150k ½W 1%	metal film	Mouser	273-150k
R134, R234	249k ¼W 1%	metal film	Mouser	271-249k
R135, R235	1k00 ¼W 1%	metal film	Mouser	271-1k
R137, R237	15k0 ¼W 1%	metal film	Mouser	271-15k
R138, R238	1k21 ¼W 1%	metal film	Mouser	271-1.21k
R139, R239	221k ½W 1%	metal film	Mouser	273-221k
R141, R144, R241, R244	8k25 ¼W 1%	metal film	Mouser	271-8.25k
R142, R143, R242, R243	15k 2W 5%	metal oxide	Mouser	262-15k
R146, R149, R153, R154, R246, R249, R253, R254	20k0 ¼W 1% metal film	Mouser	271-20k	
R147, R148,	20k 1W trimpot	cermet trimpot	Mouser	652-3306F-1-203
R247, R248				
R157, R158,	10 1W 5%	metal oxide	Mouser	261-10
R257, R258				
R159, R160,	1k 1W 5%	metal oxide	Mouser	261-1.0k
R259, R260				
R161, R261	68 1W 5%	metal oxide	Mouser	261-68
R162, R262	8k2 ¼W 5%	carbon film	Mouser	291-8.2k
R163, R263	150 ½W 5%	carbon film	Mouser	293-150
S2	input select sw	Electroswitch 4P 6 pos, D4C0406	Newark	22F611
S3	tape monitor sw	2P2T slide		
S4	stereo selector sw			re-use
T1		TR 12-3-2 power transformer		re-use
T101, T201		TR 8-5-1 output transformer		re-use
V1		GZ34/5AR4 FW rectifier		
V101, V102		12AX7 dual hi-mu triode	Raytheon	
V201, V202		ECC83/12AX7 dual hi-mu triode	Raytheon	
V103, V203		6U8/GH8 triode-pentode	RCA	
V104, V105, V204, V205	7189 beam power pentode	Telefunken	re-use	
		1⅜″ diameter can cap insulators (2)	AES	S-H120
		sockets, (6) 9-pin, shield bse (6U8)	AES	P-ST9-700
		sockets, (4) 9-pin (7189)	AES	P-ST9-511
		socket, (1) octal (5AR4)	AES	P-ST8-807
		extruded shoulder washers (13)	Digi-Key	SC1145-ND
		Terminal strip, (1) 2-lug, 1 ground	AES	P-0201H
		Terminal strips, (6) 5-lug, 1 ground	AES	P-0501H

across the 10Ω cathode resistors) and let the 222C operate while I continued to check bias current for 30 minutes. The 7189 bias drifted ±4mA and showed some interaction between the adjustments for each pair. I settled on 20mA and accepted a couple milliamps of upward drift.

Next I measured the ripple on the 7189 plate and screen supply voltages (see **Photo 16** at 5V/cm plate and 1V/cm screen). While the plate supply ripple is now down to 3Vp-p, I believed there was still too much ripple on the

PHOTO 16: AC ripple voltage at the plate and screen grid, initial test of redesigned 222C.

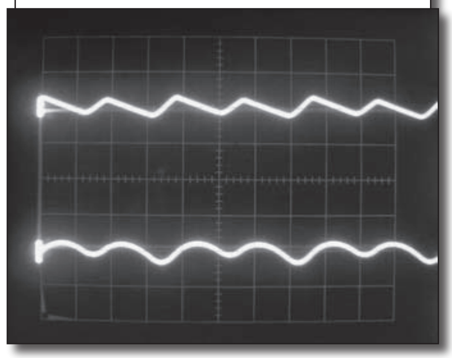
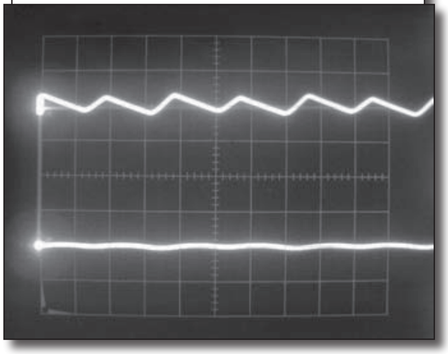


PHOTO 17: AC ripple voltage at the plate and screen grid, final design.



screens. I increased R102/R202 from 560Ω, which I originally scaled from the 222D values, to 3k5. While this reduced the screen grid ripple, the cascade effect on the other supplies made them too low.

Back to SPICE to adjust each series resistor to restore the proper B+ values to all the other tubes. The final values are those shown in **Fig. 33**. **Photo 17** shows the AC ripple voltage at the 7189 plate and screen grid with this final design of the dual mono power supply.

I again reset the output tube bias to 22mA, and after a half-hour the zero signal plate voltage supply measured +445V DC, with +440V DC at the plates. Reducing the line voltage to the 117V AC in use at the time the 222C was designed produced a still-high +432V DC at the plates. Short of rewinding the HV secondary on the power transformer, or purchasing a suitable replacement, I didn't have much recourse except to live with it.

The zero signal plate dissipation is now 9.7W versus 8.8W with the specified 400V DC plate voltage (see the two diagonal dashed lines in **Fig. 38**). While the plate voltage is 10% above the maximum, this is still less than the rated 12W plate dissipation for a 7189. I don't want to add more series resistance between the rectifier and the plate supply as regulation will suffer. Fortunately, the voltage and current swings along the linear load line in a Class AB1 push-pull circuit take the dynamic plate dissipation away from the constant power parabola in the plate curves.

Without any active voltage regulators the power supply voltages vary with both AC line voltage and with output power. The variations with line voltage at 1kHz 1W/channel output are shown in **Fig. 39**. The variation with 1kHz output power at a fixed AC line voltage of 120V AC is shown in **Fig. 40**. I have also thrown in a graph of amplifier efficiency versus output power in **Fig. 41** for your edification. At normal listening levels the efficiency of my modified 222C is about 3%. Raising all those cathodes to their thermionic operating temperatures is not free, and the 222C consumes 42W to do so.

I applied a 1kHz 0.5V RMS sine wave to the Extra Input and adjusted the loudness for 2W/channel. I let the amplifier run in for 30 minutes sitting on its bottom plate, then tipped the unit on its side and measured component temperatures with an Extech 42530 infrared non-contact digital thermometer. This unit has a laser pointer that pinpoints the component being measured. The hottest components (other than the

FIGURE 39: Variation of DC supply voltages with AC line voltage.

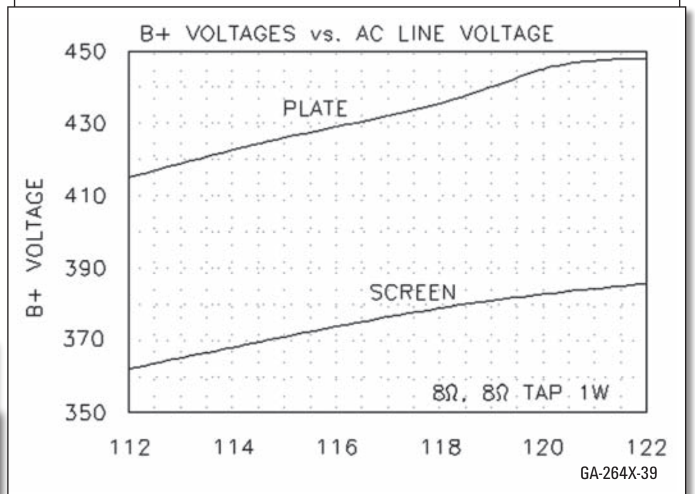


FIGURE 38: Plate curves, load line, and power dissipation.

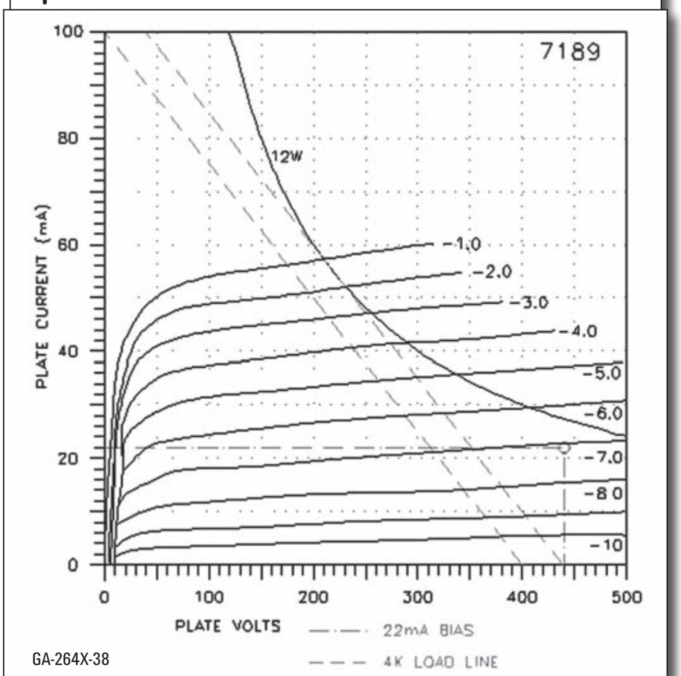
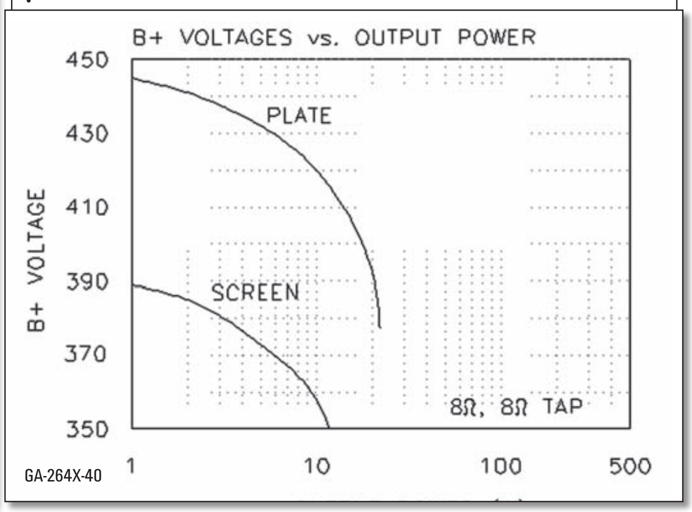


FIGURE 40: Variation of DC supply voltages with output power.



tubes) and their temperatures are listed in **Table 6**. Finally, I ruggedized the larger discrete aluminum caps to the chassis with RTV, and then installed the bottom cover and the front panel.

The front and rear of the completed unit are shown in **Photos 18** and **19**. **Photo 20** shows a top view with the four bias white adjust jacks and blue pots near the two outside output tubes. The black common bias jack is to the right near the power transformer. I used clear label tape to label the tube types on the top of the chassis, and to label the inputs and outputs on the top of the rear chassis. The latter makes it easy to identify the connections from the front of the unit.

The output tubes in **Photos 18-20** are Russian-made "6п14п" (translates in English to 6p14p—see the Sidebar in part 4 for comparison tests with the 7189).

TABLE 6: HOTTEST COMPONENT TEMPERATURES.

Component	Max Temp °C
CL1	75
R103/R203	60
Chassis at V1	60
R101/R201	50
R102/R202	50
R104/R204	48
T1	41
D1	36

FIGURE 41: Amplifier efficiency vs. output power.

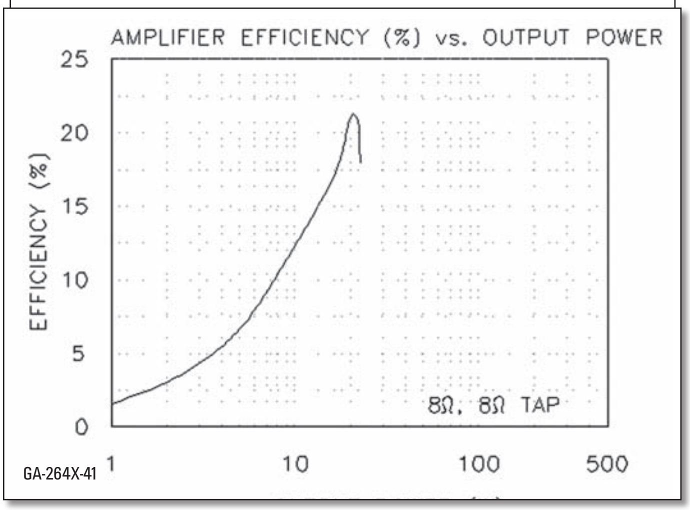
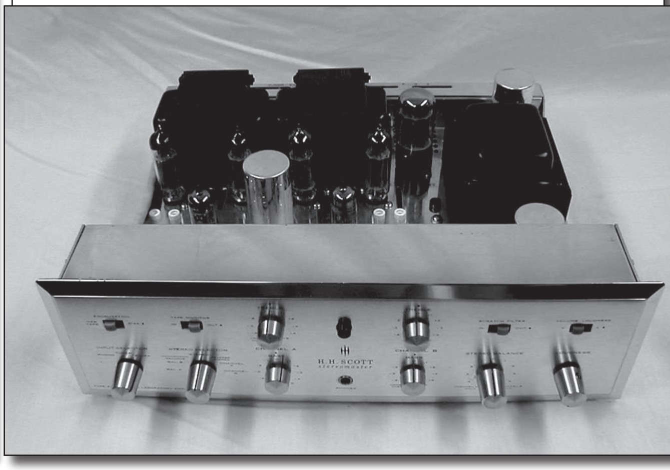


PHOTO 18: Front view of completed unit.



Next month, I'll wrap up this extensive restoration project with a listening audition and a look at the responses of the modified amp. *aX*

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8. "Product Review, The VacuTrace," Charles Hansen, *audioXpress*, pp. 56–63, Feb. '02 (The article shows plate curve traces both with the Sofia and the VacuTrace).

9. *Tube Lore*; L. Sibley; Sept. 1996, with supplements 1997, 2002; Chernay Printing.

SOURCES

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1-480-820-5411
www.tubesandmore.com

Digi-Key Corp.
701 Brooks Ave. South
Thief River Falls, MN 56701-0677
1-800-344-4539
www.digikey.com

Mouser Electronics
958 N. Main
Mansfield, TX 76063-4827
1-800-346-6873
www.mouser.com

Newark Electronics
Chicago, IL 60640
1-800-263-9275
www.newark.com

PHOTO 19: Rear view of completed unit.

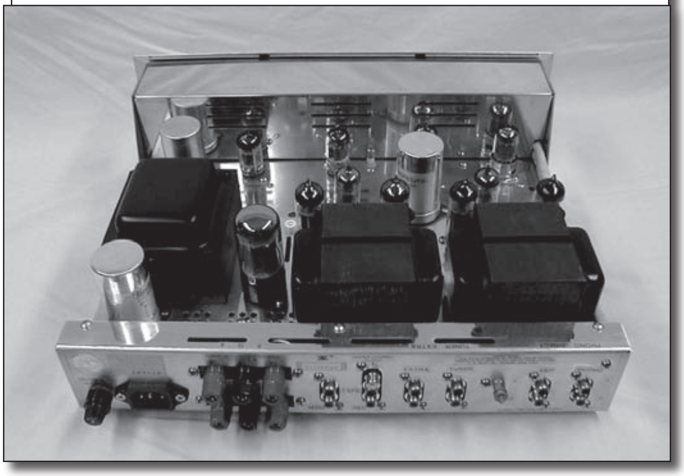
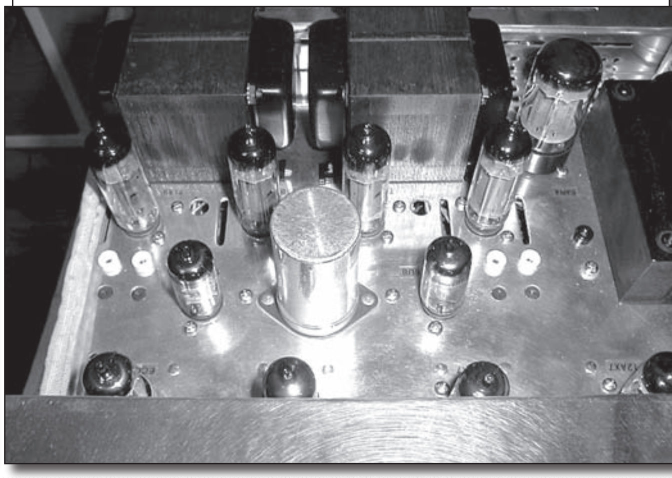


PHOTO 20: Top view of completed unit.



An Improved Split-Load Phase Inverter

Here's a new phase inverter design that avoids some of the shortcomings of the original circuit.

By Robert Bennett

The split-load phase inverter is a time-honored circuit used in audio amplifiers as early as the mid-1930s[1]. It has the virtue of simplicity and economy, using, as it does, only one triode. It has, however, a few undesirable features:

(a) It has no gain—in fact, the output is only about 95% of the input;

(b) The two outputs tend to become unbalanced at high frequencies;

(c) The two outputs have different impedances—low for the cathode output and high for the anode output;

(d) A load applied to the cathode output causes a higher output on the other (anode) output, but not vice versa.

FIXES

Now the lack of gain is easily dealt with by using another amplification stage. In Adriaan Hammer's article[2], a 12AX7 double triode is used for amplifier and phase splitter. Good HF balance between outputs can be obtained by loading the cathode output with a small capacitor—a very neat solution (**Fig. 1**). However, (c) and (d) are not so easily dealt with, and often cause trouble if the output valves draw grid current on a signal peak.

If the input to the phase splitter is made between grid and cathode rather than grid and earth, all these undesirable features are eliminated. A neat way of doing this is shown in **Fig. 2**. By means of a capacitor, the "top" end of the pentode's load resistor is connected to the cathode of the phase splitter. This means that the output of the pentode is applied between the triode's grid and cathode—exactly as required. The pentode's gain is unaffected by such treatment because its anode resistance is much greater than the load resistor.

The triode gain is half of the normal amplifier gain because it is divided between the two outputs. This way of analyzing the circuit is easier—in my opinion—than the alternative of considering it as a feedback circuit[3]. The overall voltage gain for the circuit of **Fig. 2** will be greater than 200, compared to the 70 or so for the circuit of **Fig. 1**.

Figure 2's circuit has been referred to as a bootstrap circuit, and is believed to have been invented in the late 1940s[4]. Single triode phase splitters with floating microphone inputs[5] and phase splitters in superhet (erodyne) radio circuits where the detector was floating[6] were described earlier.

Note: A power supply of about 250V would be suitable for the circuits of **Figs. 1** and **2**. Good regulation is not required, but a well-filtered supply is essential to avoid hum.

Some people consider pentodes "beyond the pale" because of extra noise or distortion compared with triodes. You can replace the pentode with a "cascode" combination of two triodes as shown in **Fig. 3**. This should reduce the noise. Whether or not the distortion produced by a cascode circuit is any better or worse than that produced by a pentode is a matter of opinion.

In **Fig. 2**, the triode grid is capacitor-coupled to the pentode anode. You can use direct-coupling of the triode to the pentode, which would be desirable if negative feedback is incorporated. To do this, the resistor R9 should be a 5kΩ potentiometer, and you must carefully adjust the bias.

Negative feedback can be provided in the same way as **Fig. 1**, by inserting a 100Ω resistor in the cathode circuit.

One aspect of phase splitter design not often mentioned, but which requires careful attention, is the heater-cathode circuit. V2 in **Figs. 1** and **2**, and V2 and V3 in **Fig. 3** have their cathodes "away up in the air" and if their heaters are at earth potential there is a considerable potential difference between heater and cathode (100-150V). This can cause heater to cathode breakdown and can also cause annoying hum.

To avoid this, the heater circuit can be biased positive to reduce the potential difference between heater and cathode. If a separate heater winding is available on the transformer, then you could use

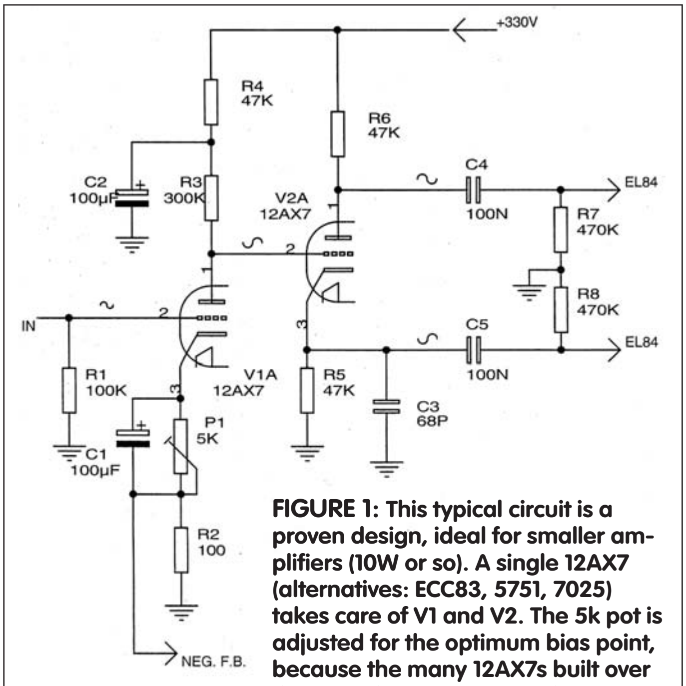


FIGURE 1: This typical circuit is a proven design, ideal for smaller amplifiers (10W or so). A single 12AX7 (alternatives: ECC83, 5751, 7025) takes care of V1 and V2. The 5k pot is adjusted for the optimum bias point, because the many 12AX7s built over the years by various manufacturers are not all identical.

a bias of +100V, or join the heater to the cathode. If this is not possible, you could use a smaller bias of +50V without straining the heater to cathode insulation of the other valves too much.

Notes on the valve types mentioned are given in a footnote[7]. *aX*

***Robert Bennett** was born in England and raised in Christchurch, New Zealand. He studied chemistry at Canterbury University, graduating M.Sc. in 1984. Since then he has worked as an analytical chemist in various places in New Zealand. He currently lives in Auckland with his partner and two children, and works for an industrial water treatment firm. He has been interested in radio and electronics since he was 15.*

REFERENCES

1. *Radio Constructor's Guide*, National Magazines Ltd. Wellington NZ (1936), p. 134. Circuit used 24A (tetrode) driving a 56 phase splitter driving push-pull 50s (see footnote 7 for notes on valve types mentioned).

2. *audioXpress* September 2003, pp. 56-57.

3. F.Langford-Smith (ed.) *The Radiotron Designer's Handbook*, 4th edition (1954), p. 523.

4. E. Jeffery, *Wireless World* 53(8), p. 274 (August 1947).

5. *Radio Constructor's Guide*, National Magazines Ltd. Wellington NZ (1936), p. 135. Circuit developed by *Wireless Weekly* magazine (Australia); used a 57 (triode connected) as phase splitter driving push-pull 56s driving push-pull 50s.

