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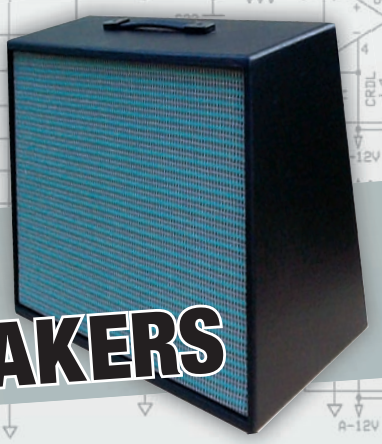

Rock On With BIG-OUTPUT SPEAKERS

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The XF410

Super-size your big output appetite with this speaker featuring four ten-inch drivers.

PHOTO 1: The XF410.

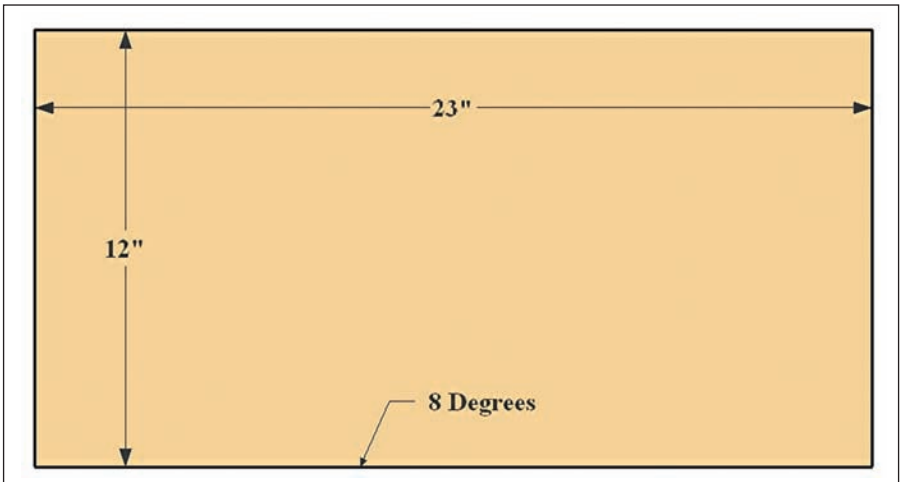


FIGURE 1: Outside bottom.

The cross-firing XF210 guitar cab (*aX* 12/08) has garnered some rave reviews, some of which you can view at http://billfitzmaurice.com/phpBB2/viewforum.php?f=24. But while a 2 × 10 should have plenty of output, there are those who aren't satisfied with less than total overkill. For them, the XF410 (**Photo 1**) is loaded with four tens.

Drivers come in many varieties, but by far the overwhelming choice of most manufacturers, and thus most players, is Eminence. To choose one that suits your needs, go to www.eminence.com. You can get Eminence drivers, and the other hardware for your build, from www.speakerhardware.com.

CUT-OUTS

The cab is made entirely from ½″ plywood. Baltic birch is nice, but any void-free grade of plywood of at least five plies is adequate for the job. Secure the joints with drywall or cabinet screws, or a pneumatic brad nailer. The joints in the cab must be airtight, so you should use a polyurethane-based adhesive, which expands when curing to fill any gaps. Polyurethane construction adhesive, such as PL Premium, applied with a caulking gun, works great and is inexpensive. Urethane woodworking glues are fine as well, but can be very expensive.

All measurements provided are somewhat approximate, being influenced by the actual thickness of the materials used and by the accuracy of your construction. You may pre-cut the parts slightly oversize for manageability, trimming as required to actual finished size as you install them. Always measure directly on the cabinet where each piece is to be installed to verify the true size required.

Start by cutting out the bottom (**Figs. 1** and **2**). The first diagram shows the

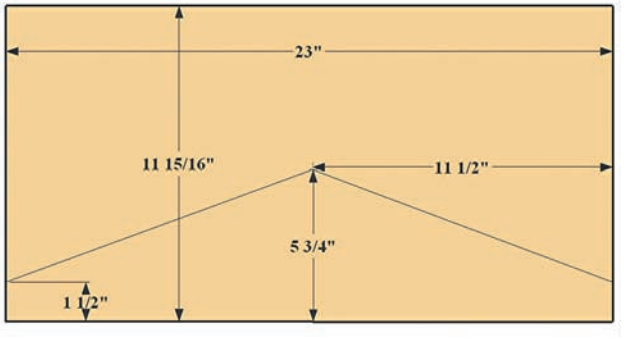


FIGURE 2: Inside bottom.