6. *RCA Receiving Tube Manual RC-13* (1937), p. 185. Circuit used a 6H6 detector driving a 6F5 phase inverter driving push-pull 6F6s or 6L6s.

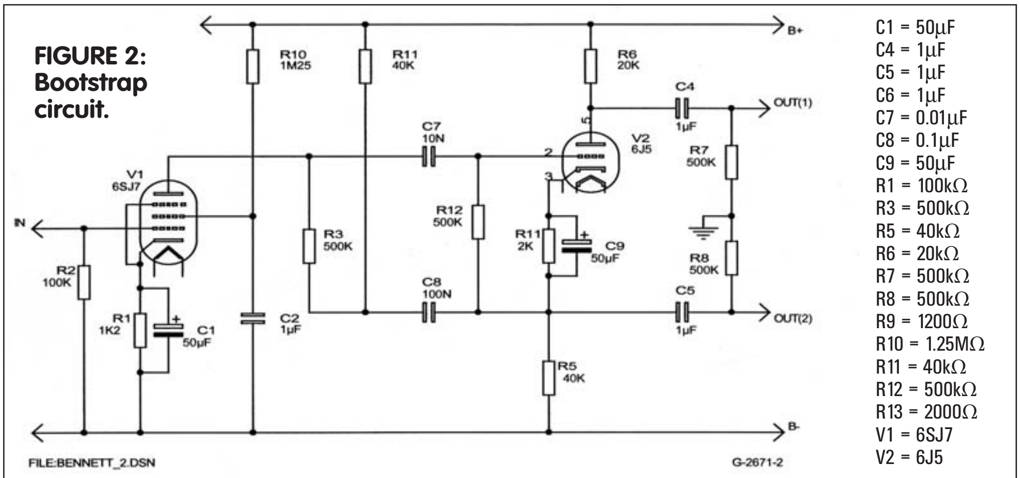


FIGURE 2: Bootstrap circuit.

7. The 57 is a pentode with a 2.5V heater, having characteristics similar to the later types 6C6 and 6J7, and fairly similar to the much later EF86/6BR7. Used as a triode, by connecting the screen and suppressor grids to the anode, it has a medium mu-factor of 19.

The 56 is a general-purpose triode with a 2.5V heater. It has a mu of 13.8.

The 50 is a large, directly heated (7.5V filament) triode output valve

with a mu of 3.8.

The 24A is a tetrode with a 2.5V heater, having characteristics roughly similar to the later pentode types 57, 6C6, and 6J7. It was normally used as a RF amplifier rather than an audio one. Because of the lack of a suppressor grid, the anode characteristics have a nasty kink at low anode voltages: to avoid this, the working anode voltage needs to be at least 50V higher than the screen voltage.

The 6F5 is a high-mu triode with characteristics similar to one section of an ECC83/12AX7.

The 6H6 is a double diode with separate cathodes. Most consumer radio circuits used a double-diode-triode such as a 6Q7 as combined detector and amplifier, but then this phase splitter circuit would not have been possible.

The 6F6 is an output pentode with an anode dissipation of 10W, an octal-based version of the earlier types 2A5 and 42.

Its characteristics were roughly similar to those of the later beam power valve type 6V6.

The 6L6 was the original earliest beam power valve in production. Beefed-up versions of the 6L6 are still available (6L6GC). The original version had a maximum rating of 19W anode dissipation.

The 6SJ7 was a later, single-ended version of the 6J7 octal-based pentode, with slightly higher gain.

The 6J5 is a general-purpose triode with characteristics very similar to one section of a 6SN7 or a ECC82/12AU7double triode.

The 12AU7/ECC82 is a double triode with a medium mu factor of 17.

The 12AX7/ECC83 is a double triode with a high mu factor of 100.

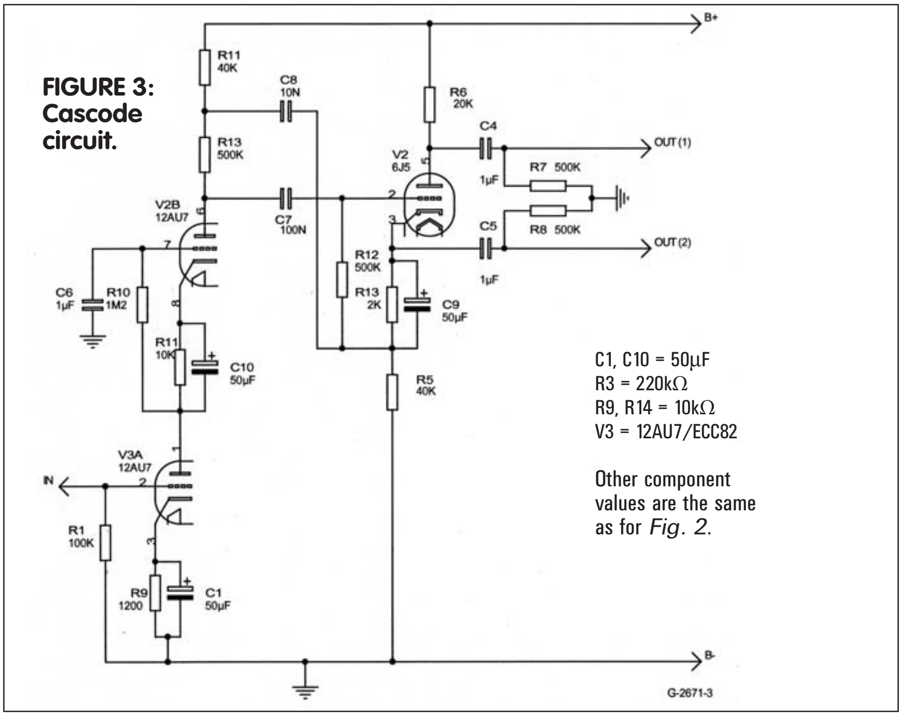


FIGURE 3: Cascode circuit.

Upgrading a Unique Stereo Column System

Revising his column speaker design, this author adds more, higher-quality drivers for an improved response.

By Roger Russell

In a previous article[1], I described a single long column of wide-range drivers and equalizer. I have now improved this version using even smaller drivers that are of much higher quality and also more expensive. The response is smoother and more extended with lower distortion. The appearance has been revised and, to me, is much more pleasing to look at. I have added protection circuitry and a grille to complete the system (**Photo 1**).

PHOTO 1: The finished system in author's living room.

THE DRIVERS

The new drivers, made by Vifa in Denmark, cost $68 each, making a total of $3400 for 50 of them. They are not available from the usual Vifa suppliers. However, they are available from Haig Audio in Michigan.

The drivers in my first article were 4″, The new ones are 3½″, with a copper shorting cylinder around the pole piece (**Photo 2**). Unlike some shorting ring designs, this ring extends through the entire height of the gap and beyond. This is known to reduce distortion—particularly in the voice range—by a factor of 10 or more. The ring also greatly reduces any impedance rise at the higher frequencies.

In addition, the cone is made of a woven fiberglass material, which has been used in various dome midranges and tweeters for many years and has proven to furnish smoother response without the peaks sometimes found in rigid materials such as those having metal cone or dome construction. The magnet structure is not only reverse-polarized but also has an external steel cover, which reduces stray magnetic fields that could affect the picture of a television set if placed in close proximity.

Each driver diaphragm assembly is very light, weighing only 2.5 grams, or 0.088 ounces. Use of a flat spider, which is the major centering device for the voice coil, assures the same linear cone travel in each direction while keeping the coil exactly centered in the magnetic gap. A thin rubber surround eliminates the possibility of deterioration often found with foam surrounds. The square basket allows closer spacing between the drivers.

MULTIPLE DRIVERS AND POWER HANDLING

Although the cone area of the new drivers is smaller, I have used 25 instead of 20 of them in each system, and each driver has more than twice the cone excursion capability. They are arranged in a column 87¼″ high. The effective cone diameter is only 2 11/16″. The area is 5.7 in^2.

So how can such tiny drivers shake the room? First, if you add up the area of all the cones, the total effective area comes to 142 in^2. In comparison, a 15″ woofer has an effective area of 120 in^2, making the 25 driver area just slightly more than a 16″ woofer—impressive! Because the total cone area is very large, very little excursion is needed at the lower frequencies.

Each driver can handle at least 12W, so 25 drivers can handle 25 × 12, or 300W. Twenty-five voice coils can dissipate more heat than one 2″ or 3″ voice coil. When used with the equalizer, the system can be thought of as having reduced efficiency as the frequency is decreased below 150Hz. This is because more power is required to maintain a constant sound level output. It is most inefficient at 30Hz, where it requires the most power. Despite this special power requirement, each driver handles only 1/25 of that.

SYSTEM SENSITIVITY

At a distance of 2m, I measure an average of 10dB higher output from the column compared to a single driver in the test box. The sensitivity of a single driver is rated at 82.5dB. This means that the sensitivity of the column is effectively 92.5dB at that distance.

But wait, this increase in sensitivity is not for the whole frequency range. It only goes from 25Hz to about 1500Hz. As frequency increases beyond 1500Hz, the output gradually becomes the same as the output of the single driver. It is even less than the single driver above

PHOTO 2: The 3½″ driver used in this system.

5kHz by a few dB, all the way to 20kHz.

This behavior is typical of columns composed of individual drivers with wide overall dispersion. The effect is related to driver size, directionality, and driver spacing[2]. The change in output for my system is compensated with equalization at the higher frequencies to maintain a constant output.

The 10dB gain advantage over much of the frequency range is why the system requires very little power most of the time. In fact, it is only ⅒ of the power compared to a single driver for the same listening level. The reduced column gain at the higher frequencies is not where most energy in music is. The fundamental portion of the spectrum for most musical instruments occurs below 1kHz, and you can take full advantage of the higher sensitivity in this area. The extra power needed to maintain good response at the higher frequencies is not a problem for 25 drivers.

Of course, for a single driver, the sound level decreases by 6dB for every doubling of distance. In the long column, sound level decreases only 3dB for every doubling of distance. So, within limits, there can be a further advantage in effective sensitivity at distances greater than 2m.

In addition, sensitivity today is often referenced to a driving voltage of 2.83V instead of 1W, which means a 4Ω system will play 3dB louder than an 8Ω system for the same voltage. However, the 4Ω system is actually drawing 2W of power from the amplifier to accomplish this while the 8Ω system draws only 1W. This is confusing for the buyer and allows the manufacturer of the 4Ω system to advertise a figure that could be mistaken for higher sensitivity.

SYSTEM PROTECTION

The completed terminal board is shown in **Photo 3**. I use a bi-stable resistor, which looks like a small ceramic disc capacitor, in series with the system. This normally has a very low resistance, but when excessive current passes through it, it switches to a high resistance and effectively prevents the system from playing. You can easily reset the resistor by turning down the volume for about 10 seconds to restore normal performance. Because the systems will handle very high power, it is unlikely that the resistors will ever trip, but if they do, you will be suddenly plunged into a very low sound level.

To overcome this sudden change, I have added two high current incandescent lamps in series and connected the series pair across the resistor. This way, if the resistor trips, the current is then directed through the lamps. The speakers will still play, but the higher resistance of the glowing lamps will limit the power to the speakers. The higher the power to the system, the brighter the lamps will glow and the higher the resistance in series with the system will be. Although this protection circuit will reduce the chance of overdriving the system, it is only an approximation and will not protect under all circumstances.

I attached the lamps with clear silicone RTV sealer, which holds the lamps firmly and will withstand vibration and reasonably high temperatures as well. Attach the two solder lugs with pop rivets in a place where they will not show later when you install the board in the cabinet.

I made the terminal board out of a piece of ¼″ tempered Masonite and sprayed it black on the visible side. Be sure the board is no higher than the diagonal of the terminal hole; otherwise, you will not be able to fit the terminal board assembly through the terminal hole of the finished cabinet. The board in **Fig. 1** is 5″ × 3¾″.

The dashed lines in **Fig. 2** show the outline of the hole in the cabinet. I used a five-way binding post that accepts a dual banana plug (**Photo 4**).

Drill the eight 3/32″ diameter mounting holes for the terminal board inside the back of the cabinet before assembling the cabinet; otherwise, you will need to drill the holes inside the finished cabinet with a drill that will need to fit through one of the driver holes. Use foam tape to seal between the terminal board and the cabinet.

THE CABINET

I decided to have a local cabinet shop make the cabinets for me at a cost of $750 per pair, including painting and the base. In **Fig. 3**, I have shown decimals as well as fractions for the driver holes in case you use computer-controlled machinery to make the front panel. The location of the braces is indicated by the arrows on the left side.

This pair of cabinets was made out of Baltic birch multi-plywood. However, you could also use ¾″ kitchen counter underlayment, which is less expensive and is a very dense and fine grain particleboard. All the pieces can be glued and nailed together with finishing nails, which are countersunk.

The center-to-center distance of the driver holes is 3 7/16″, including the space between the drivers. The distance from the bottom of the cabinet to the center of the bottom driver hole is 2⅜″. I used the same distance from the top of the cabinet to the center of the top driver hole. The terminal hole is 2½″ high and 3½″ wide and is located directly behind the second driver from the bottom. It is centered 5 13/16″ up from the bottom of the cabinet.

Figure 4 shows a cross-section of the column including a driver hole, the terminal hole, and a brace. I used 11 internal braces—½″ thick with two 2½″ circular holes—in the system to break up internal resonances and strengthen the walls of the cabinet. I have indicated that the front dimension of the brace is 4 3/16″ and the rear dimension is 5 13/16″, but you may need to adjust this slightly to fit exactly in the cabinet.

The left and right front edges of the cabinet are beveled at 45°, which could help reduce edge diffraction effects and provide a more interesting appearance. The edges of the terminal hole are also beveled the same way.

When you mount a driver in the ¾″ front board, the terminals will hit the side of the hole. I used an undercut at the inside of each driver hole to allow space for the terminals and the hookup wire that is soldered to the terminal (**Fig. 5**). Be sure the undercut is located at the lower portion of the circle as indicated. Rout at each location with ¼″ radius. The total width is 1 7/16″.

You must make the undercut before the cabinet is assembled. To do this, you must accurately locate the centers of all the driver holes and then drill a hole size that will fit the pin for the router attachment. Drill straight through the front board. You can then use the router to make the cuts on the underside of the board.

To preserve the strength of the narrow mounting board, cut the actual driver holes after the cabinet is completely assembled, using the same pinhole that

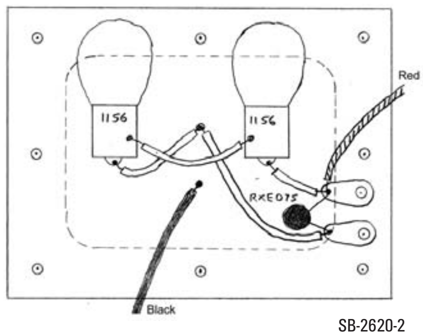


FIGURE 2: Terminal board pictorial.

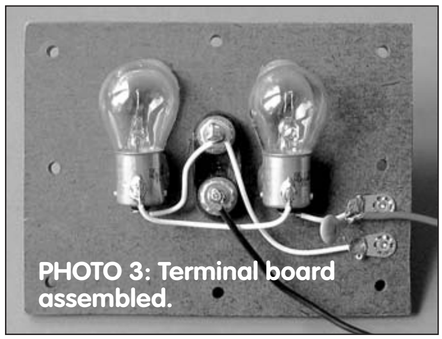


PHOTO 3: Terminal board assembled.

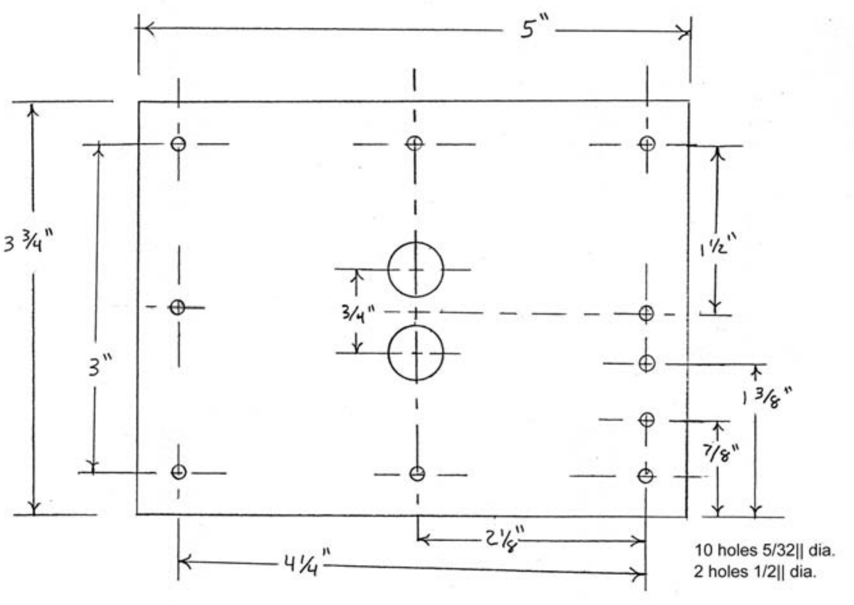


FIGURE 1: Terminal board layout.

PHOTO 4: The input terminal.

was used for the undercut. Be sure the size of the driver holes is exactly 3¹⁄₁₆″ in diameter, as there is not much space between the cabinet edge and the basket of the driver for the foam tape to provide a good seal. If the hole is smaller, the driver will not fit, and if it is larger, the foam will not have enough surface to seal.

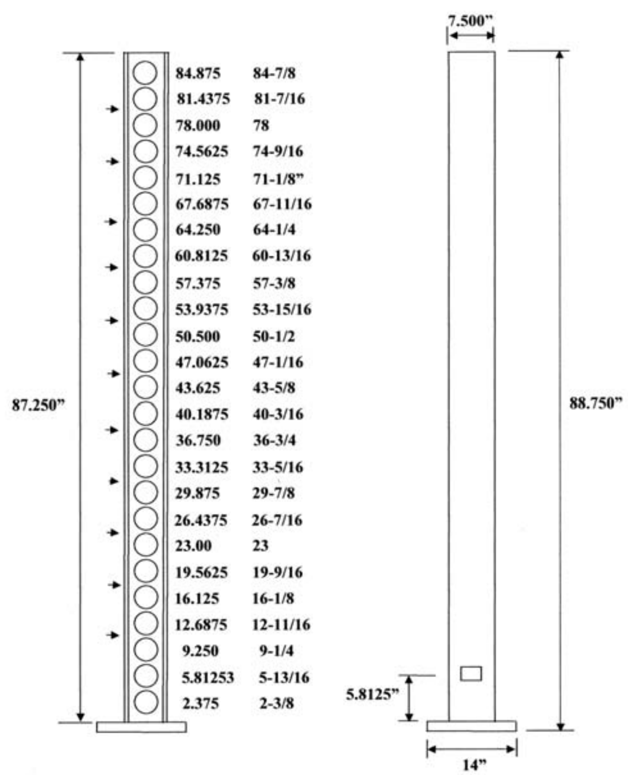


FIGURE 3: Column outline drawing. SB-2620-3

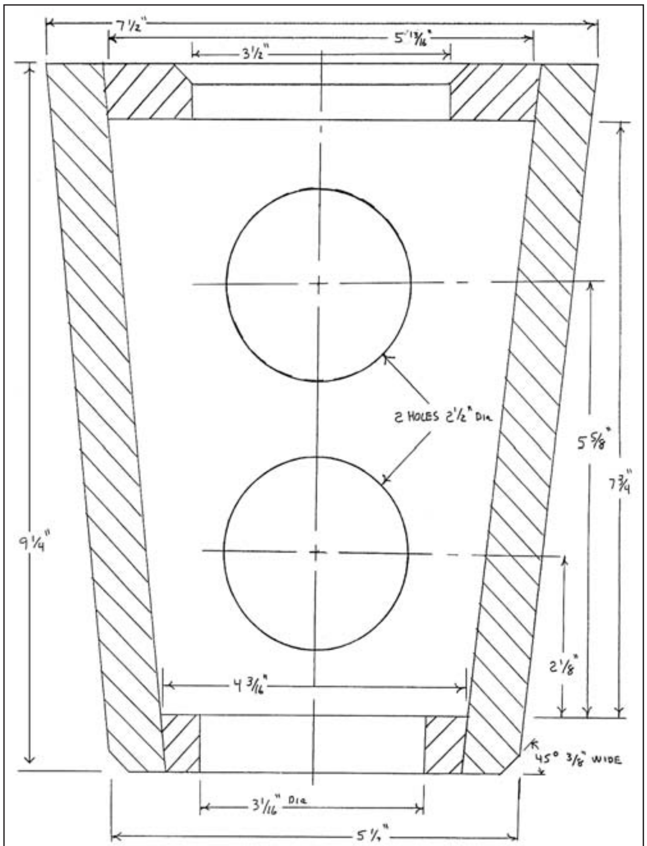


FIGURE 4: Cabinet cross-section. SB-2620-4

Figure 6 shows front and side cross-sections of the bottom of the cabinet, with two pieces of wood used at the top and bottom for strength, and two ⅜″ holes at the bottom for bolts that hold the base to the cabinet. A 2½″ wide cutout just behind the bottom driver allows room for the magnet structure, and the same cutout is made at the top of the cabinet for the top driver.

The construction of the base (**Fig. 7**) includes two ¾″ thick pieces of particleboard glued together and bolt heads countersunk in the bottom of the stand so they don't stick out below the bottom. The front and both sides are beveled at 45°. **Figure 7** shows the position of the cabinet where it is attached to the stand and the location of the holes for the bolts for the stand. These correspond to the same location in the bottom of the cabinet.

After filling and sanding, you can spray the cabinet with primer and several coats of white semi-gloss lacquer. I like white because I think the cabinets look less imposing, since they are so tall.

If the column is too tall for your ceiling height, you can make the cabinet one or two drivers shorter. Substitute an 8Ω 15W resistor for each driver that you remove from the system. Removing one or two drivers will not make a big difference, but bear in mind that the fewer drivers you use, the lower the power handling and the less bass capability there will be. In any case, the column must run from floor to ceiling in your room to be effective[3]. If your ceiling is higher than this column, you could use additional drivers, but these are not required, at least not for this design.

THE GRILLE

My first system did not include a grille. A grille frame 87¼″ long and 6″ wide could be easily bent or broken. I thought of a better solution by making the frame part of the cabinet and then adding a perforated plastic grille to it. I used my unfinished test box (**Photo 5**) to try out my idea. Some 12′-length extruded aluminum tile floor edging at Home Depot was perfect for this application. It even has holes for screws. I cut two short pieces and attached them to the test box ⅛″ from the edge of the drivers.