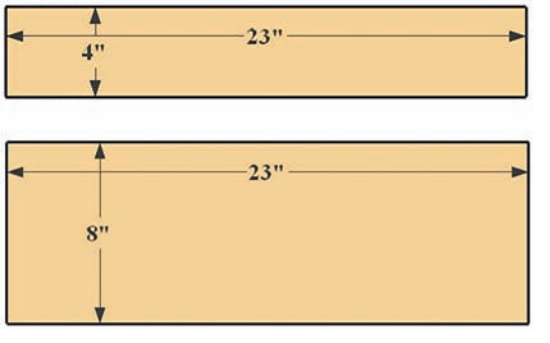


FIGURE 3: Cutting upper and lower back pieces.

underneath/outside view; the front edge is cut back at an 8° angle. The second diagram shows the inside view, with reference lines drawn at the joints of the baffle half leading edges. Cut the upper and lower back pieces (**Fig. 3**). Use these back parts in an open back or convertible cab. If you're building a sealed cab with no removable section, cut the back as a single piece measuring 23″ × 23″. Cut two sides (**Fig. 4**).

Cut the mirror-imaged baffle halves in four steps, with the basic blanks first (**Fig. 5**), and then the top and bottom edges at an 8° angle, both angles facing the same direction, forming the back tilt. As soon as you cut them out, label the two baffle halves left and right, front and back face, top and bottom. Draw lines on the baffle halves per **Fig. 6**, cutting them with the blade at a 20° angle, so that the front faces are narrower than the rear faces on the cut line. Draw lines per **Fig. 7** and cut at a 20° angle, this time with the blade tilted so that the front faces of the baffle halves end up wider than the rear faces.

Draw lines and mark two points per

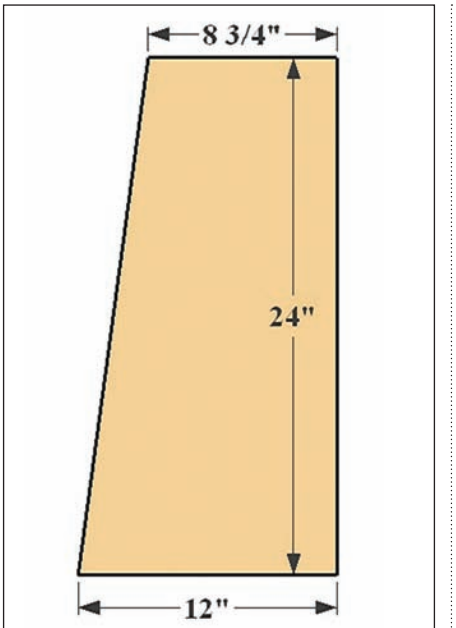


FIGURE 4: Cutting sides.

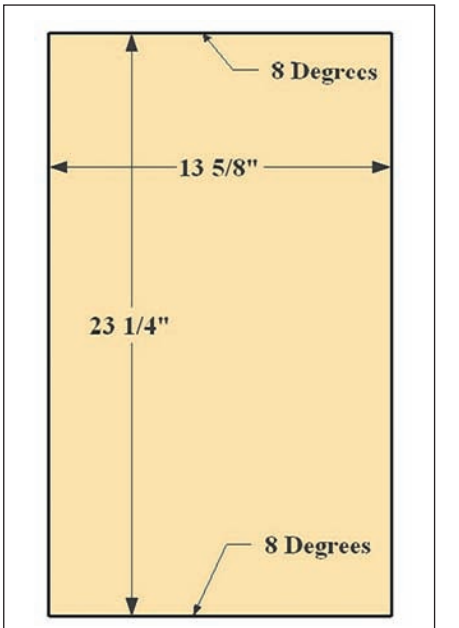


FIGURE 5: Cutting basic blanks.