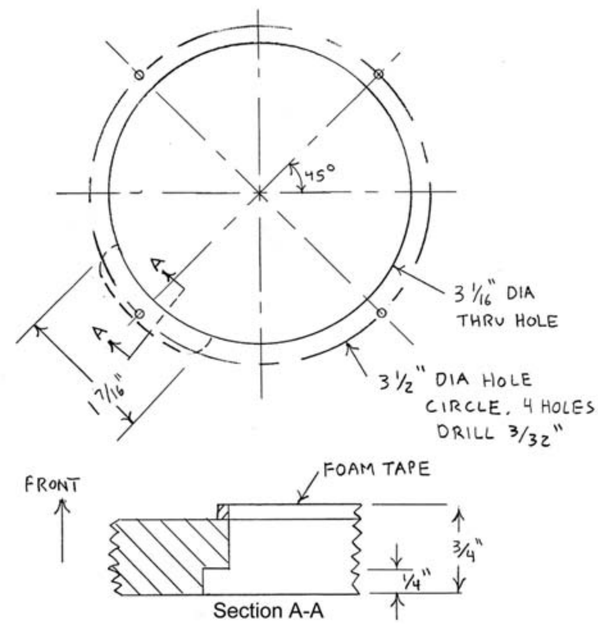


FIGURE 5: Driver hole.

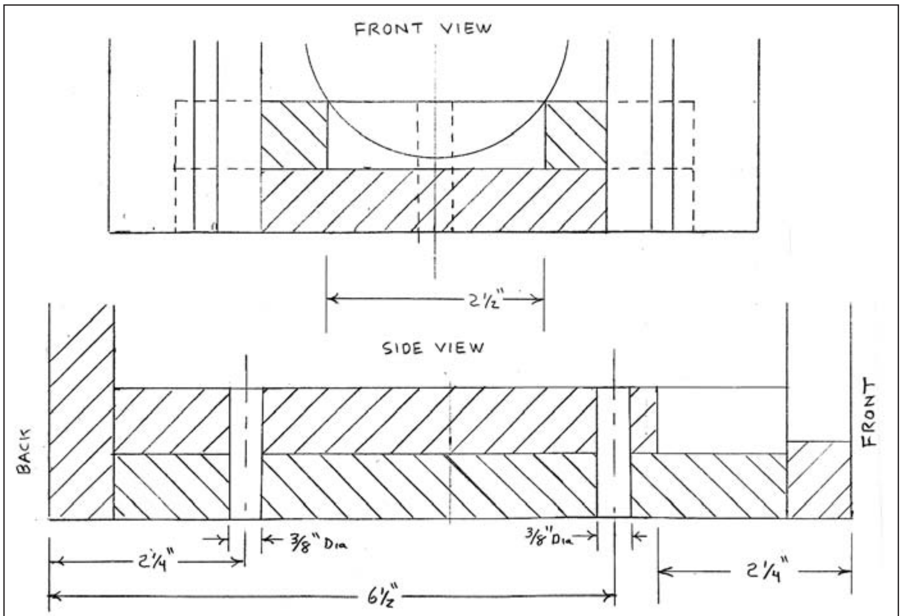


FIGURE 6: Cabinet bottom views.

I also found some perforated vinyl gutter guards that come in 36″ lengths. You'll need three for each system. The vinyl fits inside the edging on either side and bows out in the center by about ½″. The bow will allow room for the driver cone excursion. I cut off the rigid edges on both sides using a small bandsaw, being careful to keep the width at 4⅞″; otherwise, the guard will not bow out far enough or match the bow in the next section. I cut the width of each section so the holes lined up.

Although the vinyl looks very open, I calculated it to be only 58% open. Fortunately, measurements and listening show that the response is not affected. For the finished system, I sprayed the edging white to match the white cabinet, but, of course, you can choose any color to match the finish of your cabinet.

When installing the edging, I used a piece of ⅛″ Masonite about 1′ long to gauge the spacing between the edge of the driver basket and the inside edge of the extrusion. I then marked the location of the screw holes, punched and drilled with a 3⁄16″ drill. There are alternate oval and round holes in the edging. I used the round holes most of the time, but at the top and bottom of the column it might be necessary to drill an extra hole to keep the strip straight when you install the grille pieces.

FINAL ASSEMBLY

With the 3½″ R13 fiberglass left over from my first design, I made a template out of cardboard in a trapezoidal shape to match the inside dimensions of the cabinet. I then easily compressed the fiberglass and cut it with a utility knife (**Photo 6**). The dimensions are 7″ high—4¼″ at the top and 5¾″ at the bottom. I used a total of 38 pieces, with the paper backing removed, placing three pieces between each brace section and five pieces in the bottom section.

Although you can drill the driver screw holes ahead of time, I prefer to set the cabinet on its back and equally space all the drivers. Then, I mark the holes with a pencil and punch and drill them with a 3⁄32″ drill.

The preferred screws are #6 × ¾″ deep thread, pan and black. Be sure all the mounting screws are in place before testing the system. Initially, I used only two screws on opposite sides for testing and it caused the voice coil to rub. The basket is plastic and will bend a little with only two screws holding it, so I added the two remaining screws, and it worked fine.

WIRING

I wired the drivers in series-parallel (**Fig. 8**). There are five groups with five in se-


PHOTO 5: The grille construction.


PHOTO 6: Fiberglass template.


PHOTO 7: Lowest driver cutout showing bolts.

ries in each group. This comes out to the same 8Ω impedance as a single driver and is satisfactory for most amplifiers.

All the drivers are in the same electrical phase. I prefer 8Ω instead of 4Ω. The connecting wires from the amplifier to the speaker can be lighter gauge and, therefore, less costly. Terminal connections at the amplifier and speaker are less critical for contact resistance.

The wiring is shown in **Fig. 9**. I used #18 stranded hookup wires to connect all the drivers. You could use heavier wire, but the resistance of this wire is negligible compared to the speaker resistance. Wires from the positive input are red, the black are negative, and 8″ white wires connect the drivers in series. All connections are soldered.

For a single wire to a driver, I twisted ½″ length strands together and tinned with solder. Then, I bent ¼″ of this 180° with thin nose pliers, allowing me to fit the U-shape around the driver terminal, crimp it in place, and then solder. When connecting two wires to the same terminal, you can twist the pair together, tin, bend, crimp, and solder in the same way.

I used two ⅜″ × 3½″ plated bolts—along with a ⅜″ lock washer and ⅞″ washer under each bolt head and nut—(**Photo 7**) to attach the base to the cabinet. To seal between the drivers and the cabinet, I used ⅛″ thick foam with adhesive on one side, cut into strips ⅛″ wide and attached to the edge of each driver hole. The foam compresses very easily and forms a good seal.

After wiring the system, you can make

TABLE 1: DRIVER PARAMETERS.

Nominal impedance: 8Ω
Voice coil resistance: 6.3Ω
Operating power: 22.5W
Sensitivity: 82.5dB re: 1W at 1m
Frequency range: 200Hz to 15kHz
Free air resonance: 87Hz
Voice coil diameter: 20mm
Voice coil height: 8.7mm
Air gap height: 4mm
Voice coil inductance: 0.12mH
Effective diaphragm area: 38cm^2
Moving mass: 2.2g
Magnet weight: 105g
Force factor (BL): 2.95
Vas: 3.1 ltr
Qms: 3.08
Qes: 0.87
Qts: 0.68
Net weight: 420g or about 14.8 ounces

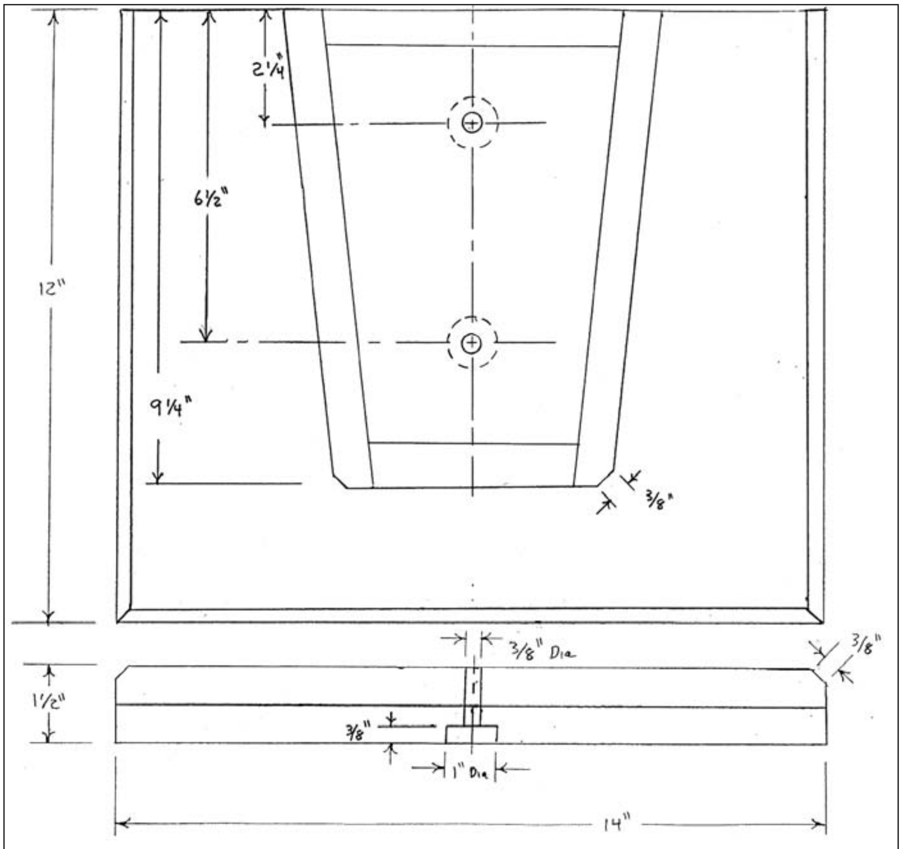


FIGURE 7: The stand construction. SB-2620-7

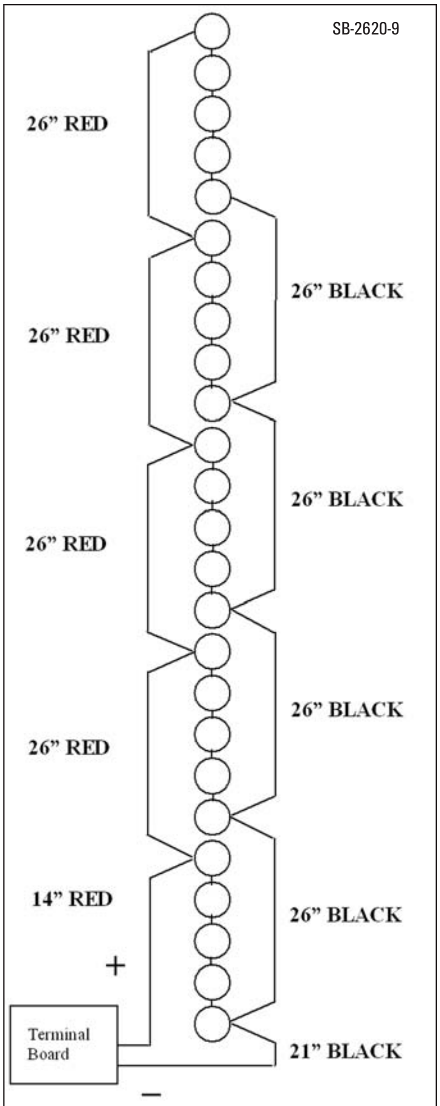


FIGURE 9: Wiring diagram.

a final polarity check by connecting a 6V battery to the input terminals. With the positive lead connected to the red terminal and touching the black to the negative, all of the cones should move outward.

EQUALIZATION

Equalization is used to compensate for the response at low frequencies[4]. The equalization curve is significantly different from the first system and must be changed to match the new drivers. In the cabinet, the drivers go down to about 150Hz and then output decreases, reaching a slope of 12dB/octave. I initially made the bass equalization curve to be close to the opposite of this rolloff, but later modified this slightly for a listening correction. Compensation up to 20dB is used.

I again used a McIntosh MQ107 equalizer (**Photo 8**) for the response adjustments. The equalizer was made from 1981 to 1993 and is often available through eBay or other sales places. It has low-frequency and high-frequency compensation controls plus seven adjustable filters, which can be ⅓ octave or 1 octave width and each boost or cut 12dB in the ⅓ octave mode or 15dB in the octave mode.

You can select the center frequen-

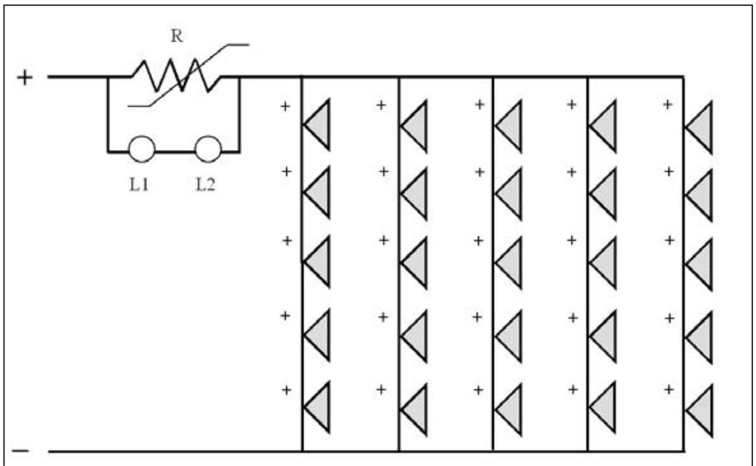


FIGURE 8: Schematic. SB-2620-8

TABLE 2 CAPACITOR VALUES.

	Narrow Filter		Wide Filter	
Frequency	Capacitor A	Cap. B	Capacitor A	Cap. B
20	1.0	.39	–	–
25	1.0	.27	4.7	.056
31.5	.82	.22	–	–
40	.56	.18	–	–
50	.56	.12	2.2	.022
63	.39	.12	–	–
80	.22	.12	–	–
100	.22	.082	1.0	.012
125	.18	.056	–	–
160	.12	.056	–	–
200	.12	.039	.56	.0082
250	.12	.022	–	–
315	.082	.022	–	–
400	.056	.018	–	–
500	.056	.012	–	
630	.039	.012	–	–
800	.022	.012	–	–
1000	.022	.0082	–	–

cy and filter width by plugging in two small capacitors for each filter. **Table 2** shows capacitor values (in microfarads) for center frequencies and width from 20Hz to 1kHz.

You can insert capacitors in the corresponding A and B slots in the equalizer. Three holes are available for each capacitor, but only two holes are actually used, depending on the lead spacing of the capacitors. Each filter has a control with a center detent that can adjust the amplitude to boost or cut. The center position is electrically flat.

A compartment box with capacitors originally came with my equalizer but may not be included with yours. Recommended capacitors, which you can purchase from Mouser Electronics, are metallized polyester or equivalent for values up to 0.56µF. Use non-polar electrolytic capacitors for higher values. Capacitors must have a voltage rating of 50V or higher.

Six of the seven filters are used for each channel. **Figure 10** shows the frequencies, capacitors, and control settings that I used. The low-frequency compensation controls are not used and are set to one. The high-frequency compensation controls are set at maximum. The same equalizer settings are used in both channels.

You can use additional frequencies by calculating new capacitor values in **Table 3**. Note that "stock" values are normally the only ones available. If, for example, you calculate a capacitor value of 0.095µF, the closest value available will be 0.1µF.

You can use other equalizers as long as the initial adjustments can accurately follow the response curve in **Fig. 13**. The closer the initial equalization matches the opposite of the driver rolloff, the better the low frequencies will sound. However, you may decide to go a step further and compensate for other influences, such as room resonances. High-frequency equalization is also needed to produce a smooth and very flat response out to 15kHz and beyond where the response gradually decreases. You can adjust the highs to suit your preference with these controls at the right side of the equalizer.

TABLE 3: CAPACITOR CALCULATION.

Capacitor	Narrow Filter	Wide Filter
Capacitor A	23.5 divided by frequency in Hz	118 divided by frequency in Hz
Capacitor B	7.2 divided by frequency in Hz	1.44 divided by frequency in Hz

MEASUREMENTS

I have driven both speakers with random-phased pink noise for the response curve in **Fig. 11**. I used a Bruel & Kjaer 4133 condenser microphone that has very flat response out to 40kHz. The output goes to a General Radio ⅓ octave analyzer and chart recorder

PHOTO 8: The McIntosh MQ107 equalizer, with cover removed.

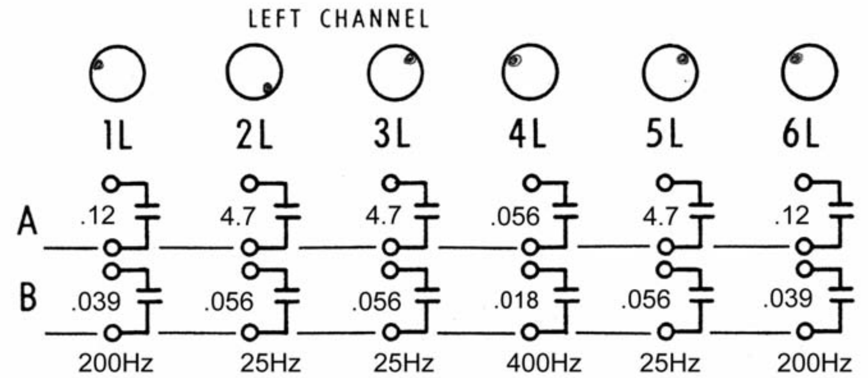


FIGURE 10: MQ107 filter settings. SB-2620-10

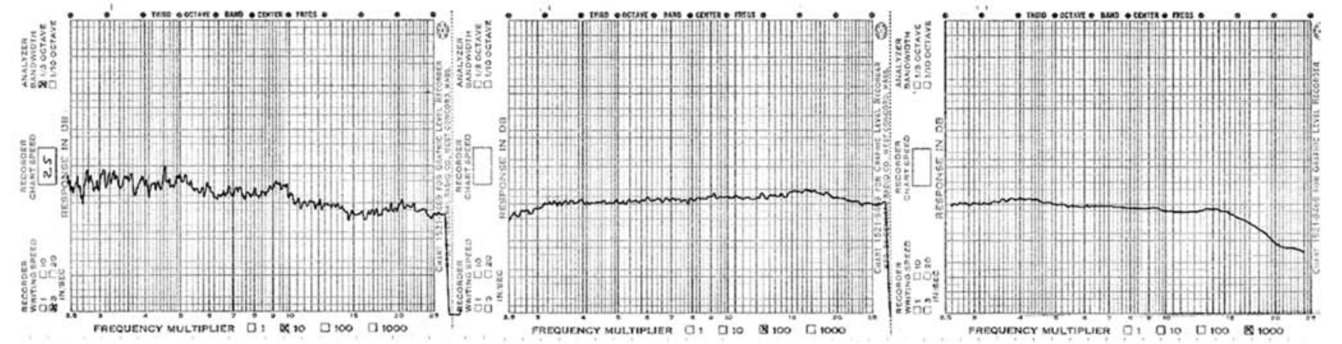


FIGURE 11: Combined system response. Each major division on the graph is 10dB. The first section at the left is 25–250Hz, the center section is 250–2500Hz, and the right section is 2500–25kHz. SB-2620-11

that automatically sweeps from 25Hz to 25kHz.

I located the microphone at the listening area about 9´ from each speaker and 4´ from the floor. The response in my room is very smooth and extends beyond 15kHz. The low-frequency response is slightly higher than the rest of the range, which many people seem to prefer[5], including me. You can always adjust this to suit your own preference.

When multiple drivers are used to cover the same frequency range, you can sometimes observe an effect called "comb filtering," which is caused by reinforcement and cancellation of the sound from drivers that are different distances from the microphone. It is most easily shown with a response curve using a sine wave sweep for the measurement. However, you hear music differently than what is indicated by a pure tone graph. This is due to the masking effect and is more like listening with multiple ⅓ octave filters or perhaps wider[6,7]. When making a listening test, a constant wideband sound source, such as the pink noise that I have used, makes any changes easier to hear.

One of the major factors is driver spacing. So when does driver spacing in a long column become noticeable compared to one, long, continuous column such as a ribbon? The closer the drivers are, the less perceptible any variation in sound is. A spacing of 1´ between drivers is obviously too large, and if you put your ear 6˝ from such a column and move up and down, you will be able to clearly hear individual sources, even far back from the column.

However, the closely spaced drivers used in this column perform far better. By the time you move back 3 to 4´ or more, the sound from the column appears to be uniform. In addition, with music, you can come as close as 1´ and the sound still appears to be continuous.

The response of a single driver in a small test box is shown by the curve in **Fig. 12**. The single driver response does not account for other factors when all of the drivers are working together in a column, near a wall, or for room resonances.

Mutual coupling between closely spaced drivers at low frequencies can increase bass output by several dB. Although the effect is most noticeable when the drivers are all located in close proximity to each other, such as a square or circle, some gain may also be realized by orienting the drivers in a column. In addition, with the system near the wall, it is essentially radiating into a solid angle of 180° instead of 360°, giving 3dB of gain.

Room resonances certainly have an important influence on the response as well, particularly back at the listening position. In making the final adjustments, I needed to make some minor corrections, mostly in the voice area. It was only a couple of dB, but it made a dramatic improvement. Also, my expertise in adjusting the filters in the equalizer has improved. Altogether, the sum total of these factors influenced the final equalization curve (**Fig. 13**), which is noticeably different for the new drivers.

Equalization below 30Hz is reduced to prevent excessive infrasonic power from being delivered to the system. However, the acoustic response in the room is still going strong at 25Hz.

The impedance curve for the entire system is shown in **Fig. 14**. The impedance for the new drivers, having the copper shorting cylinder, increases very little at the higher frequencies compared to drivers without the copper cylinder.

SAFETY FIRST

Each system weighs 81 lbs. If you place the system on a soft carpet with thick padding underneath, you should add a set of four floor spikes (available from Parts Express) to make it more stable. You can drill holes in the bottom of the thick base near the corners for the spikes, which are thin and go through

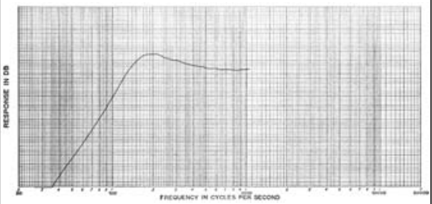


FIGURE 12: Response of a single driver. Each major division is 5dB. SB-2620-12

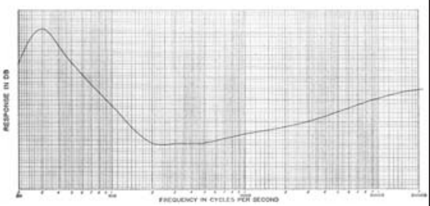


FIGURE 13: Equalizer response. Each major division is 5dB. The measurement is from 20Hz to 20kHz. SB-2620-13

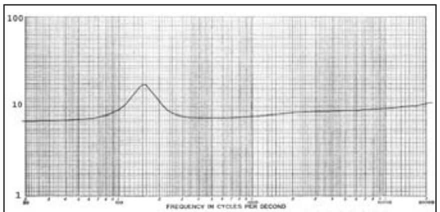


FIGURE 14: Impedance curve. The vertical scale is in ohms and is shown on a dB scale of 5dB per major division. The measurement is from 20Hz to 20kHz. System resonance is 155Hz. SB-2620-14

the carpet without damaging it. You could also make a larger base.