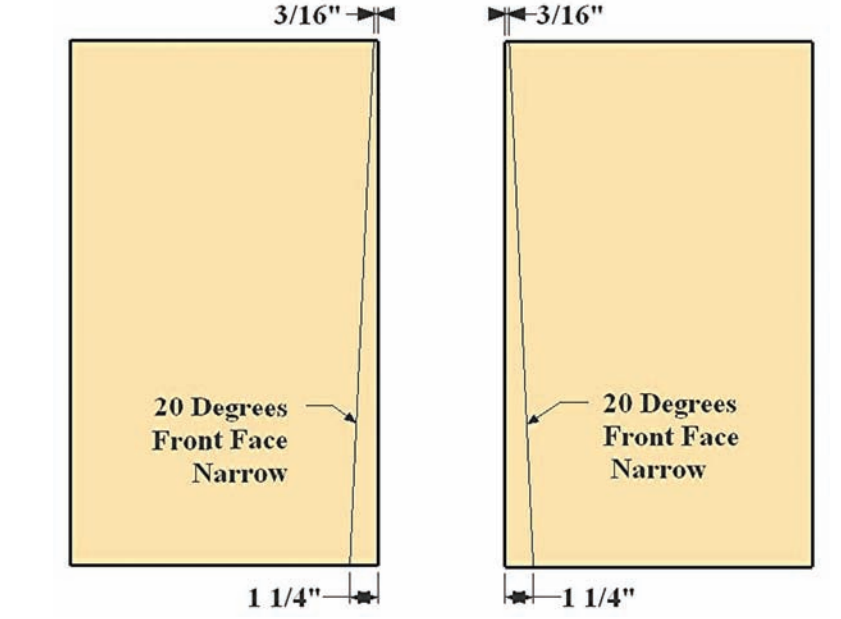


FIGURE 6: Front face narrow on two baffle halves.

FIGURE 7: Front face wide.

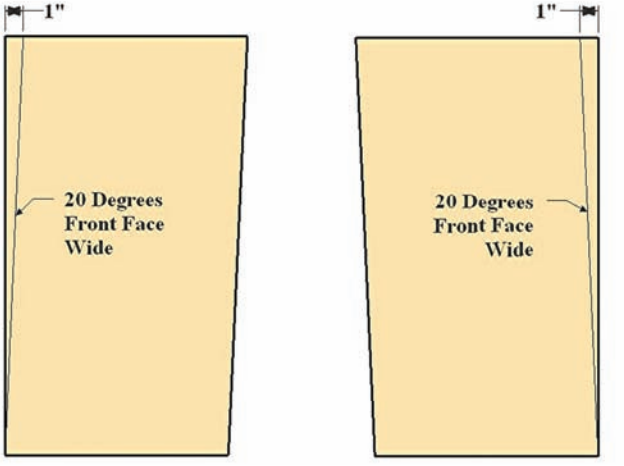


Fig. 8. The points are centers for the driver holes which you must cut. The driver holes are sized according to the driver specification sheets. Most tens will have about a 9″ hole. Mount the drivers to the baffles from the front and attach them with either screws or bolts and T-nuts. Insert a driver through each mounting hole, marking through the frame the locations for pilots. Remove the driver and drill the pilots. Repeat with the other baffle half. Install T-nuts from the rear of the baffle, pulling them through with a bolt driven by hand.

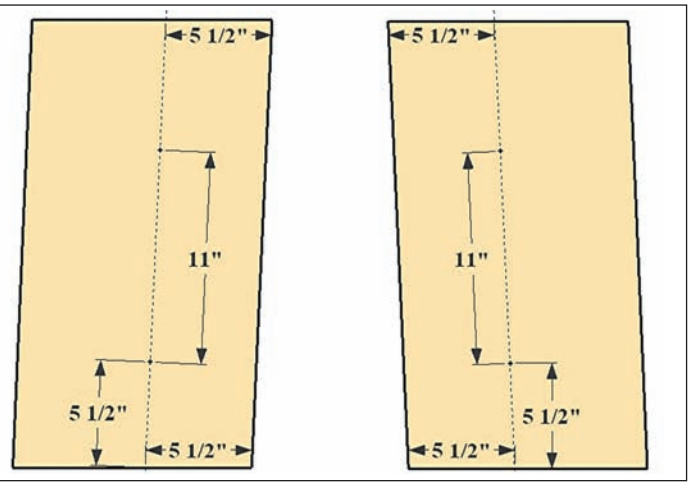
ASSEMBLY

Cut three assembly jigs. The first is an approximately 18″ long piece of plywood, the end view shown in **Fig. 9**. The other two jigs are identical, both cut from plywood per **Fig. 10**. Attach one side to the bottom, with the side overlapping the outer edge of the bottom.

To fasten, hold the two pieces in alignment with a 2 × 2 caul and clamps (**Fig. 11**). Clamp assembly jig 1 flush to the leading edge of the side, with the angled edge matching where the baffle half will mate to the side. Attach the first baffle half, holding it tight to the jig while driving fasteners through the side (**Fig. 12**). Line up the baffle half with the reference line on the bottom while driving fasteners through the bottom. If the baffle halves are warped, cut an 8° edge on a 2 × 2 caul and clamp or screw it to the reference line, then clamp or screw the baffle half to it to pull the warp out before fastening the joint to the bottom.

Screw assembly jigs 2 and 3 to the first baffle half, near the top and bottom. Attach the second baffle half to the assembly, screwing it to the jigs (**Fig. 13**). Place wax paper or polyethylene between the jigs and the baffle halves so that they won't be glued together. Install the lower back, or a one-piece back for a sealed cab, with a caul and clamps holding the parts secure (**Fig. 14**). Install the upper back, using another caul and clamps to hold everything secure while fastening (**Fig. 15**). Cut the top, per **Figs. 16** and **17**. Install the top, then remove the guideboards. Install the second side.

FIGURE 8: Measuring for driver holes.



For a convertible cab that you can use either sealed or open back, install inch wide flanges around the sides and top of the back opening (**Fig. 18**). Omit these for an open back. Cut a 1″ to 1.5″ hole in the upper back, offset to one side. Back that hole on the inside with a piece of thin plywood as a jack mounting plate.

For a convertible box, cut a back door (**Fig. 19**) and door flange. The flange is a 1½″ wide strip attached to the bottom of the door, sized wide enough to just fit between the flanges on the cabinet sides (**Fig. 20**). When in place on the cab, the flange goes inside the lower cabinet back, while a pair of latches clamp the door tight at the top. You may use surface-mounted "butterfly" style latches, or inlaid latches for protection.

You may use individual round grilles over each driver. If you prefer a full grille, cut two grille backer sides and one each top and bottom (**Figs. 21** and **22**). The backer sides install tightly to the baffle/side joints, with the backer top and bottom connecting them across the top and bottom (**Fig. 23**). Cut the sides at 8° on both ends, to match the angle of the baffles, as is one edge on the top and bottom. Cut three grille

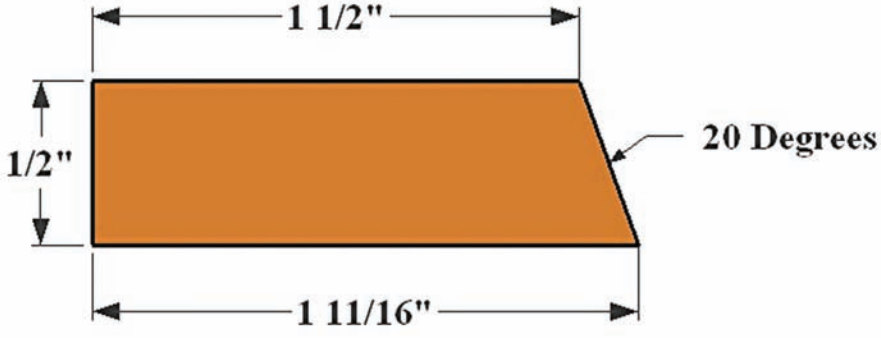


FIGURE 9: Assembly jig 1.

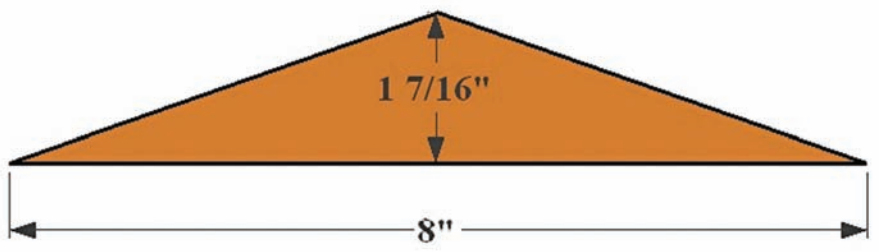


FIGURE 10: Dimensions for assembly jigs 2 and 3.

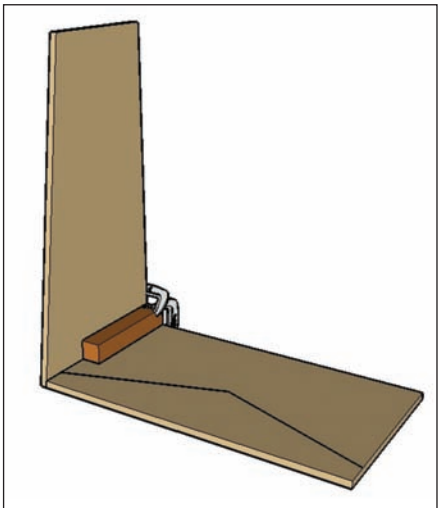


FIGURE 11: Clamping assembly jig.

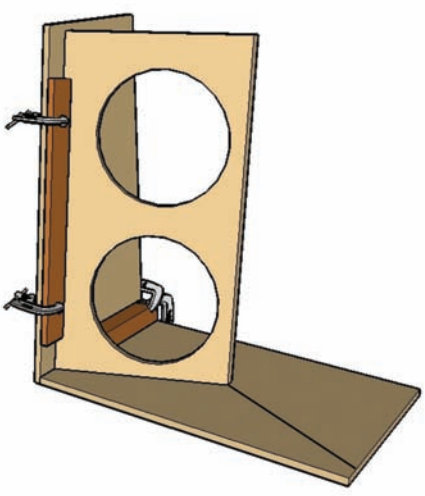


FIGURE 12: Attaching first baffle half.