You can achieve further stabilization by anchoring the top of the cabinet firmly to the wall using one of two different friction lid supports available from Home Depot. The first is made by Ives-Schlage, H. B. Ives, New Haven, CT 06511 (www.schlagelock.com). Two screws anchor the support to the wall and two more screws anchor the support to the top of the cabinet. The support is adjustable for the angle and distance of the cabinet to the wall.

The second is a lid support made by The Stanley Works, New Britain, CT 06053 (www.stanleyworks.com). Two screws anchor the bracket to the wall and one screw anchors the bracket to the top of the cabinet. Again, the support is adjustable.

USING THE SYSTEM

A typical listening arrangement forms an equilateral triangle between the listener and each speaker. In my living room (**Photo 1**) the speakers are about 8´ apart, so my best listening spot, although not critical, is around 8 or 9´ from each speaker. The systems are about 3˝ from the wall. High-frequency dispersion is wider with these drivers, and I prefer listening without having the systems toed in toward the listening area.

Typically, only 2 or 3W are all that is needed for a good listening level and most program material. However, it is always good to have a power reserve for program peaks. More important, deep bass will require more power because of the equalization requirements. I have been using a power amplifier that can deliver 260W per channel into 8Ω. It is probably much more than is needed, but I like to have a reserve of power available, just in case.

You will notice that as you approach the columns, they do not become as loud as an ordinary system. You can listen as close as 3´ from the wall and still get excellent stereo sound, including spaciousness. So you can also use these systems in a fairly small room, where you can't avoid being close to them.

CONCLUSION

The new system was a financial risk for me because I was unsure whether the finished product would justify the cost. As it turned out, in my quest for better sound, the improvement was very significant and well worth the expense.

After adjusting for what you see versus the excellent bass that you hear, you'll discover more benefits. The response of the system is very smooth, and is free from any resonances, peaks, or dips. I found the coherence—when an entire orchestra is playing—to be amazing. You may realize, perhaps for the first time, that the stereo image, depth, and spaciousness provided by this system is totally mesmerizing, and you are aware only of what the artist has produced.

The superior imaging of the system lets you locate the entire characteristic sound of each instrument more accurately in space. When all the instruments are playing, you experience their complete sound at their individual locations, as opposed to becoming all muddied up with each other. This is true whether you are listening to the softest or loudest passages. Details are incredibly realistic and the impact of bass drums is surprising.

The column is very directional vertically, so floor and ceiling reflections that would normally interfere with the direct radiation are effectively eliminated. Because this long column runs from floor to ceiling, the sound will always be the same whether you are standing, sitting, or at any other height. This is a significant advantage over traditional systems that not only radiate sound as much vertically as horizontally but also exhibit response changes with listening height. At the same time, the horizontal dispersion provided by the smaller drivers in the column ensures a wide radiation angle over most of the frequency range.

You will also experience a completely effortless sound throughout the entire frequency range, due to the large, effective cone area that couples so well to the air. Normally, the area available for mids and highs in most systems is very small, and single drivers are easily driven into distortion at higher levels. Not only is the sound from this column system effortless, but it also fills the room from top to bottom and appears to play louder than a typical point-source system. When you combine all of these impressions, it goes beyond overwhelming. **aX**

***Roger Russell** has a degree in electrical engineering from Rensselaer Polytechnic Institute, Troy, NY. He has written several articles about loudspeaker design, testing, and other audio devices. Several patents have been awarded for some of his designs. Retired now, he was a member of the Audio Engineering Society and the International Society for General Semantics for many years. Besides his love of stereo sound and music, his other interests range from photography to mystery clocks and science fiction. He has a web page at www.roger-russell.com.*

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2. "Performance of Series-Parallel Speaker Arrays," James F. Novak, *Audio*, 9/61, p. 23.

3. Patent number 4,267,405 Stereo Speaker System for Creating Stereo Images awarded to Roger Russell, May 12, 1981. This patent describes closely spaced drivers arranged in a long column.

4. Patent number 3,715,501 Loudspeaker System awarded to Roger Russell, February 6, 1973. This patent describes the use of equalization to compensate for the low-frequency rolloff of a loudspeaker system.

5. "Speaker Evaluation: Ear or Machine? Part 2," Roger Russell, *The Audio Amateur*, 1/75.

6. *Acoustical Engineering*, Harry F. Olson, Ph.D., Van Nostrand Company, Inc. 1957, pp. 572–573.

7 *Acoustics Handbook—Application Note 100*, Hewlett-Packard Company, 1501 Page Mill Road, Palo Alto, CA, Nov. 1968, pp. 24-25.

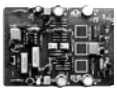
PARTS LIST AND SUPPLIERS

R: Polyswitch RXE-075, 60V and 40A maximum
L1, L2: Incandescent automotive lamps GE 1156
D1 to D25: Vifa model TG9FSD10-08 wide range driver

Parts Express
725 Pleasant Valley Drive
Springboro, OH 45066-1158
1-800-338-0531 *www.partsexpress.com*

Mouser Electronics
1000 N. Main Street
Mansfield, TX 76063
1-800-346-6873 *www.mouser.com*

Vifa drivers
Haig Audio
1-248-425-1027 *kenhaig@earthlink.net*



A First-Rate Audio Test Instrument

Part 2: *Construction*

Back

Part 2 of this series completes the trio, adding a super-versatile dummy load to your bench.

By Todd Sharp

With modifications to the sine-wave generator, voltmeter, and respective power supplies completed, it's now time to strip the HP 353a patch panel down to its birthday suit, give it a face-lift, and replace its insides with a selectable impedance dummy load (**Photo 1**). Actually, it's much more than a simple dummy load. In addition to providing various load options, you can also add a few other features and a simple attenuator network. Depending on how you set the switches (S1 and S2), you may select the following load options: shorted, open, 4Ω (400W), 8Ω (200W), 16Ω (400W), speaker only.

The 4, 8, 16Ω positions also route the device under test (DUT) signal through a simple attenuator (R9 and R10), which feeds the external speaker jack (J2). You supply the speaker. This provides a "listen through the load" function so that you may load your DUT at high power into its rated load while listening through to the speaker that you have attached to the speaker jack. You can adjust the speaker level (R10) to suit.

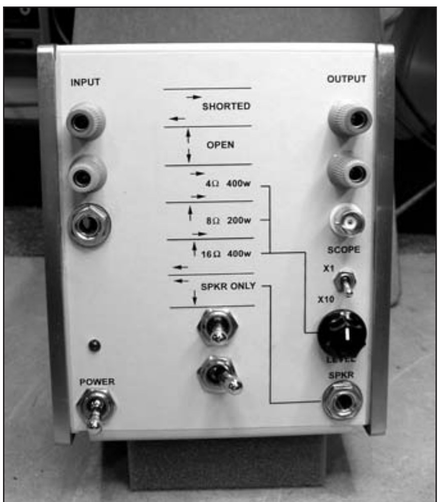


PHOTO 1: Load box front.

This feature offers a negligible amount of error to the actual load impedance, but if you care to modify the pot as I have done, when fully counterclockwise (CCW), it is virtually out of the circuit and introduces no error while also providing a silent mode. You must open up the pot and remove the wafer. With a sharp knife, cut a small separation in the carbon deposited film as shown in **Photo 2**. This opens the circuit when the pot is all the way down and thus lifts the speaker connection completely.

PHOTO 2: Minor pot surgery.

The attenuated signal presented to your speaker will contain a little distortion, but then this function doesn't require high fidelity. This is very handy for troubleshooting the intermittent and "it only cuts out after it's been on a while" kind of troubles as you can listen to, as well as view, high power signals through the load without blowing your head off. This is a virtual 400W signal tracer—at reasonable ear volume.

The other features are: input connections (BP2, BP4, J1), AC voltmeter output terminals (BP1, BP3), and a scope output BNC connector with a built-in ×10 attenuator (SW3, R11, C1). In the SPKR ONLY position, the DUT bypasses straight through to your connected speaker while still offering AC voltmeter and scope monitoring. Be careful with this setting if you are running signal through the dummy load at high power and suddenly switch to SPKR ONLY. If you want to see the cat jump 18′ sideways, this does the trick. Of course, it could also blow your speaker!

Monitoring high power waveforms with your oscilloscope normally requires an attenuator probe, because this signal will exceed the limit of your scope's input. This is commonly called ×10 because when you read your scope's Volts/Div you need to mentally multiply by 10 for the true reading. I built in the ×10 function so that you can simply attach a BNC jumper cable between the load box and your oscilloscope—no probes necessary.

Another feature provided for is a ×10 high frequency compensation adjust (C1). In practice I rarely use my scope for such accuracy of measurement when viewing audio waveforms, but more for an overall picture and easy indicator of waveform distortions. This is what the RMS ACVM is all about and why such a piece is included in the set. The 403b offers a more accurate indication of level, not to mention a dB reference for frequency response measurements. You can also make simple power measurements with the set by applying the power formula: V^2/R (RMS voltage squared (times itself) divided by R (Rload in ohms) = average power.

Note that Ra and Rb are shown as 8Ω 200W resistors. You are actually going to construct these two separate 8Ω 200W resistor networks by wiring up four 8Ω 50W (R1-R4) and (R5-R8) resistors in series/parallel (**Fig. 2**).

FAN CIRCUIT

Finally, you add a fan to dissipate heat. Even though the load resistors have heat fins and are mounted on finned aluminum heatsinks, this does end up being a tightly packed little box with virtually no air flow—so add a tiny but useful 12V fan to pull the heat out and away.

The fan is controlled by a simple temp sensor LM35DT (IC1) powered by a nifty little 12V switching supply (PS Module). You could go to greater lengths with more sensitive circuitry, but this simple circuit does the job just fine.

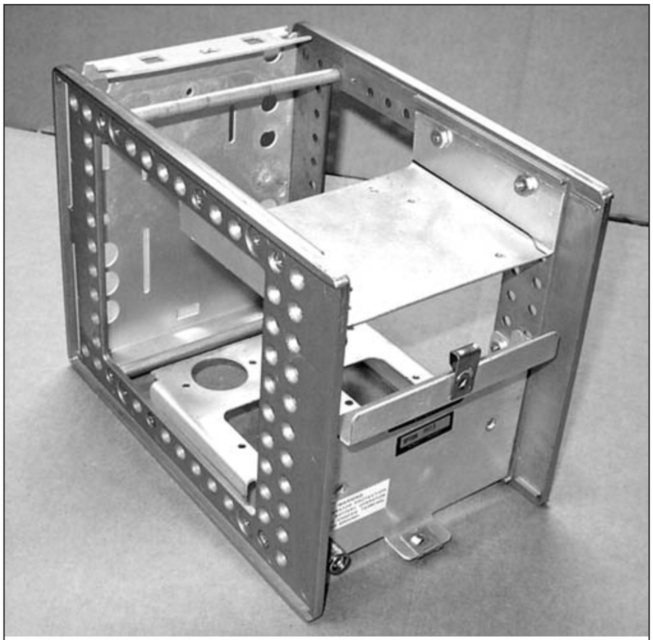


PHOTO 3: Stripped chassis showing bracket location.

As with the 204c, I added a power switch (SW4) to the front panel (FP) for this supply along with an LED (CR1). This LED has two functions: it provides a constant load to the 12V supply, while showing on/off status. Past experiences have taught me never to turn on a switching supply without a load attached. Perhaps you have also learned much about electronics by blowing things up! Anyhow, I used a green LED that draws about 25mA. The fan kicks in when the resistors reach about 100° F (**Fig. 2**).

CONSTRUCTION

Start by completely disassembling your 353a—taking notice of how it comes apart so you can put it back together later! The only electronic parts you are going to salvage here are the binding posts—that is, if your 353 comes with binding posts. I have noticed that some have TT type jacks instead. We built two of these here at our shop, and found it necessary to accommodate a slightly different front panel layout for the two different (binding post vs TT jack) types. I have made both panel layouts available in pdf form to print on sticker paper for the cosmetically conscious.

You will also be taking care to isolate every connector in and out of the load box from chassis ground to alleviate grounding complications. These little HP boxes, which are actually quite friendly to work with, come apart and go back together almost like erector sets. There are multiple holes to run your screws through so that you might move the inner brackets and support bars wherever you want them to be.

The picture that you are looking at in **Photo 3**, for the sake of clarity shows an upside-down view of the frame of the load box. This takes advantage of the existing panel holes and puts the fan and heated components at the top of the enclosure where the heat will collect. I promise this will make more sense to you later when you put it all back together.

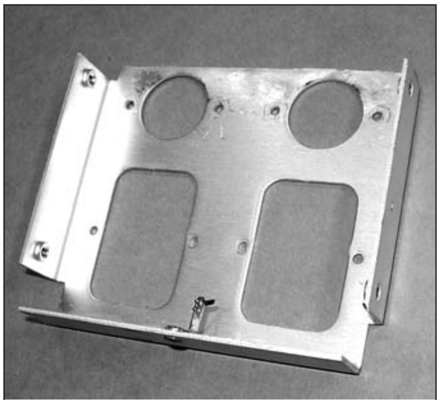


PHOTO 4: Bracket after cut.

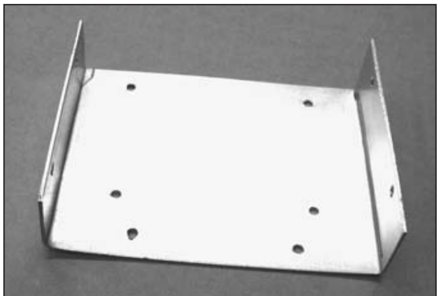


PHOTO 5: Fan supply and temperature sensor circuit mounting bracket.

RESISTOR/HEATSINK MOUNTING BRACKET

Locate the aluminum bracket with the four transformers mounted to it. This will become your resistor/heatsink mounting bracket. Start by removing the transformers, wiring harnesses, and the terminal strip. You will then need to cut one piece off this bracket (**Photo 5**).

The folded edge that had the terminal strip mounted to it and the wiring harnesses going through it must go—in order to make room for some structural changes. The opposite edge with the ground tab remains. A standard hacksaw will do just fine. A bench vise would also come in handy as well as a flat file to smooth edges (**Photo 4**).

FAN SUPPLY AND TEMP SENSOR CIRCUIT MOUNTING BRACKET

You will need to fabricate an additional mounting plate to accommodate your power supply and temp sensing circuitry (**Photo 5**). I made this with some 1/16″ aluminum sheet.

I made cuts with a hacksaw and smoothed edges with a flat file. The cut piece measures 7.0″ × 3.5″. Mark off a line at exactly 1.25″ from each edge (the 7″ length) and initiate a 90° fold at each line. This should end up being a 4.5″ × 3.5″ plate with a 1.25″ "ear" at each end of the 4.5″ length.

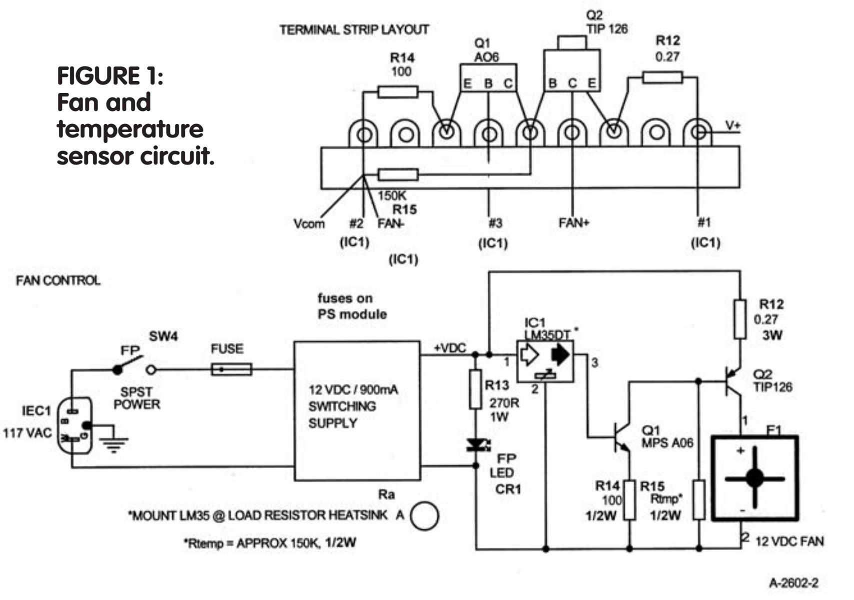
If you are not accustomed to working metal like this, it is much easier than you might think. 1⁄16″ aluminum is very easy to cut, drill, and fold with a standard 3″ shop vise by locking the jaws down just a smidge below where you want the fold to break and simply bending it over the top of the vise. Use the resistor/heatsink bracket as a sizing template. Repeat for the other side and you're done, except for the various screw holes you will need to drill for affixing to the HP side rails. Wedge your mounting bracket into place and mark off screw holes for mounting to the side rails. Affix with 6⁄32 hardware.

The next step involves fixing the switching supply board to the bracket you just made. If you are using the Cincon board from Mouser, then you will need to figure how to mount it.

For some reason they manufacture these open frame supply boards without any screw holes or brackets for mounting, and there is no room on the board to drill mount holes. There are simply four turrets sticking out of the bottom that happen to be the input and output connections. I stuck these through a 3.25″ × 2.5″ piece of vector perfboard, and soldered solid wire wraps to the bottom side and then ran them to push in style micro clips (**Photo 6**).

This not only breaks out the in/out points to convenient wiring clips, but it

FIGURE 1: Fan and temperature sensor circuit.



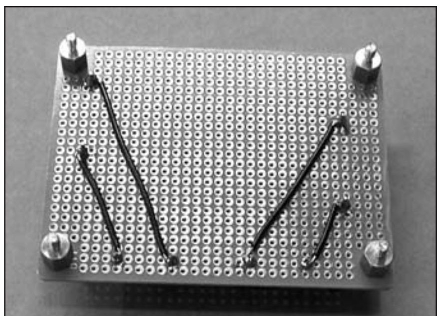


PHOTO 6: Board with wire wraps.

also provides a sturdy mounting to the perfboard, which does have screw holes! Use four 4/40 (0.25″ height) standoffs to mount this board to the bracket. Place your board over the bracket and mark off where you want the screws to come through to mate to the standoffs. Drill these mounting holes and mount your standoffs.

Next mark screw holes for your terminal strip and mount that using 6/32 hardware (**Photo 7**). Wire up the components as shown in **Fig. 1**.

FAN AND IEC CONNECTOR CUTOUT

Next, grab the rear panel section. If you don't own a 3″ circular chassis punch, then you must figure out a way to draw a 3″ diameter hole centered in the back panel as shown in **Photo 8**. Be sure to allow room for the IEC plug in the next step. Drill, cut, file, and cuss your way to a fairly neat 3″ hole. Mark off screw holes, drill and mount the 60mm fan—taking care to note which way the air will flow. You want to pull hot air out of the box, not pull crud *into* the box.

I drilled out four mount holes but used only three screws. I used the remaining hole to run the fan wires through. Take the fan out and set it aside for the moment.

The next trick is to mount an IEC connector next to the fan. Mark off a semi-rectangle to accommodate the IEC connector. Drill and file your way to a suitable rectangle. Mark and drill out the two screw holes to affix the connector to the panel. Use 6/32 hardware.

Attach a 12″ length of black wire (#20 stranded) to the hot terminal of the AC connector and a 9″ length of white (#20 stranded) wire to the neutral terminal of the AC connector. Be sure to insulate these connections with heat-shrink tubing. Attach a 4″ green (#18 wire) to the middle ground terminal and insulate as well. The black wire will go to your front panel power switch, the white to one of the AC inputs on the power supply board, and the green to the chassis tab underneath your heatsinks.

Notice how you attach the heatsinks to the mounting bracket. If you wedge 6/32 screws in at just the right place in the heat fins, you can use already existing mounting holes (**Photo 9**).

Notice the little 90° bracket added at approximately the center of the flat side of the heatsink for stability. You will need to drill and thread a 4/40 screw hole for this bracket. Read ahead a bit further for tips on tapping the heatsink for 4/40 threaded screws. Once you have the heatsink mounting figured out, remove the heatsinks from the bracket and lay out four of the 8Ω 50W resistors on each to see how they will fit.

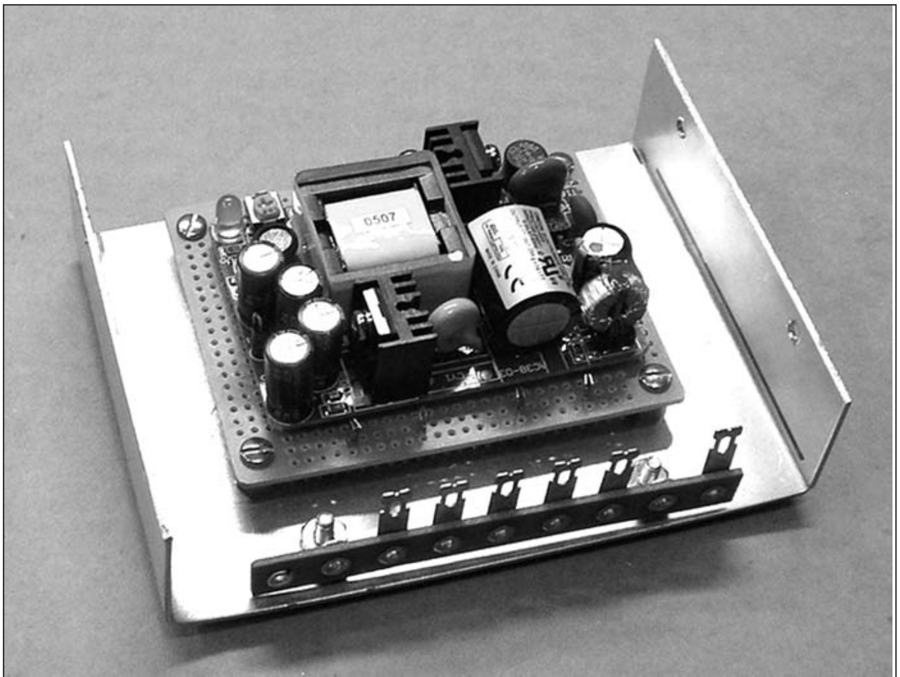


PHOTO 7: Mounting board to bracket.