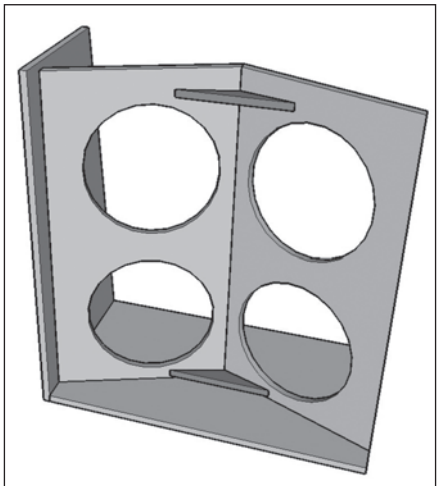


FIGURE 13: Screwing second baffle half to the jigs.

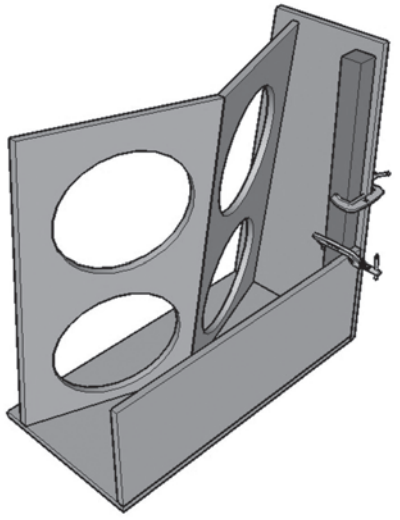


FIGURE 14: Installing lower back, holding parts secure.

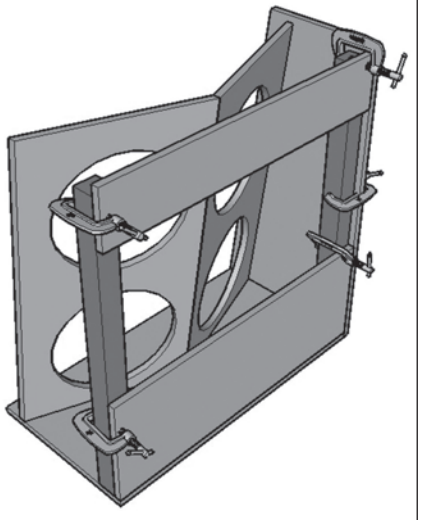


FIGURE 15: Fastening.

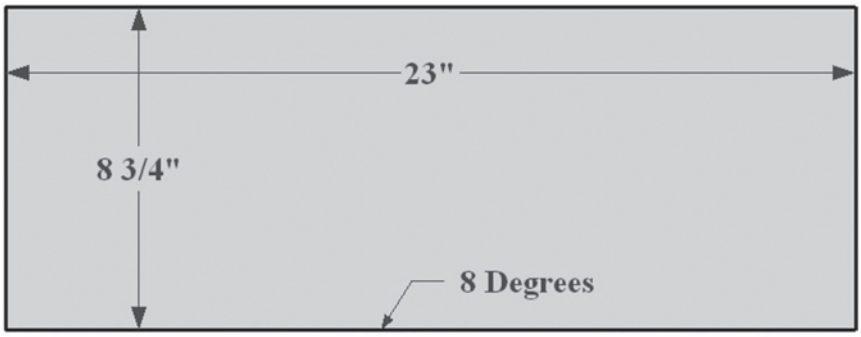


FIGURE 16: Cutting the top.

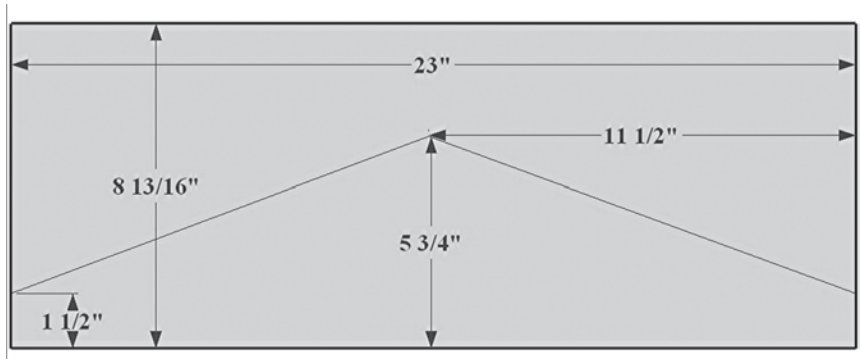


FIGURE 17: Dimensions for top cut.