MOUNT RESISTORS TO HEATSINKS

You will make up two resistor network blocks, Ra and Rb, each consisting of four 8Ω 50W resistors wired in series parallel. Refer to **Fig. 2**. Notice that Ra and Rb are indicated on the schematic as 8Ω 200W resistors. See **Photo 10** for mounting layout.

Note that these particular heatsinks have a few nubs protruding. You can work around them if you place the finned resistors just right. Place the resistors and mark screw holes. Take extra care to mark, center punch, and drill your holes in exactly the right places. Drill your screw holes with a 7⁄64 drill bit.

Once you have the holes drilled, insert a 4⁄40 threading tap and tap the holes. Use caution here and use some cutting oil. A tapping die bit will snap off mid turn if you force it even a little when it wants to stop. If you feel it resist, back up a little to clear the cut thread and try again until it continues to cut thread with as little force as possible. The items you need, along with a tap "T" handle, should be available at any good hardware store. Buy an extra 4⁄40 tapping bit in case you break the first bit (**Photo 9**).

Mount the resistors on the heatsinks, four on each using 4⁄40 screws. Apply some thermal grease before mounting. Wire the resistors together, but just tack-solder the ends, where you will later attach hookup wires. I used #16 solid to join the resistors.

The LM35 sensor (TO220 type package) will need to go down first, underneath the joined connection of one of the resistor network blocks. Note how the sensor is located on the right resistor block with its legs pointed out. This will be your "8Ω" load resistor.

Notice on the schematic that the 4 and 16Ω settings use both resistor networks (400W), while the 8Ω setting uses only one block (200W). Make sure

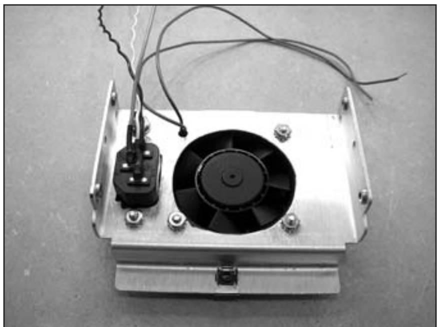


PHOTO 8: 3″ cutout.

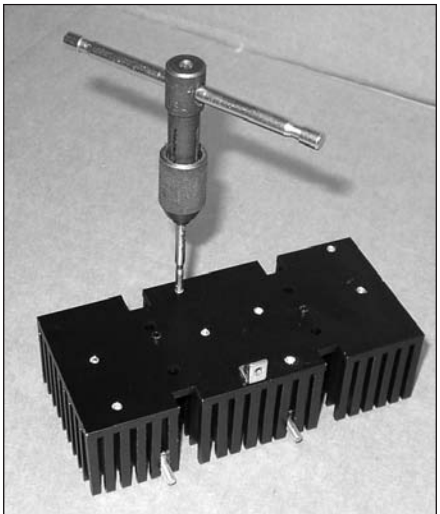


PHOTO 9: Attaching heatsinks to the mounting bracket.

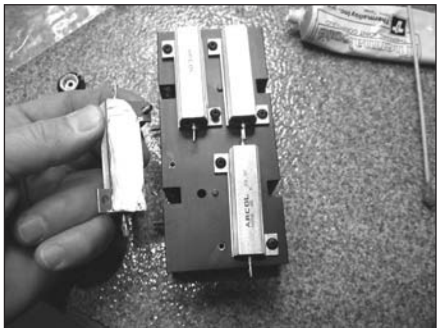


PHOTO 10: Mounting resistors to heatsink.

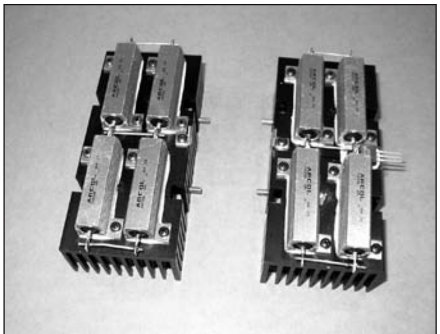


PHOTO 11: Load resistor.

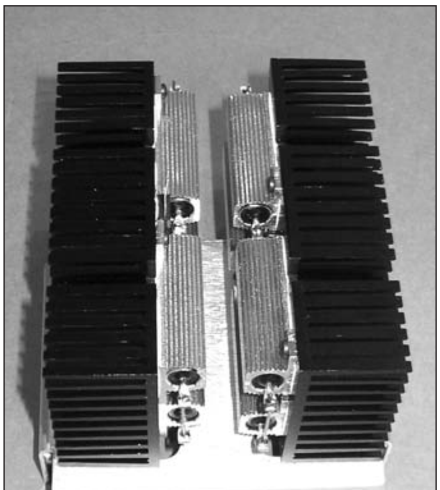
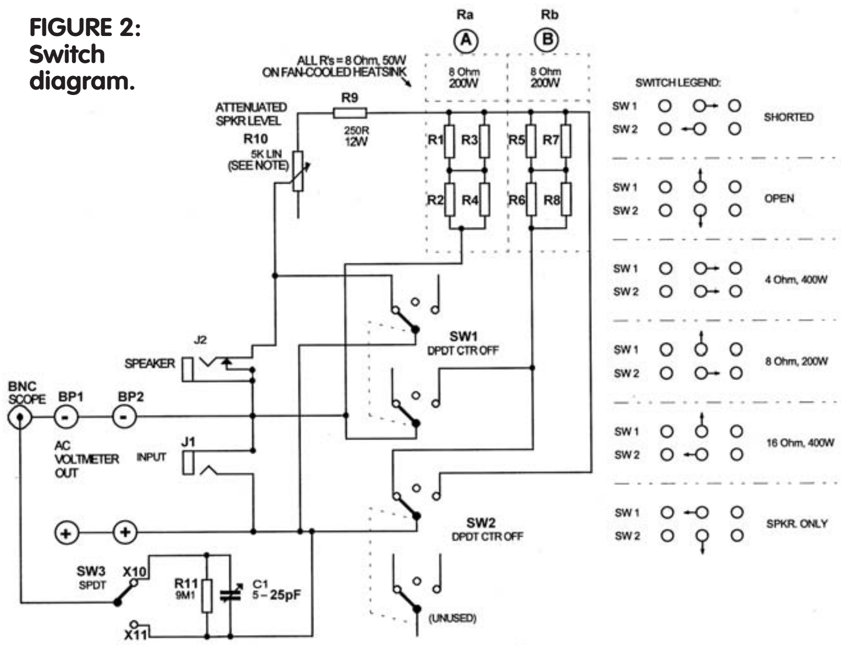


PHOTO 12: Chassis bracket.

FIGURE 2: Switch diagram.



that the one with the sensor is your 8Ω load resistor (**Photo 10**). Use the same drill and tap materials for the LM35 temp sensor mounting. Use a mica insulator, insulated shoulder washer, and thermal compound.

Once you have affixed the eight resistors and the LM35 sensor to the heatsinks (**Photo 11**), you can mount them to the chassis bracket (**Photo 12**).

X10 TRIM ADJUST CAP

Affix the ×10 hf trimmer cap (C1) to side rail. You will need to drill additional mounting holes in the side rail (**Photo 13**) and then countersink the screw heads with a slightly larger diameter drill bit in order to affix your adjustable trimmer in place. Attach two leads about 3″ in length and let them dangle until you get ready to wire up the front panel. Make sure the trimmer is isolated from the chassis. Drill an access hole in the side cover later for easy adjustment access.

FRONT PANEL

Once you have the front panel disassembled completely, download and print out a plain paper copy of the front panel template layout and use this to line up and center-punch the "X" points for the three new holes you'll be drilling in the front panel according to the template designations. Trim this print out and apply a little transparent tape to hold it in place. As far as I know, there are two types of HP 353a patch panels: one with binding post in and outs, and the other with TT type patch plugs. Use this layout for the binding post type: www.amprepair.com/LBfrontpanelBP.zip. Use this for the TT style: http://www.amprepair.com/LBfrontpanelTT.zip.

Drill small pilot holes and then your final-size holes. The SW1 and SW2 holes below the legend need to be ½″, the X1/X10 SW hole ¼″, and the speaker jack and power switch holes must be widened to ½″. When all your holes are drilled and de-burred, sand it down to get the old lettering off (3M nylon dish scouring pad works well).

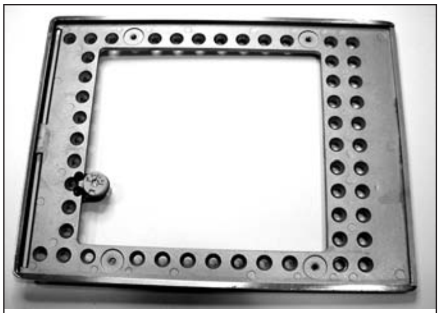


PHOTO 13: Trimmer cap mounting.

Once all the front panel work is done, print out the panel pdf again on sticker paper such as Avery #3383. Tip: let the ink set on the printed paper for about an hour before applying to your panel. Line up your front panel print and cut to size with a sharp razor and straight edge and apply to your front panel.

Put a protective piece of paper or backing material over your panel sticker to burnish down to the face plate, making sure it's free of bubbles or ripples, trim around each hole where needed with an X-acto knife (**Photo 14**). Test all your "fittings." When you are satisfied everything is just right, apply a coat of polyurethane spray. I recommend about three coats.

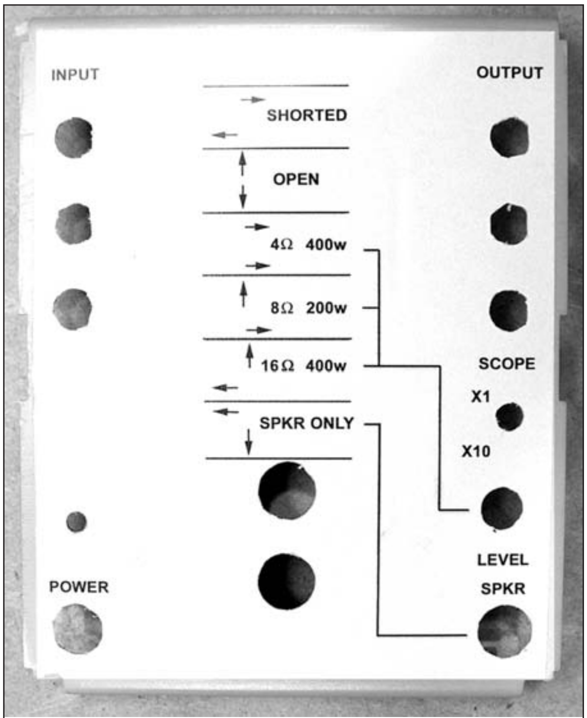


PHOTO 14: Front panel layout.

Another tip: don't try to cover the panel with thick coats—thin applications and more of them always look best.

You should let the final coat of polyurethane set up for 24 hours before assembling. I cut my front sticker so that it just covers the front and does not fold over the top or bottom, nor does it extend on to the outer tabs that snap in to the side supports. I figured this way the panel would fit better, hold up better, and stay flat and clean looking. Now that you have the cutting, drilling, and mounting done, it is time to mount components on the front panel and wire it up.

Don't over-torque your front panel nuts. The sticker label looks nice, but it is really a poor man's silk screen and it will wrinkle if you torque too hard. I recommend that you use the heaviest gauge wire that you can work with for the load hookups. I think #16 stranded should be fine, but #14 would be better. I wired one up with #12 stranded wire (a noble attempt), but found it to be just too big and bulky in such a small space. You will appreciate the flexibility of stranded wire, which will allow you to easily pull the front panel away for ease of wireup. Leave about 2˝ slack for front panel pull-away room.

Check the SW1 and SW2 switch contacts with an ohmmeter before you wire them up. I have seen certain types "make" in the same position as the bat, and others work the opposite. If you use the part specified for SW1 and SW2, they should "make" opposite to the bat position.

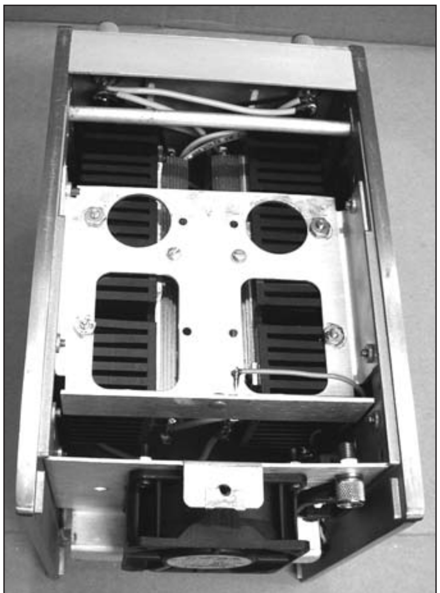


PHOTO 15: Assembly (top view).

Wire up the front panel components as shown in **Fig. 2**. Heavy wire is not needed for the scope connections. All connections in and out of the unit should be isolated from the chassis and ground. Use the specified fiber and shoulder washers where needed.

FINAL ASSEMBLY

Finger-tight-assemble all the components in place while leaving the side, top, and bottom covers off. Attach an ohmmeter to either set of binding posts and verify that you have everything wired up right—that the SW1 and SW2 positioning matches the front panel legend. Check that the speaker only connects to the speaker jack correctly, and so on.

If everything looks good, proceed

AUDIO LOAD PARTS SOURCES

Component	Description	Source	Part Number	Quantity
Resistors				
R1-R8	8Ω 50W aluminum resistor	Mouser	284-HS50-8.0F	8
R9	250R 12W resistor	Digi-Key	B12J250_ND	1
R10	5K CTS pot	Digi-Key	CT 2158-ND	1
R11	9M1 1W pot	Mouser	294-9.1M	1
R12	.27R 3W resistor	Mouser	71-CPF3-G-0.27	1
R13	270R 1W resistor	Mouser	294-270	1
R14	100R 1/2W resistor	Mouser	660-CF1/2L 101J	1
R15	150K 1/2W (Rtemp) (see text)	Mouser	660-CF1/2L 154J	1
C1	Trimmer cap (5–25pF)	Electronic Goldmine	G9371	1
Connectors				
J1, J2	Switchcraft 11	Mouser	502-11	2
	Shoulder washer	Mouser	534-3069	12
	Fiber washer	Mouser	534-3222	12
BNC	BNC connector (isolated sleeve)	Mouser	571-227726-2	1
BP1, BP2	Binding post (Blk) (if needed)	Mouser	530-111-0103-1	2
BP3, BP4	Binding post (Red) (if needed)	Mouser	530-111-0102-1	2
Switches				
SW1, SW2	Carling DPDT ON-OFF-ON	Mouser	691-2GM51-73	2
SW3	Alco SPDT mini toggle	Mouser	506-A101SYZQ	1
SW4	Carling SPST power switch	Mouser	691-110-73	1
Semiconductors				
Q1	MPS A06	Digi-Key	MPSA06-ND	1
Q2	Tip 126 pnp Darlington	Digi-Key	TIP126OS-ND	1
CR1	LED	Digi-Key	160-1504-ND	1
IC1	LM 35DT temp sensor	Digi-Key	LM35DT-ND	1
Fan & Power Supply				
PS module	12V DC switching supply	Mouser	418-CFM1002S	1
IEC1	IEC connector	Mouser	161-0707-1	1
Line cord cable	IEC power cord (90° angle)	Mouser	686-17536	1
	12V DC 2.5W fan 60 × 25mm	Electronic Goldmine	G13229	1 [1]
	Fan guard	Digi-Key	CR 221-ND	1
Misc				
HP 353A patch panel (part of HP 3550B test set, see text)				1
8 point terminal strip		Mouser	58-1008	1
Heatsinks (HS 10198)		Electronic Goldmine	G 2209	2 [2]
Vector board				
Vector T-42 push in terminals				
Hardware				
Heatsink (TO220) kit		Mouser	534-4724	1
4/40 standoffs for power supply PC mounting		Mouser	534-8714	4
A handful of 4/40 hardware & a 4/40 tap set for mounting power resistors to heatsink				
A handful of 6/32 hardware (screws, nuts, lockwashers). Thermal compound.				

1– *Alternate fan source* Digi-Key P-9995-ND 1

2– *approx 4.75˝ L × 2.25˝ W × 1.25˝ D finned aluminum heatsink. A similar sized .5˝ slab of aluminum will work sufficiently. The author has a fair quantity of these heatsinks if the listed source is out of stock. Email me at* www.amprepair.com.

Mouser	*www.mouser.com*	800-346-6873
Digi-Key	*www.digikey.com*	800-344-4539
Electronic Goldmine	*www.goldmine-elec.com*	800-445-0697

to check the 12V supply and fan operation. Make sure the LED is attached before you turn on the power switch. You will need to heat up the load resistor sensor in order to verify that the temp sensing circuit is functioning properly. You can do this with a heat gun or maybe a hair dryer or, of course, you could hook an amp up at 8Ω and run a sine wave into it till the 8Ω load resistor becomes hot. The sensor should kick the fan in at about 100° F. Adjust Rtemp (R15) to tweak the threshold if you need to (typically 100k to 150k).

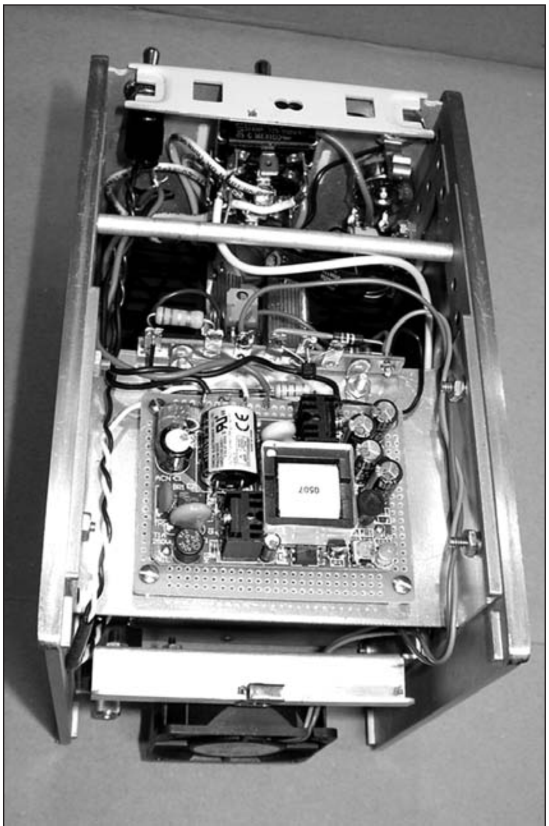


PHOTO 16: Assembly (bottom).

PHOTO 17: Fan vent cutout.

If the fan circuitry is working right, then it is time to check the other functions. Attach an 8Ω 100W (or whatever you prefer) speaker to the speaker jack (J2). Check the speaker-only function. Next place the load setting to the appropriate position for your DUT, apply a signal to the input (BP2, BP4, J1), and check the attenuator through to speaker function.

Hook up the AC voltmeter and scope. Check that these connections are functioning properly.

The ×1 scope setting passes the signal at the input straight to the scope, while the ×10 divides the input signal to the scope by a factor of 10. Line up the side panel on the right side. Mark off and drill a 5/16 hole so the trimmer cap may be accessed. Tighten down all your screws; install the top, bottom, and both side panels before adjusting high-frequency compensation.

The high-frequency compensation trimmer is most easily set by viewing a 10V peak to peak 1kHz square wave: square-wave generator straight into the load box, load box set to "open" scope switch (SW3) at ×1, with your scope attached and set to 2V per div. Observe that the square wave is flat. Without touching the square-wave generator, flip the switch (SW3) to ×10 and change your scope's Vdiv to .2V per division. Adjust the trimmer cap for a flat square wave. A nonconductive adjustment tool works best for this. Note that the placement of the wires attached to the trimmer cap will affect this adjustment.

If you don't have a square wave, use the 204c sine generator; establish the same reference 10V p-p level as above at 1kHz. Without touching the 204c amplitude setting, change SW3 to the x10 setting and change the scope input to .2V per division. Sweep from 1kHz to 20kHz while adjusting trimmer (C1) for a flat response. Repeat this procedure any time you change BNC cable or oscilloscope.

INTO THE CARRY CASE

You could, of course, install these three components on your bench separately or mount the load box back into the carry case. You will need to access the wiring to the existing IEC inlet on the steel carry case and attach the additional AC cord (line cord) 90° angle type IEC plug for the load box.

Measure off where the fan will want to vent at the back of the steel case. I planted the load box in the center, so in my unit this turned out to be 9⅜″ from the right side and 8⁹⁄₁₆″ from the left side, 2⅞″ from the top, and 4¾″ from the bottom. All these measurements direct you to the center point of the fan and are made with the unit sitting on its

PHOTO 18: Completed rear view.

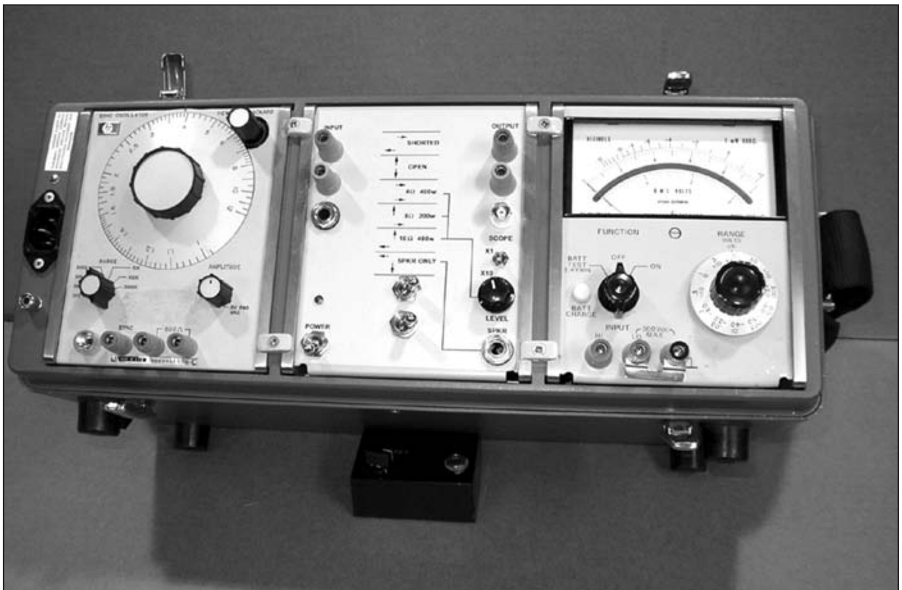


PHOTO 19: Completed restored unit.

face. This is the kind of thing that you definitely want to measure six times and cut once (**Photo 17**).

Did I say steel case? Again, if you are the proud owner of a 3″ circular chassis punch, you may now throw your head back and laugh out loud at how easy this 3″ hole is going to be.