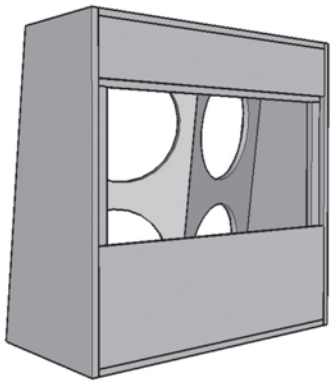


FIGURE 18: Installing flanges around the sides and top of the back opening.

frame verticals and two grille frame top/bottom (**Figs. 24** and **25**). Assemble them (**Fig. 26**), ensuring the top and bottom angles face the right way to fit onto the cabinet. You can assemble the grille with glue only, clamped until it sets, but a biscuit jointer is the ideal tool for this job.

FINISHING UP

Apply a finish. Carpet is a durable option, while Tolex or tweed offers a vintage look. For a modern touch use DuraTex, which you can roll or spray on, and is also available from www.

FIGURE 19: Cutting a back door.

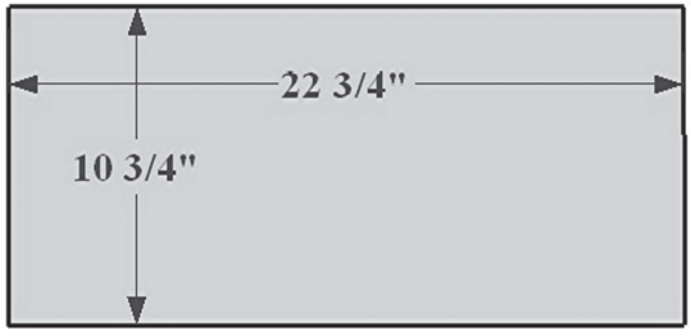
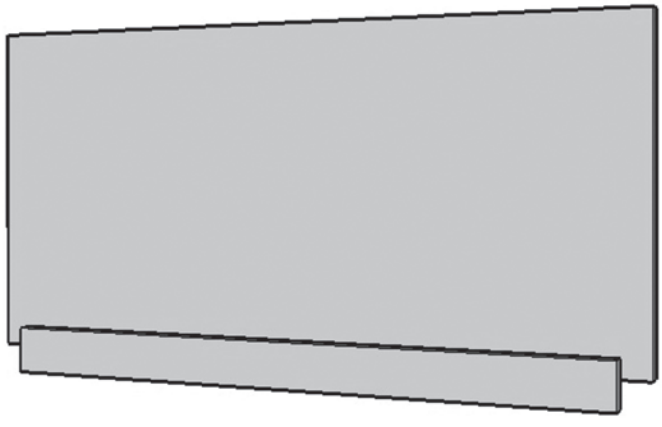


FIGURE 20: Positioning the door flange.



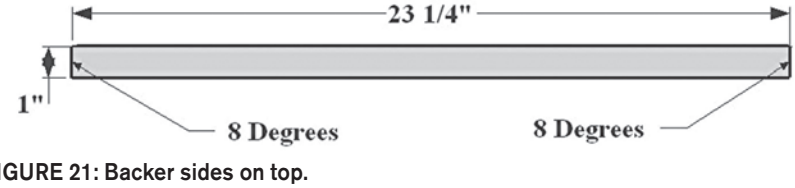


FIGURE 21: Backer sides on top.

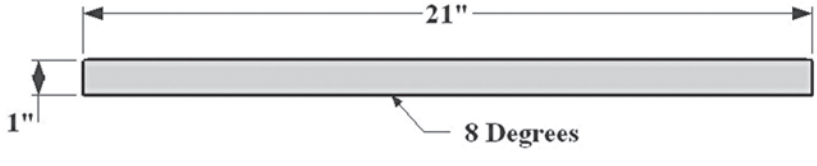


FIGURE 22: Backer sides on bottom.

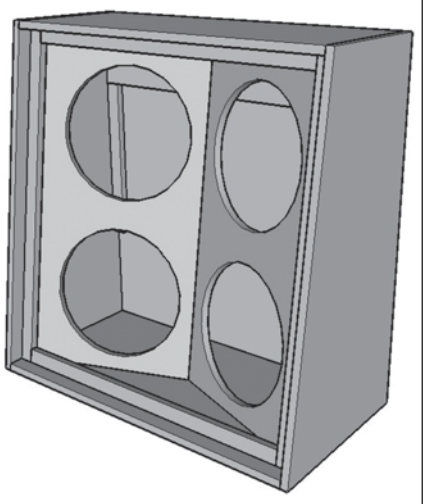


FIGURE 23: Installing backer sides.

speakerplans.com. Attach grille cloth to the grille. Use foam speaker gasket tape on the convertible back flanges for an airtight rattle-free fit. Fully line the inside of the cabinet with a 1″ to 1½″ thick acoustic foam or open-celled polyurethane mattress-topper foam, secured with staples or adhesive. Install handles and protective corners as desired. Install grille-securing hardware to the grille and backers, or use screws with finish washers driven through the grille face. Install the jack. Wire the drivers to the jack. For impedance equal to one driver's impedance, wire them series/parallel; for impedance equal to one-quarter of one driver's impedance, wire them in parallel. Install the drivers and the grille. Rock on. *aX*

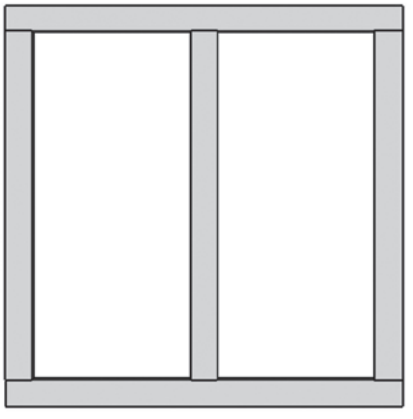


FIGURE 26: Assembled grille frame.

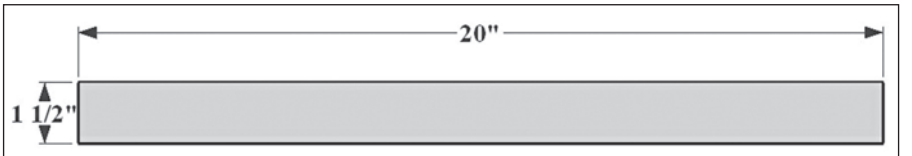


FIGURE 24: Grille frame vertical piece (3).

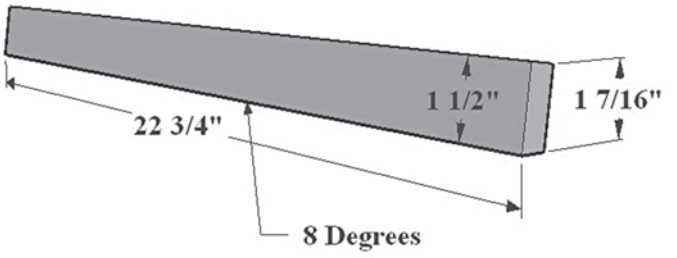


FIGURE 25: Grille frame horizontal piece (2).

Major parts nominal sizes, oversized for rough cutting. Confirm all final parts sizes per plans text.

Bottom	2¼″ × 23¼″
Sides	12¼″ × 23¼″
Lower Back	8¼″ × 23¼″
Upper Back	4¼″ × 23¼″
Baffle Half	14″ × 23¾″
Top	8 3/8″ × 23¼″
Back Door	11″ × 23″

ezDAC

The author presents an easy-to-build—and good-sounding—upsampling digital to analog converter.

PHOTO 1: The DAC.

I had wanted to build a DAC for a long time, but other audio duties kept me from doing so. After some on-line searches, I was able to find many do-it-yourself kits, as well as some assembled DACs (mainly from Asian sellers) at such cheap prices that I was tempted to give them a try. However, soldering and building DIY projects acts as a kind of meditation for me, recharging my mind. So, I definitely wanted to assemble a DAC rather than buy a ready-to-play one.

I first needed to decide between oversampling or nonoversampling topology. Some audiophiles believe upsampling is the wrong way to go, but in the end, it is a philosophical decision. Anyway, I chose upsampling.

My next question was: Which DAC chip to employ? I had the chance to listen to the Accuphase DP57 CD player, which I liked very much and which uses the TI PCM1796 (www.ti.com), so I decided on either this family of chips or the AD1853 (www.analog.com), which is used in another Accuphase player. Both those Analog Devices and Texas Instruments DACs have differential current output, leaving full freedom in choosing the current to voltage conversion (I/V) circuit.

After much web surfing, I ended up with two DAC options: the Analog Devices 1853 evaluation board (now replaced by the EB1955), which, at the time, came at a cheap price. The second option was to buy the ezDAC PCB by Evan Zamir. Because I wanted to gain some experience with SMD components soldering and to be free to choose the components to be used, I decided on the ezDAC (**Photo 1**).

Evan is a professor of mechanical engineering at Georgia Tech University, but most of all he is a DIY audio enthusiast who decided to produce a small number of PCBs just for fun and passion, without any commercial gain. So I contacted him and ordered the PCB, plus some spare components. I waited a long time to receive the box, due to postal delay, but when it finally arrived I was impressed by the quality of the board. You can find more information on Evan's website: www.ezdiyaudio.com.