The less fortunate will now prepare for a workout regimen of drilling, filing, cussing, breaking a few metal coping saw blades, and so on. As you can see, I placed a protective fan guard over the cutout.

A few final words about using this or any "dummy load": It is generally not a good idea to run a sine wave, or any other constant amplitude waveform, into a loaded amplifier at high power for extended periods. In other words, don't let your 100W amp sit there burning an 85W sine wave into your dummy load for an hour, to see how well it will hold up. The reason is that the average amplifier's power transformer will in time likely overheat when reproducing a constant amplitude signal such as from a waveform generator.

It is best to "burn in" an output-loaded amp with a fluctuating input source such as a radio or music CD. When using such fluctuating input source, by all means test the amp at high power for extended periods—after all, that would be what an amp is supposed to do. There are exceptions to this statement (industrial- and laboratory-grade amplifiers), and you may generally identify them by their massive power transformers. AC grounding note: to avoid ground loops and other gremlins, I always ground the DUT and lift all the external test equipment grounds.

I am sure you will find this set to be an extremely useful multifunctional test, troubleshooting, and evaluation tool (**Photo 19**). I would like to thank Mr. Thorpe McKenzie for providing the impetus for this project and article. Submit any comments you may have to the letters column in *aX*. Thanks and happy building. *aX*

Mad Katy

A 125W per channel KT88 Amplifier – Part 2

Here's a look at this unit's power supply and construction guidelines.

By Dennis Colin

Back

POWER SUPPLY SUMMARY

1. +500V: KT88 plate supply, unregulated but very solid with 1500µF energy storage (188 joules—do not short!).

2. +450V regulated: plate supply for all other tubes. The two plate supplies are delayed about 20 seconds after mains power turn-on; they're shut down in standby or if the negative bias voltage fails; the +450V supply turns on with about a 1 second soft-start to avoid transients when switching from standby to active.

3. –200V regulated: supplies KT88 bias and linear resistive current sources for cathode-coupled input and driver tubes, and 6SN7 cathode-followers.

4. ±6.3V regulated: powers input and driver tubes' heaters and servo-balance op amps.

5. 12.6V AC: powers series-paired KT88 and 6SN7 heaters.

Five toroidal transformers (Hammond and Plitron) are used, but you could design one custom unit. The AC mains input is EMI filtered, current-surge limited by thermistors, and voltage-surge limited by an MOV. Six fuses are used with corresponding status LEDs.

Following is a detailed description of the power-supply schematic (**Fig. 17**):

AC INPUT

PE1 is an EMI-filtered power-entry module. CL1, 2, 3 are thermistor inrush-current limiters. These are *required*, because as Plitron points out, large toroids draw a large magnetizing current surge: connecting T2 *with no load* to the AC line in one test blew the 15A house breaker! (Once energized, though, the transformer draws only several mA unloaded.) With the added huge surge of the 1500µF cap bank (C1–4) charging to 500V, turn-on would be totally unmanageable. The thermistors' high cold R limits the inrush current to 11A RMS.

A 10A SB fuse, heavy-duty switch, and an MOV voltage-transient limiter complete the AC input.

+500—T1, a 9V 9A toroid, is used as an auto transformer to increase the primary voltage to T2 by 9V. This was needed to obtain 500V DC output. The increased T2 primary V is well within ratings, because T2 is rated for 50Hz as well as 60Hz. At 60Hz a transformer can handle 20% more V than at 50Hz, because it's the V/F ratio that determines flux (therefore closeness to saturation) in a given winding and core.

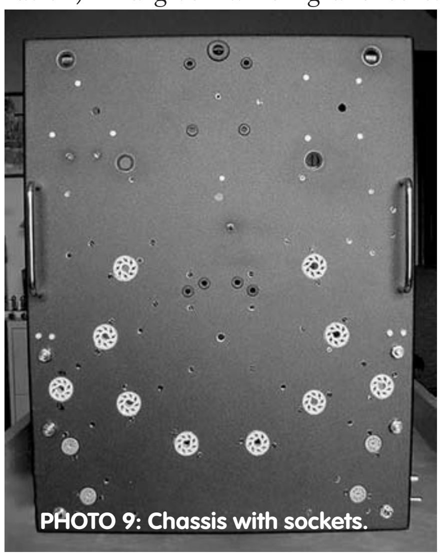


PHOTO 9: Chassis with sockets.

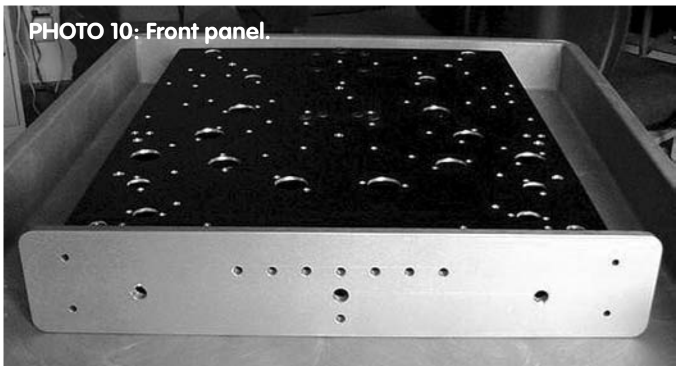


PHOTO 10: Front panel.

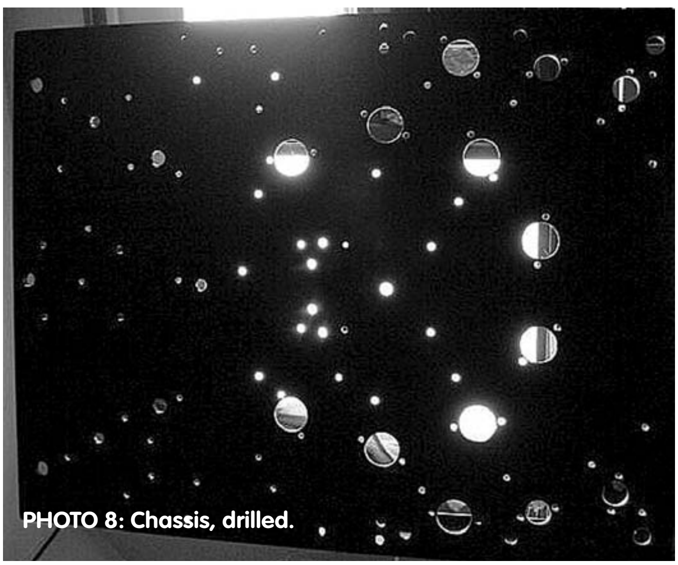


PHOTO 8: Chassis, drilled.

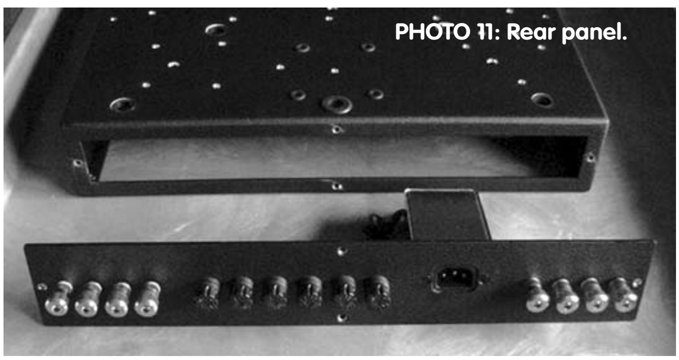


PHOTO 11: Rear panel.

BR1 and C1–4 complete the basic DC supply.

Q1 and associates are not a regulator (because I wanted maximum peak power), but a 500V switch for standby (SB) and delayed turn-on. When the network of R20, 21, 22 allows, Q2 is held off, allowing C5 to charge to the HV input through R5, with a 27mS time constant for some "softness." Then Q1 is fully conducting with only about 3V drop at full load (1.2A). Note: increasing the R5 C5 time constant much, or trying to use Q1 as a regulator, will blow Q1 with 1.2A load current. With the switch used here, there are no problems.

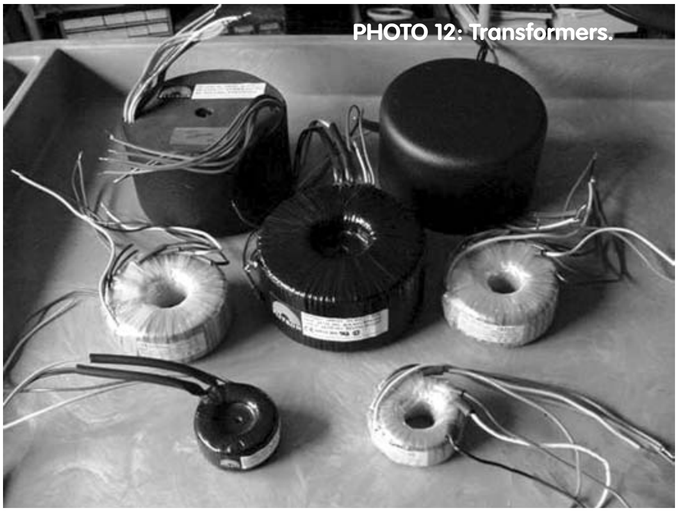


PHOTO 12: Transformers.

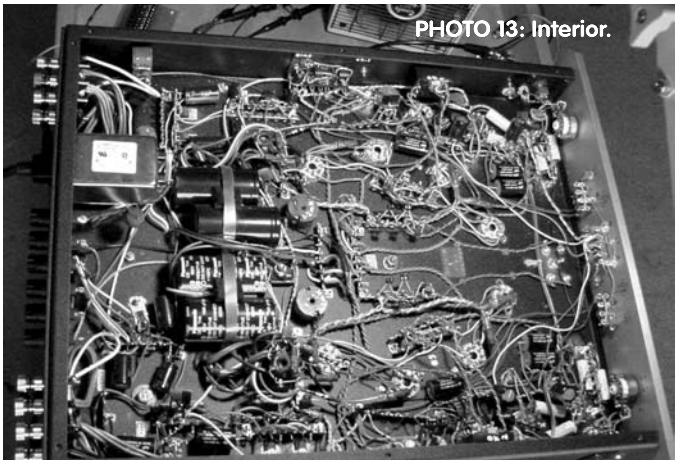


PHOTO 13: Interior.

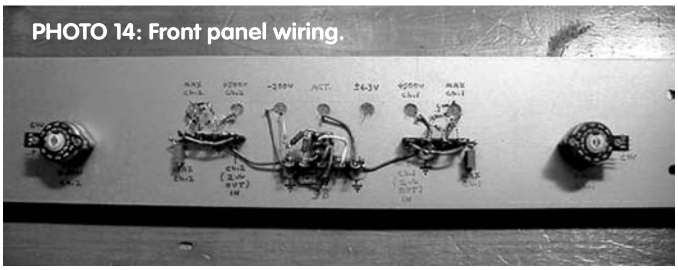


PHOTO 14: Front panel wiring.

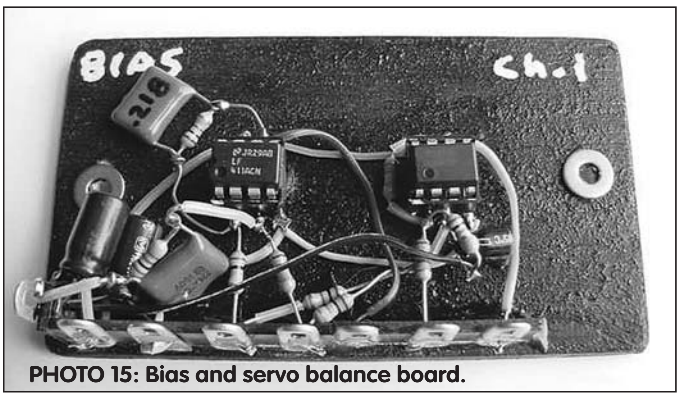


PHOTO 15: Bias and servo balance board.

But to give a sense of the power this small (TO-220) FET can switch, I connected a 500Ω 500W resistor to the raw +500V across the cap bank, *very carefully* with clip leads (using one hand, rubber shoes on dry carpet, no other body part within a foot of any conducting object). When I disconnected one clip lead, interrupting 1A at 500V DC (much worse than AC due to lack of cycle dips), a brilliant, beautiful, sizzling arc grew to a length of about ½″—this removed any ideas I had about using a relay! But the FET works well and eliminates a mechanical component.

With no output from the –200V supply (due to this bias voltage failing or its turn-on lag), Q2 is quickly turned on by R20; this switches and keeps Q1 (and the output) off. The –200V supply would be enough to overcome this, biasing the Vgs of Q2 negative through R22, which would activate the 500V output, except for Q5.

Upon AC turn-on, the +6.3V supply is

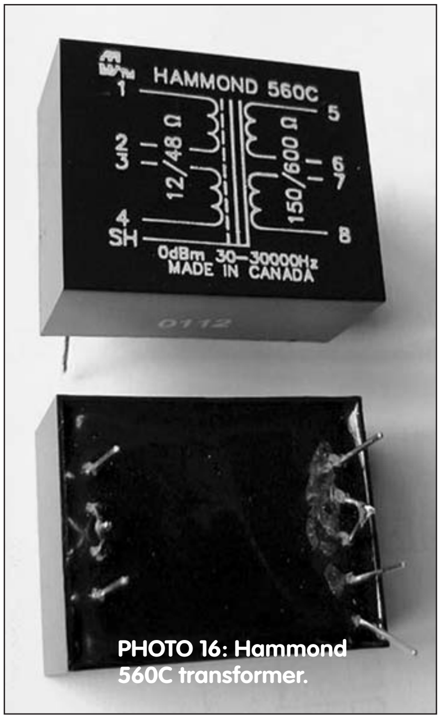


PHOTO 16: Hammond 560C transformer.

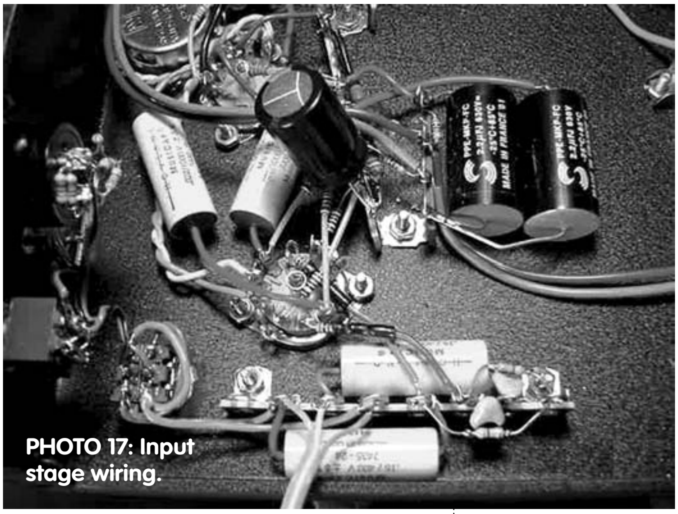


PHOTO 17: Input stage wiring.

PHOTO 18: Optional fan.

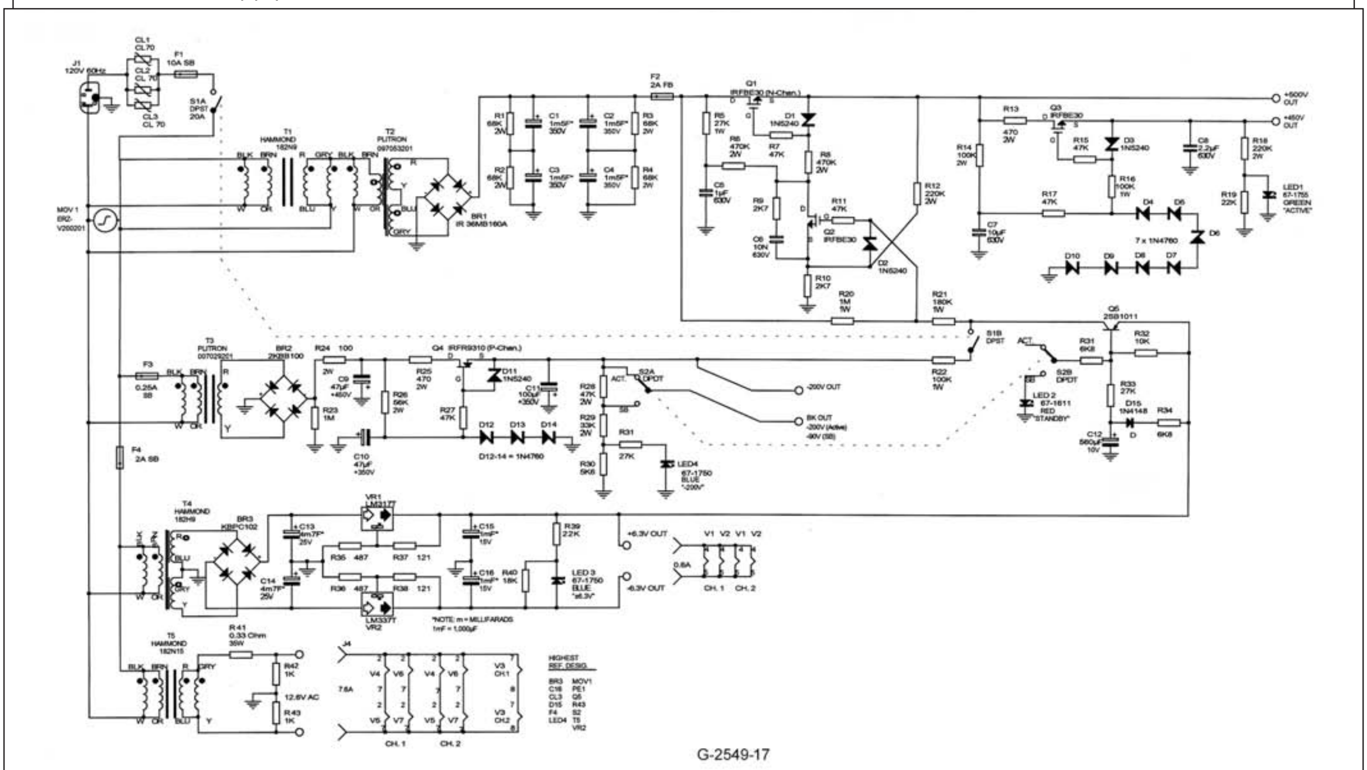
applied to the Q5 emitter. Because C12 is initially discharged, Q5 is quickly turned on (before the HV supplies charge up). Q5 collector current fed into R21/R22 turns Q2 on, keeping the 500V output off. After about 20 seconds, C12 charges enough to turn Q5 off, if S2B is in the "active" position, and the 500V turns on. With S2 in "standby" position, R31 and LED 2 keep Q5 on, and the 500V off.

The purpose of R12 (feeding Q2 source) is to provide some (positive FB) hysteresis to the switching circuit, to ensure no "hanging around" between on and off, which would blow Q1. Note the switch S1B in the –200V path through R22. This is a second pole of the main AC switch S1A, and the purpose is as follows: without S1B, if the amplifier was in standby when the AC was switched off, the amp became active several seconds later, producing a few seconds of decaying, distorted audio. This was because (a) the –200V decays slowly upon power-off, and (b) the –6.3V decays quickly, preventing Q5 from continuing the "standby" function long enough for the cap bank C1–4 to discharge. S1B interrupts the (decaying) –200V, making the 500V Q1 switch "think" the bias has failed; this keeps the 500V off.

The turn-on timing cap C12 is discharged upon AC power-off by D15 and R34; without R34 the discharge would be as quick as that of the –6.3V (quick because it's supplying tube heaters). Before adding R34, a ¼ second AC power glitch during a storm shut the amp off, and I had to wait the 20-second delay for the music—not acceptable! With R34 = 6.8k, the AC power can be interrupted for about 2 seconds, without initiating the 20-second turn-on delay.

+450V REGULATOR—Q3 is a conventional regulated source-follower,

FIGURE 17: Power-supply schematic.



with D4–10 as the reference. Because the load is only about 50mA (input and driver tubes), Q3 is not strained as a linear regulator, nor by the one-second "slow-start" time constant of R14–C7. The latter prevents bias surges or other audible transients from occurring at AC power and standby switching. All such switching is completely smooth and free of audible transients.

-200V BIAS REGULATOR—T3 is a 200V AC toroid, supplying about –300V DC to C9. Q4 and associates are similar to the +450V slow-start source-follower regulator, except for using a P-channel FET.

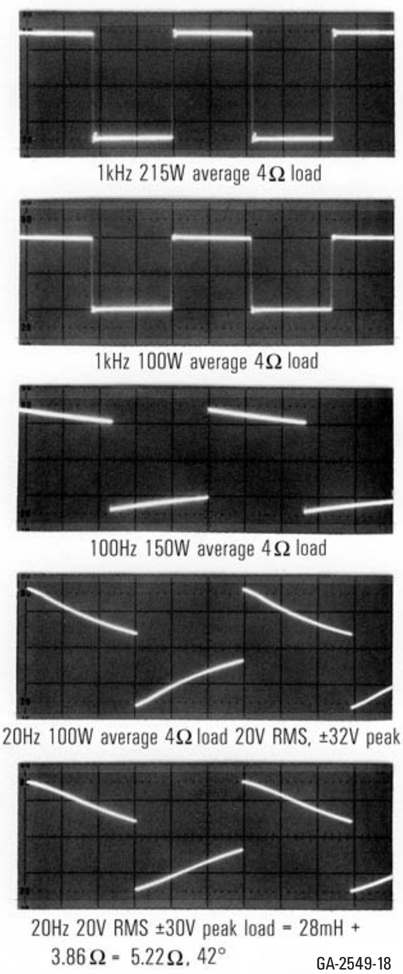
Note the "BK out" tapped down to –90V by R28, 29, 30, and S2A. "BK" stands for "Bias, Cathode," and its purpose is to reduce the voltage driving the 6SN7 cathode followers during standby; without plate voltage, the 6SN7 cathodes would otherwise reach –200V, which is the peak heater-cathode 6SN7 rating.

The KT88s need only about –130V peak, but I wanted –200V because the 6SN7 cathodes (directly driving the KT88 grids) must maintain enough current to keep their low output resistance on negative cycle peaks, avoiding slewing distortion.

±6.3V REGULATORS—Not much to say here; perfectly conventional linear regulators supplied by toroid T4. In addition to the servo-balance op amps, the ±6.3V outputs power (differentially) the 12.6V heater pins (4, 5) of the 12AX7 and 12AT7 tubes.

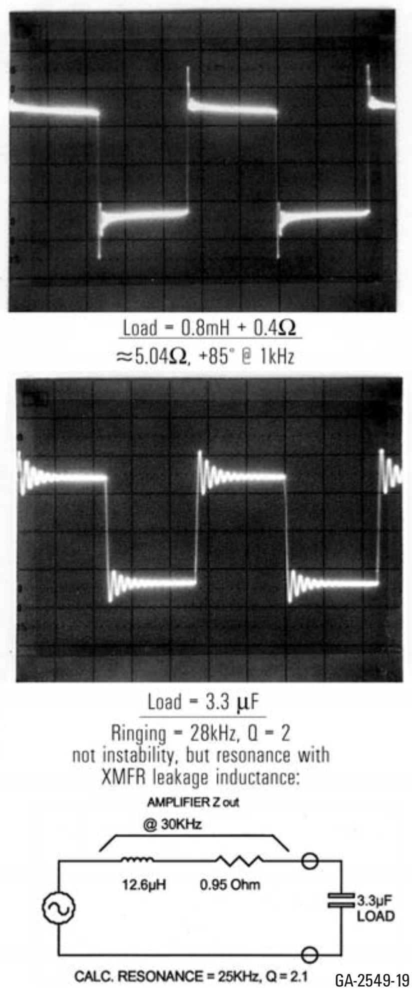
12.6V AC—Powering the 6SN7 and KT88 heaters (series pairs), this uses a 15V 8A toroid. R41 drops the loaded voltage to very close to 12.6V, and provides some surge-current limiting. With this plus that provided by CL1–3, the KT88 heater surge is only 3.9A (2.4 × the 1.6A hot current). Without any limiting, the cold surge would be about 10A.

FIGURE 18: Squarewave responses.



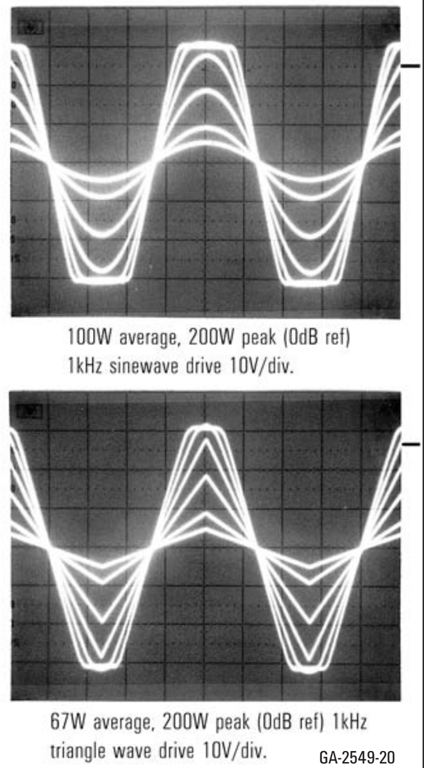
GA-2549-18

FIGURE 19: Stability with extreme loads.
1kHz square wave response, output = 28.3V RMS (would be 200W with 4Ω load).



GA-2549-19

FIGURE 20: Overdrive limiting characteristics.
4Ω load 0dB ref – rated maximum power. Input levels (re 0dB ref) = +4.4, +2.5, 0, –3.5, –9.5, –15.5dB



GA-2549-20

This power supply is probably more extravagant, and certainly more expensive (five toroid transformers!) than necessary. But I wanted it to be very clean (including very low magnetic leakage) and rugged. Regarding the latter, it (and the amplifier) has withstood hours of abuse (extended overdriven square-wave tests, shorting the amp outputs to test the current-limiting, driving a ⅙ HP motor, making sparks) and months of normal music playing.

CONSTRUCTION

The chassis is a Plitron PAT-CASE-22 (22″ × 17″ × 3″), about $220 (US). Extremely solid, it's also very good-looking, with ⅛″ thick black-crinkle finished aluminum surfaces and a ⅜″ thick gold-painted front panel (**Photos 8–11**). One safety caution: The chassis is supplied with black (anodized?) machine screws. These do *not* provide front, rear, and bottom panel electrical contact with the chassis! (In case of some short, would you want the front panel charged up to 500V?)

You *must* scrape the screw threads and the inside panel threads at least partially bare. Likewise, chassis mounting holes for all exposed components (handles, transformers, controls, and so on) and internal ground points (terminal strips, and so on) must be surrounded by a ring of bare metal (a hand drill grinding tip works well). Please observe the utmost safety precautions—when measuring in the chassis, use only one hand; let no other body part come near any conductive object!

CHASSIS LAYOUT

I don't have dimensioned layout drawings, having "winged it" on grid paper. But you can scan the photos and computer-scale them if you want the same layout I used. If so, the KT88s are on a 10″ diameter circle, spaced 36° apart (ten tubes would complete the circle; eight are used).

Five toroid transformers are used in the power supply (**Photo 12**). Chassis placement, front to back, is:

1. T1 stacked atop T5, centered on KT88 circle.
2. T2, the plate supply transformer.
3. T3 stacked atop T4, at the back of the chassis.

While there is no problem stacking the toroids (very low magnetic leakage and heat), Plitron mentions a "shorted turn condition": If you connect an AC voltmeter from chassis to the mounting bolt of, say, T2, it will read about 0.7V RMS. The bolt and meter/leads constitute a one-turn secondary looping through the core; 0.7V/turn is the winding induction. The warning is to not have any grounded metal (strap, metal enclosure if used, and so on) contact the toroid mounting bolt top. With low enough path resistance, that 0.7V AC could (with the capacity of T2) produce 1500A of current in the mounting bolt, and you'd have quite an arc-weld to remove!

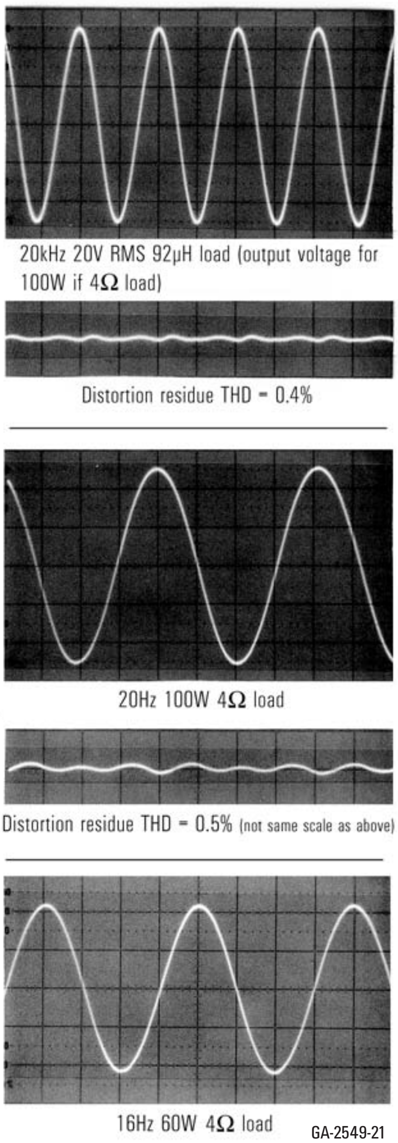
On the chassis top are (1) polarity switches—near front panel, non-inverting with handle away from panel, and (2) bias pots, just behind the 6SN7s. Note: the pots to the front of the bias-setting ones are not used—they were planned as linearizing adjustments, but proved to be unnecessary.

TUBE LAYOUT AND WIRING

I gave much thought to minimizing hum, noise, crosstalk, and unwanted feedback ("self-crosstalk"). The effort was well worth it: output noise (20Hz–20kHz) is 0.053mV RMS, hum is 0.3mV, and crosstalk is –86dB up to 10kHz, -75dB at 20kHz. Steps taken were:

1. Tube layout: admittedly, I wanted the circular pattern of KT88s for its

FIGURE 21: Sinewave outputs.



20kHz 20V RMS 92μH load (output voltage for 100W if 4Ω load)

Distortion residue THD = 0.4%

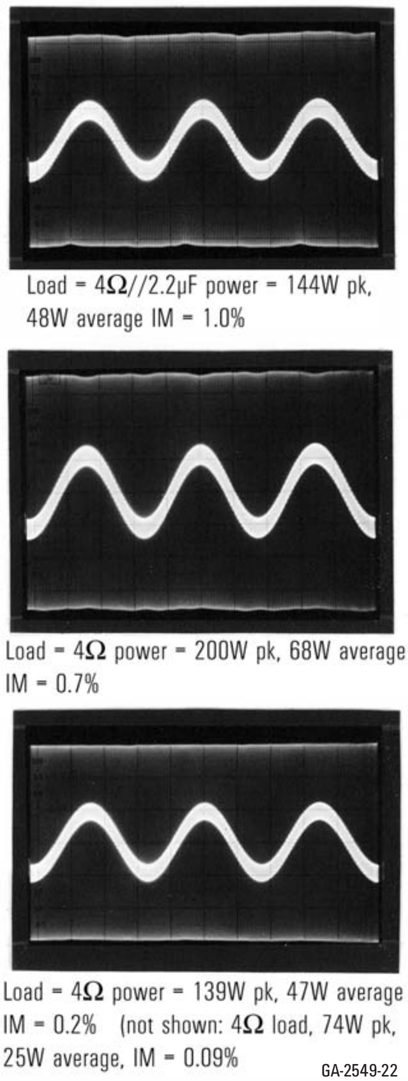
20Hz 100W 4Ω load

Distortion residue THD = 0.5% (not same scale as above)

16Hz 60W 4Ω load

GA-2549-21

FIGURE 22: Amplifier IM test.
200Hz + 10kHz, 4:1 voltage ratio. Inner trace = amplifier output. Outer trace = filtered 10kHz output.



Load = 4Ω//2.2μF power = 144W pk, 48W average IM = 1.0%

Load = 4Ω power = 200W pk, 68W average IM = 0.7%

Load = 4Ω power = 139W pk, 47W average IM = 0.2% (not shown: 4Ω load, 74W pk, 25W average, IM = 0.09%

GA-2549-22

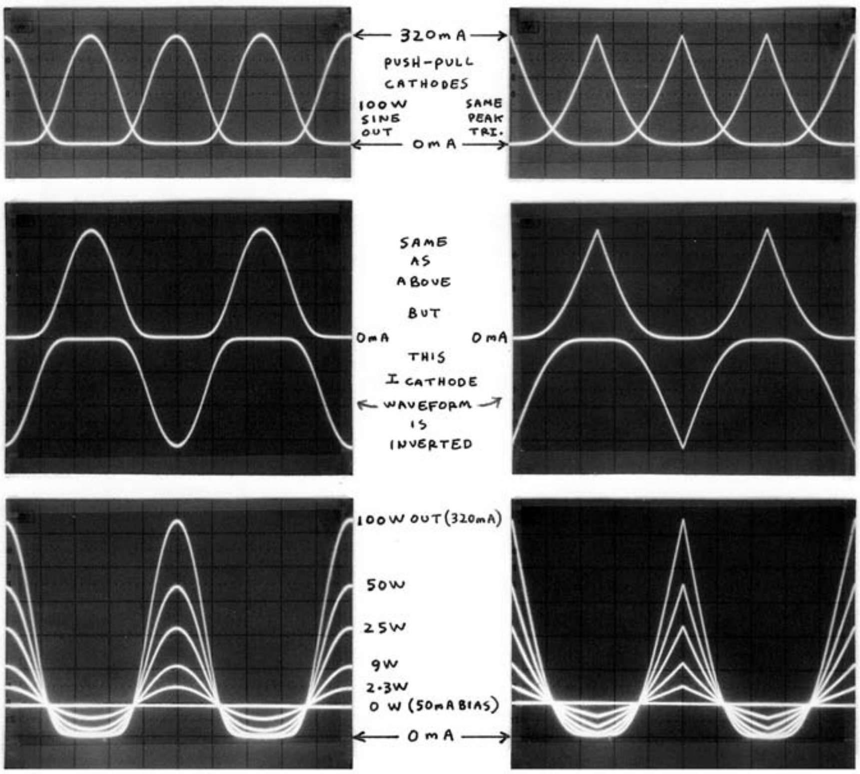
aesthetics, the glowing “ring of fire.” But it also serves audio purposes: First, the centered heater transformer toroid is close to all the KT88s; minimum voltage drop and hum radiation. Second, the KT88s’ placement alternates push, pull, push, pull, rather than adjacent paralleled tubes. This significantly cancels the radiated electrostatic field of each plate’s 700V p-p maximum signal, to minimize HF capacitive feedback to the input wiring.

The 12AX7 input tubes and associated components (**Photo 17**) are nestled tightly near the front corners of the chassis; audio wiring is close to the chassis floor.

2. Wiring: I used Alpha 600V #24 for the +500V and +450V wiring, 300V #16 for the AC line feeds, 300V #18 for the 12.6V AC heaters, and 300V #22 for everything else.

Note that the KT88 plates and screens are wired with twisted pairs, each conducting a balanced push-pull signal. This is facilitated by the KT88s’ adjacent push-pull placement described above. The KT88 heaters are wired in series pairs of push-pull tubes; this has a nice advantage that if one tube’s heater opens, its push-pull partner will also become deactivated, thus preventing a huge DC imbalance in the output transformer. Actually, the servo-balance alone would reduce such an imbalance to about 4mA, roughly quarter-saturating the transformer but allowing reduced-power functionality.

FIGURE 23: Class AB current waveforms (per KT88; 4 used).



Sine wave

Po average W	Po peak W	I/KT88 average mA
100	200	110.0
50	100	82.9
25	50	67.4
9	18	56.2
2.25	4.5	51.8

Square (not shown)

Po average W	Po peak w	I/KT88 average mA
200	200	160.7
100	100	114.2
50	50	84.3
18	18	62.1
4.5	4.5	53.0

Triangle wave

Po average W	Po peak W	I/KT88 average mA
67	200	92.5
33	100	72.5
17	50	62.1
6	18	54.3
1.5	4.5	51.2

GA-2549-23

I used many terminal strips to keep the extensive amount of wiring as uncluttered and close to the chassis as possible. One point worth mentioning is that the KT88 base shell (pin 1) is grounded; I’ve seen some schematics that don’t show this. Ground it for safety!

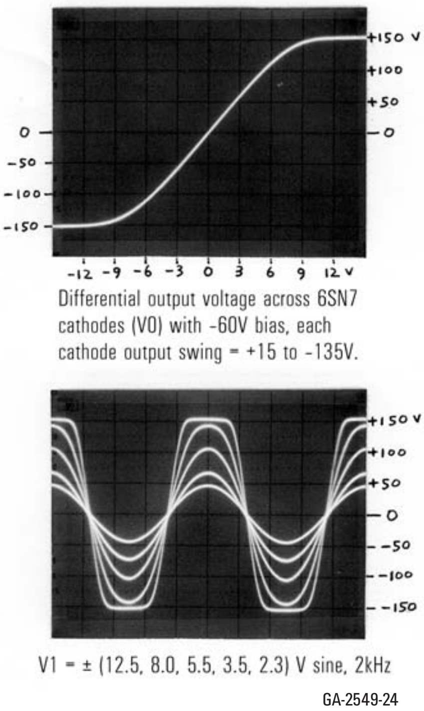
OPERATING TEMPERATURES

Table 2 shows the temperatures of various parts of the amp, after being on for one hour, measured with an IR thermometer. Also shown is the cooling provided by a small, quiet fan assembly blowing down over the toroids centered among the KT88s (**Photo 18**).

Without the fan, the hottest spot (excluding tubes, of course) was the chassis atop the 0.33Ω heater dropping resistor (R16, **Fig. 17**)—this spot reached 76° C (170° F). This is not excessive, because (1) all components are conservatively operated with regard to their ratings, and (2) the most heat-degradable components, the electrolytic supply caps, are rated for 105° C.

The temps in **Table 2** were measured at an ambient of 24° C (75° F) and with the amp on but no signal, where the AC line consumption is 398W. For a

FIGURE 24: 12AT7 Driver + 6SN7 cathode follower transfer function and limiting waveforms (used to linearize output stage).



worst-case heat scenario, consider both channels driven to clipping with typical music (500W AC consumption) and an (unlikely) room ambient of 100°F (38° C). Then the hottest chassis spot would reach about 103° C. So the fan is probably advisable if you listen at full power in the tropics with no air conditioning, but under more reasonable circumstances you should not need it.

All components are conservatively operated. Maximum KT88 plate + screen dissipation is 28W; the Genelex rating is 40W recommended, 46W absolute maximum. The tubes should last a very long time.

With the JJ KT88s, performance changes very little over a (cathode current) bias range of 45–65mA. After about 30 hours of break-in (measuring and listening), bias was set to 50mA (±2mA among all eight tubes). One year later, with about 150 accumulated hours, the bias had changed by only about 1mA.

Figures 18–25 show a variety of test waveforms.

THE SOUND

Ideally, an amplifier shouldn't have any. In the real world, an added "warm," "romantic," "tubey" sound may be valued to compensate recording "dryness," and so on, or simply for preference. But the goal of Mad Katy is faithful signal (therefore sound) reproduction of the source, be that good or bad.

Over the year since the amp was completed, I've done much listening and performed several comparison tests. Following is a summary of results. But please note: the tubes are biased, but I'm not—after spending two years on this project, I certainly wanted to know whether there were any problems with the result—so I've almost fanatically searched for sonic anomalies.

Also, my hearing is good, but extends to 14kHz (at 85dB SPL). So those who can hear that highest half-octave (to 20kHz) might hear things I didn't (not much musical energy up there, though). Summarizing my observations:

1. Normal listening: very satisfying; natural, uncolored, and resolving with well-

FIGURE 25: AC Line input waveforms.

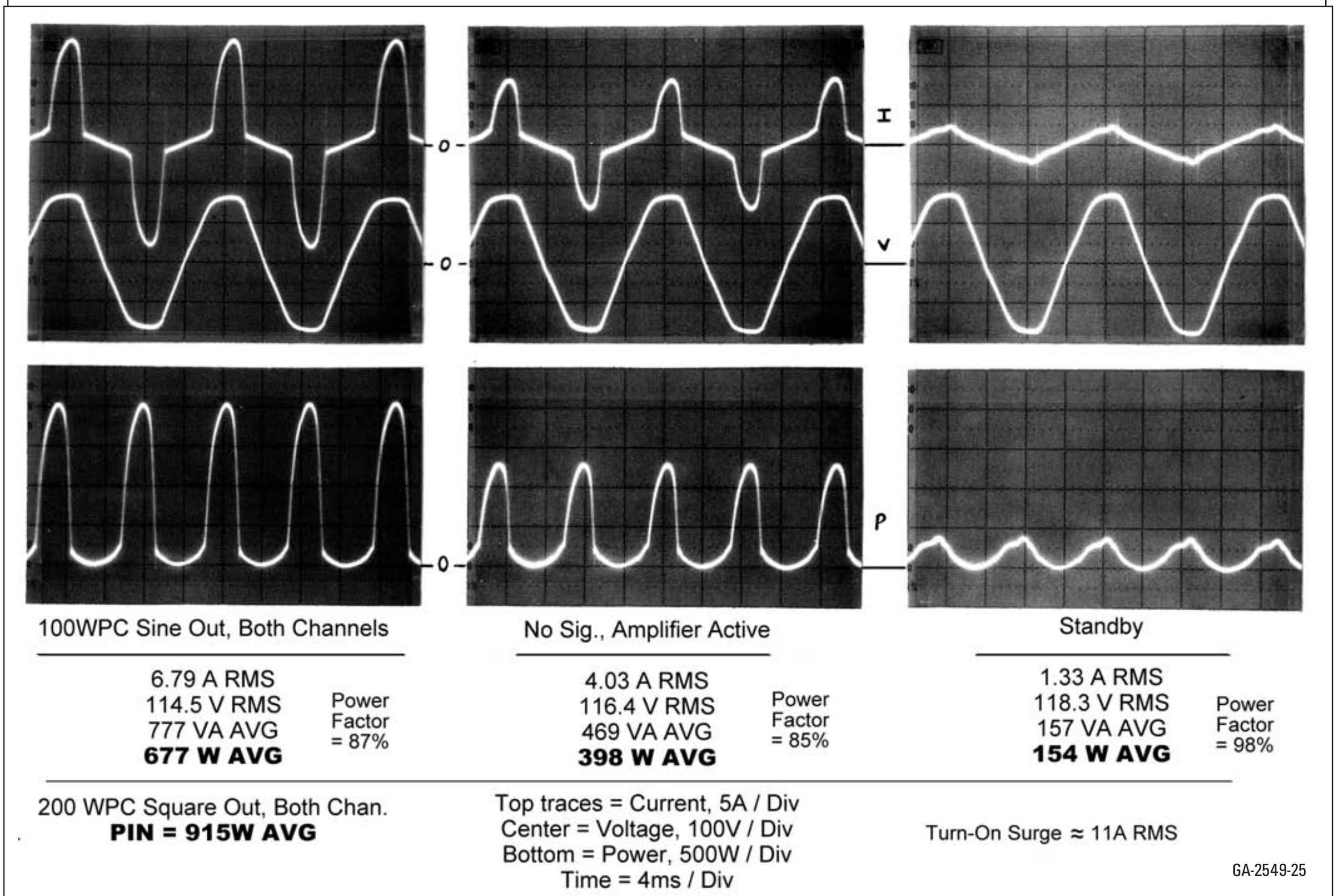


TABLE 2: OPERATING TEMPERATURES.

Temps after 1 hour on (active, no signal)
Ambient T = 24° C = 75.2° F°
Fan: Papst 3412 L (on 6.3V DC – virtually inaudible)
Rated 12V; used at + 6.3V, 61mA
Mounted over 9V and 12.6V transformers, blowing down

Item		*T, ° C no fan*	*T, ° C with fan*	*Fan cooling ΔT, °C°*
Tubes	KT88	195°	163°	32°
(maximum	6SN7	128°	111°	17°
envelope	12AT7	118°	100°	18°
temperature)	12AX7	81°	68°	13°
Transformers	+500V	51°	40°	11°
(usage	12.6V AC	70°	51°	19°
specified)	9V AC	63°	45°	18°
	–200V	58°	51°	7°
	±6.3V	48°	47°	1°
	output	33°	31°	2°
Chassis over 0.33Ω 12.6V AC ballast		76°	60°	16°
Near KT88s		65°	51°	14°
Top, near front		57°	46°	11°
Sides, mid		47°	41°	6°

recorded SACD music. No sense of strain up to clipping (155W per channel peaks).

2. Live-miked voice and a 2″ diameter bell: I used the Panasonic WM-60 AY capsule (used in Mitey Mike II, *SB* 4/97) and the Swans M1 speaker with excellent ribbon tweeter, coherence, and neutrality (*SB* 3/99).

With the bell, the amp was driven to just below clipping on the (very high peak) metal impact transients, which were reproduced cleanly and as solidly as with a comparison solid-state amp. But both the transient "clink" and the resonating "clang" sounded more like the actual metal bell, with the tube amp. Voice was more natural with the tube amp; vowel sounds had that natural (not euphonically softened) "roundness" or "fullness" of tone.

3. Intrinsic fidelity test: This is based on "Sonic Comparison of Power Amp Output vs. Input" (*aX* 12/04). **Figure 26** is the test setup I used with Mad Katy. The setup includes:

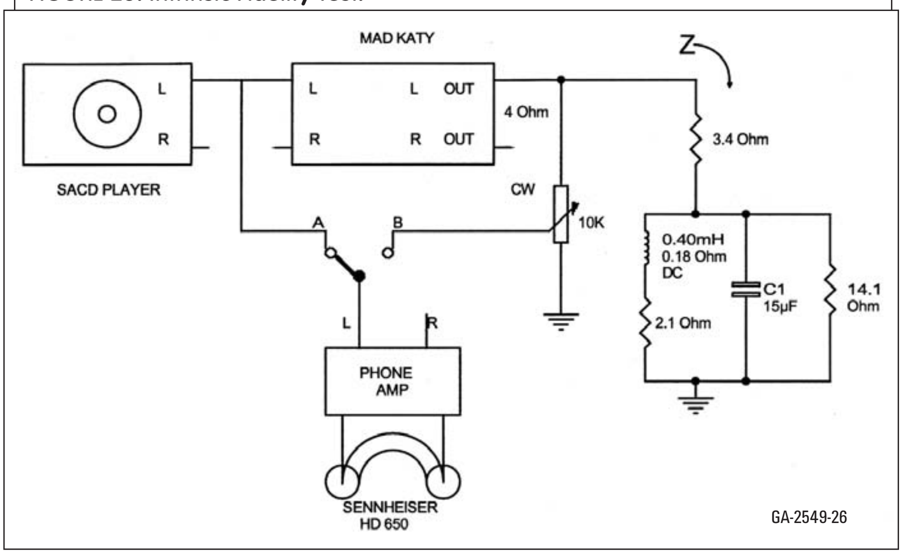
a) a reactive and varying load. This does not produce the typical speaker's bass resonance impedance rise, but Mad Katy had not shown sensitivity to LF impedance variations in previous tests. However, the amp did appear to produce a slight "softening" of HF detail with a speaker whose impedance dipped to 3Ω from 10–20kHz, when connected to the amp's 4Ω tap. So I chose the load in **Fig. 26** to have a large |z| variation (2.84:1) from 2–20kHz, reaching a low of 3.46Ω at 20kHz. Load phase angle ranges from +12 to –32°.

b) The exceptionally clean and resolving Sennheiser HD 650 headphones. These have a –3dB BW of 16Hz–30kHz and can reproduce an SPL of 125dB. They are expensive but worth every penny.

c) The phone amp described in "A High Quality Solid State Headphone Amp" (*aX* 11/05). Pure Class A, very transparent, can drive the HD 650 to 119dB SPL, noise floor is 4dB SPL.

With a test tone, the 10k pot was adjusted (both channels, one shown in **Fig. 26**) for A/B (input/output) level matching to within 0.05dB. The amp's input attenuators were set up so that 0dBFS (2.0V RMS) full-scale peaks drove the amp to 150W per channel, just below clipping. I listened with headphone volumes ranging from about 70–100dB SPL average, about 82-112dB peak.

FIGURE 26: Intrinsic Fidelity Test.



With a variety of SACD and CD music, I switched between the amp's input and output, sometimes rapidly, sometimes after hearing a complete musical selection, and sometimes rapidly A/B'ing the first 1–3 notes of a piece.

I could not hear any difference I was certain of. By this I mean that (1) for the most part I heard no difference at all, and (2) when I occasionally thought I might have heard some difference, which if at all existing was on the edge of perceptibility, the possibly-heard difference evaporated upon extended switching repetition.

The lack of audible input/output differences may lead some to believe that I have ears of tin, am biased more than the tubes, and/or am in denial. However, despite the current plethora of audibility claims regarding amplifiers (not to mention a few oxygen atoms in a wire), there are much less-publicized (and much more scientifically rigorous) evaluation studies of amplifier sonics, including one by Peter Walker of Quad fame. The latter test, among some others, concluded that the amps compared produced no audible difference.

Among those power amps designed for the most faithful reproduction (as opposed to those designed to euphonically compensate/complement recording/playback chain deficiencies or one's tastes), I suspect that the intrinsic fidelity test of **Fig. 26** would reveal little or no audible in/out differences.

Therefore, considering the largely ignored studies mentioned above, I would conclude *not* that Mad Katy is by any means "perfect" or "the world's best," or any such nonsense, but rather that it is sufficiently neutral, transparent, solid, and resolving to allow one to set it, forget it, and enjoy the music, with 250W (⅓ horsepower) coming from electrons working in a vacuum. ***aX***

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AUDIO NEWS

Pass Labs is pleased to announce that Texas Instruments has purchased a non-exclusive license to US patent #5,376,899. The amplifier topology, marketed by Pass under the tradename “SuperSymmetry” provides a means of achieving high performance with balanced symmetric feedback and minimal gain stages.

SuperSymmetry has been the basis for amplifiers marketed by Pass since the 1998 introduction of the X-1000, a 1000W power amplifier using only two gain stages—differential input transistors driving complementary power output transistors.

“The beauty of the design lies in the combination of power and simplicity. We believe that the highest quality sound is achieved with a simple gain path, and SuperSymmetry gives us high performance with minimal complexity and negative feedback,” says inventor Nelson Pass.

Yard Sale

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Ad Index

CORRECTION

In Yury Gelgor's "A Balanced Nonoversampling Retro Dac" (*aX* 6/06), there is a problem with the power supply shown in Figure 6. **Figure 1** shown here is the corrected diagram. We urge that you please disregard the original figure for the safety of the reader and others.

CORRECTION

Arghhh! In the interview with Peter Bartók (June 2006), I accidentally sent in an incorrect rendition of Fig. 1. A quick glance will reveal that the last 12AU7 stage is in fact a cathode follower, not a voltage amplifier (and, indeed, without a plate resistor it wouldn't be doing much voltage amplifying). **Figure 2** shows the corrected figure; the topology is the same one later made popular in the Audio Research SP3 and subsequent preamps, with beefier tubes. My apologies for the error.

Paul J. Stamler

SOLID-STATE SE AMP

According to Ed Simon's article in the April 2006 issue ("A Solid State Single-Ended Power Amp"), the amplifier is rated at 15W continuous (not RMS!) power output into an 8Ω load. As the author states, such an amplifier would require a peak current of 1.9A, and the output stage must be biased to at least this current.

However, further along in the article the author decides to reduce the bias current to 600mA. I have no quarrel with that, ex-

FIGURE 1: Gelgor's corrected power supply circuit.

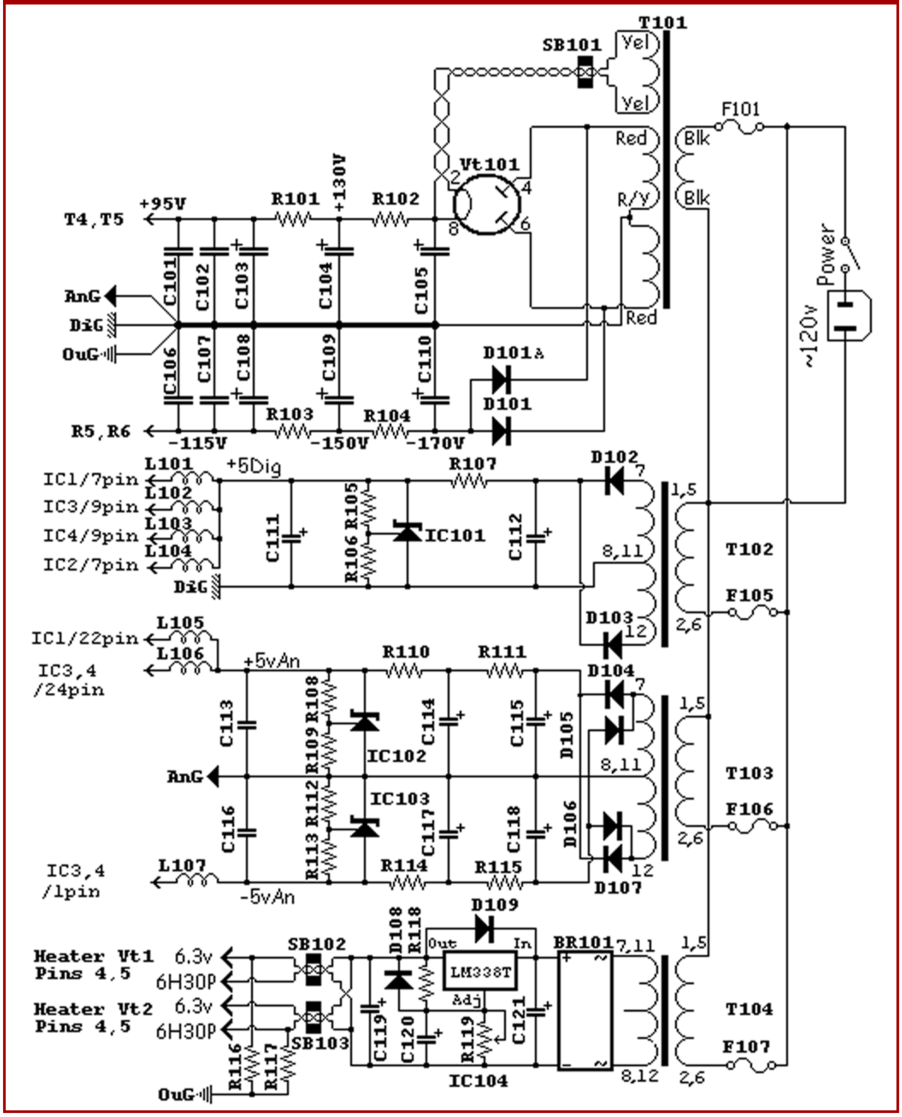
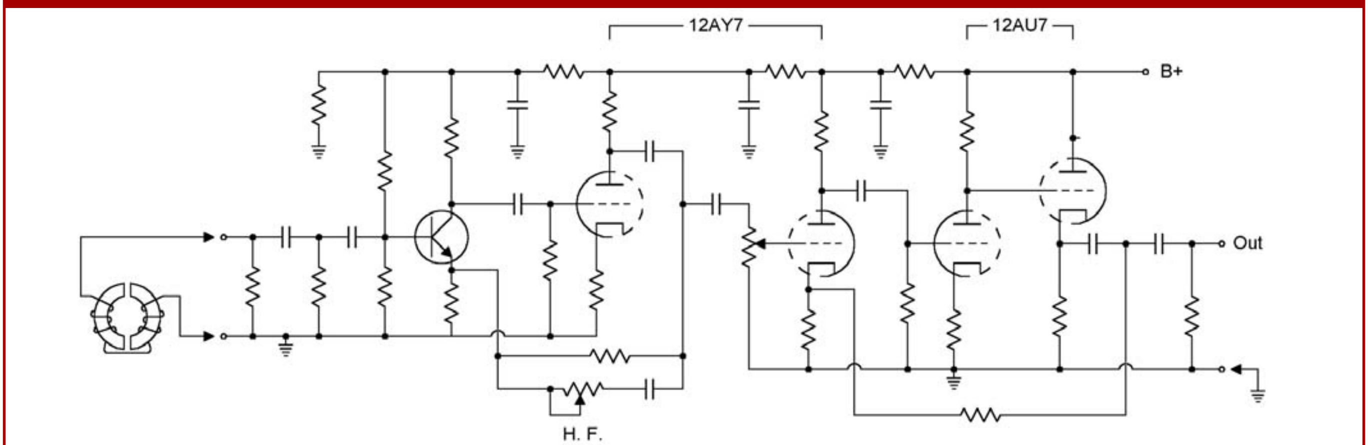


FIGURE 2: Stamler's corrected hybrid tape playback preamp.



cept that he neglected to mention that this would limit the peak current to the speaker load to that same amount (at least for symmetrical output), and the maximum power output would be reduced to about 1.5W continuous, a far cry from 15W.

Otherwise, I found the article very interesting and commend the author for letting the reader know his design decisions in detail.

Jaime Arbona
Mayaguez, Puerto Rico

Ed Simon responds:
I am sorry that my prose left you with a different meaning than I intended. I did not make clear that the heat dissipation is different from the amplifier's output. The 15W figure is derived from my discussion of the power transistor's ability to dissipate heat.

I was careless and should have explained more clearly that in a Class A amplifier the output device dissipates far more energy than is delivered to the load. In a Class B amplifier way more energy makes it into the load. That is why most amplifiers are Class AB. The continuous bias current makes Class AB sound far better than just Class B, and the Class AB amplifier is more efficient than a Class A amplifier.

In a linear amplifier the output device(s) varies the voltage and current from a fixed supply (the "rails") to the load. During the reproduction of any waveform when the output is exactly at half voltage, the output device(s) has half of the supply voltage across it, as does the load; since the current is the same, the device must dissipate as much power as the load. When the voltage out is equal to the supply voltage (rails), then in theory maximum current is delivered to the load, but since there is no or, in practice, a very small voltage across the output device(s), there is very little power dissipated in the output device(s). Almost all the power has gone into the load.

When there is no output voltage, all of the supply voltage is across the output device(s), but in a Class B amplifier there is no current going anywhere, thus no power dissipation. In a Class AB amplifier only the small amount of current used to bias the output transistors is dissipated. In a Class A amplifier all of the power is dissipated because the maximum linear output current is going through the output device(s) and none into the load.

Almost all courses or books that cover this material also teach that maximum power transfer occurs when the load impedance is the same as the source impedance. However, what they often neglect to mention is that under those conditions the power source dissipates as much energy as does the load.

Yes, I deliberately did not give the amplifier's actual power output. A 600mA bias current can deliver an RMS current of only 424mA. The 21V supply can deliver only 14.15V RMS. Thus the maximum power would be 6W (power is power not RMS power, RMS × RMS cancels) split between the optimum load and the amplifier. The optimum load resistance would be 16.69Ω for the article's amplifier.

The power into such a load would be 3W, before any asymmetry. The peak current would be much greater, allowing the amplifier to behave as if it were the equivalent of 15.5W of power. That is one of the reasons why Class A amplifiers sound as though they have more power than the equivalent Class AB amplifiers.

You have also correctly calculated that the power into the speaker is reduced at lower impedances. Loudspeakers heat up as they are played. This causes the voice coil to increase its resistance. This effect is called thermal compression.

In an amplifier of this design, as the impedance goes up so does the power available to the loudspeaker. That is one of the reasons why Class A amplifiers are often called more dynamic than other designs. They tend to counteract the thermal and mechanical compression inherent in most loudspeaker designs.

I read Ed Simon's "A Solid-State Single Ended Power Amp," which I found rather intriguing; however, I do have some personal observations and questions regarding the design.

I found it interesting that the lack of symmetry in the drive is justified, and I also question the quality of sound. That may be due to ignorance on my part, but here is my thinking process: As the cone of the speaker is drawn in from the driving section of the circuit, electrical energy is converted to mechanical energy. When the driving energy is released, the stored

Continued on p.62

Continued from p.59

mechanical energy would force the cone of the speaker outward. If all the converted energy is released after storing the mechanical energy, then bidirectional drive would not be required at all.

It seems to me that some speakers would not have the capability to rebound from the stored energy. In fact, it would not surprise me that every speaker is different, or that the current source developed by Q5 would need to be different for different speakers that are driven. The amount of drive for the return of energy to the wave energy in the air should balance the energy that was being stored.

If a speaker had the capability to return all the energy that was stored in driving it backwards, then the current source would not be needed, for the mechanical energy stored would have been converted to the sound wave in the air!

So, here's the big question: Why are push-pull topologies used? If the assumption is that the speaker has the capability to push the air with minimal drive, then even the push-pull topology would distort sound, for it is pushing the air with more energy than it was pulling with, for the total energy in moving the air would be the sum of the stored mechanical energy of the speaker plus the additional energy of the driver pushing the speaker more.

It seems to me that speaker design is such that it requires push-pull drive, but it may be possible to design a speaker that would not require a push-pull design. It is difficult for me to comprehend how a non-symmetric drive can produce sound that is not, in some degree, distorted. How is the sound quality and distortion of the proposed amplifier evaluated and how does it compare with a push-pull driver of the same capability with the same power being delivered to the speaker?

Granted, the design is simple and "plurality should not be assumed without necessity." "Keeping it simple, stupid" is a good concept, but there are times when "keeping it simple" must be balanced with stupidity. Perhaps I am ignorant regarding the design and in that case I will need to better understand or more fully appreciate the meaning of "plurality should not be assumed without necessity."

Robert G. Black Jr.
Oro Valley, Ariz.

Ed Simon responds:

I think a bit of clarification is in order. A push-pull amplifier can also be Class A, which is generally considered the best-sounding amplifier. If you built one that way, the limit on amplifier power would most likely be heat dissipation. The drive to the loudspeaker would be symmetric until you ran out of power and then you would get clipping. How the amplifier is designed to handle clipping would affect the sound quality of the amplifier. At all signal levels there would be more areas for distortion other than clipping to occur, as there are more active devices in the signal path and every one has some form of nonlinearity.

In a single-ended amplifier the output is symmetric until you reach the limit of the current source. During most recorded music the drive becomes push only on loud bass passages. At that time the woofer cone can reach a 10 to 20dB greater output than from most push-pull configurations of the same power rating. If the loudspeaker took the power in and the cone didn't move, the inductance of the voice coil would cause the voltage to increase until it found a place to go. That is why there are diodes from the output lead back to the power supply in most solid-state amps.

Treating the cone as a moving mass without any electrical conversion is not a good model, as obviously there would be no cone movement from the start. Also any losses from the mechanical movement would still have compensations from the amplifier. The only time greater power would be needed is to stop the cone suddenly. The good news is that in music plosive starts are common, instantaneous silence is not.

Push-pull topologies are great for driving resistive loads. Unfortunately, they have distortion artifacts throughout the audible spectrum that are more unpleasant to the ear than a single-ended design. Their advantage is greater power out for each output device when they are used as Class AB. The question is, "Does greater power with a noticeable form of distortion sound better or worse than less power with a distortion that may occur more often but is less objectionable?" Most folks who listen to a single-ended design clearly prefer them when the power is equal. *aX*

Back

Show Report from the Netherlands

A couple of interesting trends were in evidence at this recent Dutch show.

By Jan Didden

PHOTO 1: Audio amateurs trying to absorb the fine points of digital filtering.

The third annual Dutch DIY Audio meet was held this year in early March[1]. After a hesitating start in 2004, the Dutch show has quickly established itself as an important event for serious audio amateurs from the Netherlands, Belgium, and Germany. This year's event was attended by about 200 participants, many of them bringing their equipment to show and demo in the available rooms.

It may seem preposterous to compare such an amateur event with the prestigious Audio Engineering Society (AES) conventions, but for me there were two striking similarities. First, apart from the actual demos, there was an active schedule of briefings and lectures, with two or three in progress at any one time. Which posed a familiar problem: Which one do you attend, while trying to shoehorn in as much exposure to the interesting ideas and research topics?

PHOTOS 2-4: A few of the open-baffle dipole speakers on demo at the show.

Second, many serious amateurs firmly abandoned a focus on the umpteenth current source topology and how to bias a LED, turning instead to a professional systems point of view. That is not to say that current sources and LEDs are not important; I cut my own teeth on those same issues, and it is a step that cannot be skipped. But modern audio technology developments are possible with the increasing availability of mathematical analysis tools and simulators, many of them available as student versions or even for free, such as the powerful free Spice simulator from Linear Technology. The lecturers outlined two developments in DIY audio that rely on modern tools.

BRUTE FORCE?

Mainstream audio systems make increasing use of digital signal processing, and one area that has attracted much DIY attention is digital filtering and crossover design. Digital crossover design takes the digital signal from the CD/DVD player (S/PDIF) and uses a digital signal processor to divide the signal components into three bands (for a three-way system), sending each band to a dedicated DAC, each feeding a dedicated power amp. The filtering is basically a software-driven process and can be changed and tuned *ad infinitum* with little more than a PC. Two general classes of filters are the Infinite Impulse Response (IIR) and the Finite Impulse Response (FIR) filters[2].

I am not a digital filter expert, but my understanding is that the IIR lets you implement analog filter characteristics in the digital domain. That is, using a digital processor, you do some math on the data, then send it to a DAC and out comes the signal as if it has been going through, say, a 24dB/octave Butterworth filter. But all this is done in the digital domain and can even be changed on-the-fly.

FIR filters are even more powerful in that they can arbitrarily and independently set bandwidth, slope, delay, impulse response, and so on. Thus, you can have a 24dB/octave low-pass filter with zero phase shift and constant delay. The advantages are obvious.

Response deviations or differences in group delay between individual drivers can easily be compensated. Room resonances can also be compensated for. Disadvantages of analog filters, such as unavoidable phase shifts, delay variations, and non-symmetrical impulse responses, can completely be avoided (if you use FIR filters). With the continuing fall in price of integrated processors, digital implementations are coming within the reach of audio amateurs.

There were two interesting lectures at the DDIY 2006 on digital crossover filtering. Simon-Thijs de Feber selected the AL3101 processor after looking at several candidates, and has all but finished a modular three-way crossover based on IIR in an AL3101 DSP feeding into a trio of AD1854 DACs. Gerard Jansen took the other route, and has built an FIR implementation on the EMU10K1 processor on a Soundblaster Live! card. The software used to program the EMU DSP is called BruteFir, runs under Linux, and is essentially free!

WHICH POLE?

The second interesting topic, which generated much interest at the DDIY 2006, was dipole speakers, which feature no real enclosure but just a baffle with the drivers, so these radiate not only to the front but also to the rear. Hans Laros presented an in-depth design process using (again) analysis tools such as MathCAD, LspCAD Pro, and others. Many actual designs were demonstrated, and invariably what caught people's attention was the total absence of "enclosure sound."

Attention in dipole speakers has fluctuated in past years, but one obvious disadvantage was that at certain frequencies, there would be an acoustic short between the rear- and front waves, leading to a response dip. The resulting on-axis frequency response thus resembled a comb filter. However, with active compensation and using high-excursion drivers, excellent results can be obtained, as has been demonstrated by Siegfried Linkwitz of Linkwitz-Riley filter fame.

At the show, we were able to compare dipole speakers with an expensive closed box, and the differences were glaring and not at all subtle, with the dipole having very low coloration and strong articulation, especially in the low-frequency range. This is a development to be watched.

IN CLOSING

As noted, today's audio DIY is really state-of-the-art, with amateurs having the advantages of low-cost advanced tools and parts, without the constraints of having to produce a competitor in today's cutthroat market. I will surely be present again at DDIY 2007; maybe I can even demo something of my own. You see, I have this advanced current source to bias this LED. . . *aX*

REFERENCES/FOOTNOTES

1. Normally DDIY is held in early April, but because the organizer and his wife were expecting a baby, the event was moved forward to avoid interfering with Mother Nature herself. We welcomed Sarah-Ysabel in early April, proving that audio DIYers do have a life!

2. www.dspguide.com/pdfbook.htm.