The PCB I used is version 1.0, although the upgraded v. 1.5 is available. These boards are similar, with the newer one including some jumper-able power rails (which allow easier modifications to the regulator section) and the addition of solder mask-silkscreen layers. The ezDAC architecture is shown in **Fig. 1**.

The S/PDIF digital input enters a 1:1 pulse transformer, which offers some galvanic isolation to the noise present in the input signal. The quality of the transformer is very important here. The DIY audio community recommends the used model: the Newava S22083 (www.newava.com) alias Schott 22083. Making a good pulse transformer is an art, because some compromises exist; one parameter that has a big impact on the performance is the *leakage inductance*, the lower the better.

Because the Newava is hard to find in Europe, while US DIYers can find it at Digi-Key (www.digikey.com), I investigated some possible substitutes, finding the Umec UT14013 and the C&D 76601/3 as valid candidates, though with different PIN dispositions. I tried to buy them from a local distributor and reseller, with no success. So I e-mailed the manufacturers asking for a sample, explaining to them my project and a follow-up article, but getting the same result: zero. So, unfortunately, I can't provide a pulse transformer comparison. Last minute note: While writing this article, I checked my usual components resellers again, and discovered that Farnell (www.farnell.com) now carries the Murata 76601/3, rebranded model of the C&D one. I added this to my "things to do" list.

The receiving stage consists of the Cirrus Logic CS8416 (www.cirrus.com), a widely used chip, followed by the asynchronous sample rate converter (ASRC), which samples the input signal to 96kHz. You can choose between the AD1896 or the Texas Instruments SRC4192 as ASRC. An external crystal oscillator generates the clock for the ASRC and the DAC, bypassing the CS8416 clock recovery.

The conversion stage is performed by the Texas Instruments PCM1794 or the PCM1798, two of the best-sounding DAC chips on the market. The PCM1798 is cheaper, but it is a little bit noisier and outputs less current than its bigger brother. As I said before,

the PCM1794/8 outputs current, so the needed I/V conversion is done using a simple resistor on each signal leg.

The analog output stage converts the differential signal, coming from the I/V stage, to unbalance and to amplify it. To do this job the AD8610 is used, but any other PIN-compatible integrated IC can be used, such as the National LME49710 (www.national.com) or the TI OPA134. A high-quality capacitor, placed on the signal path, applies a mild filtering.

The power supply section consists of four voltage regulator chips. The variable LM3x7 type is preferred over the fixed one for its better *ripple rejection*. Some modifications are possible in this section as follows:

The basic configuration needs to connect an external filtered ±15V DC supply to the ezDAC, with the onboard regulators converting the voltage to 3.3, 5, ±12V DC. The second step—the one I utilize—is made by using a separate power supply to feed the digital and analog sections of the ezDAC, leaving the onboard regulators to do their jobs. The resulting improvement in the use of different transformers and filtering for the analog and digital supply is noticeable, so I highly recommend it. To make it possible, you need to cut a board trace as indicated by the black arrow in **Photo 2**. With the PCB version 1.5 you don't need to cut anything, because a jumper is present for the purpose. The third possibility is to completely bypass the onboard regulators by providing a clean, low impedance source of regulated power: for example, you could employ Jung's super regulator (*AE* 4/2000).

EZDAC CONSTRUCTION

As usual, the first components to be soldered are the smallest, in this case, the surface-mount ones, preferably starting from the center of the board. Using tweezers, a magnifying lens and some flux, and applying a small amount of solder on one of the two pads I first placed the SMD resistors and capacitors over the pads and heated the already soldered pad. Once done, I applied (with the tweezers) a small amount of pressure on the part and re-heated the pad to be sure that the component was flat against the PCB. Finally, I soldered the other side of the part. I checked the nearby traces with the ohmmeter to be sure everything was properly connected.

The surface-mount chips are a step forward in difficulty, especially the PCM1794 and the AD1896 that have a SSOP package. I prefer to put a small amount of solder on two opposing corner pads, then place the IC over the pads and heat one of the pads with solder, adjusting the component so that it lines up with all the pads. Then I soldered the second pad applying some pressure so that the chip was flat and aligned. With this done, I soldered the remaining pads.

I inspected the work—with a magnifying lens and the ohmmeter—to ensure there weren't any unwanted shorts and for the proper pin-pad connection. This is the most time-consuming part of construction, but ensures an almost trouble-free first power-on of the DAC. I found it very helpful to utilize some flux (be sure to remove the residues because some kinds of flux are corrosive) and a solder with 2% of silver. For those who

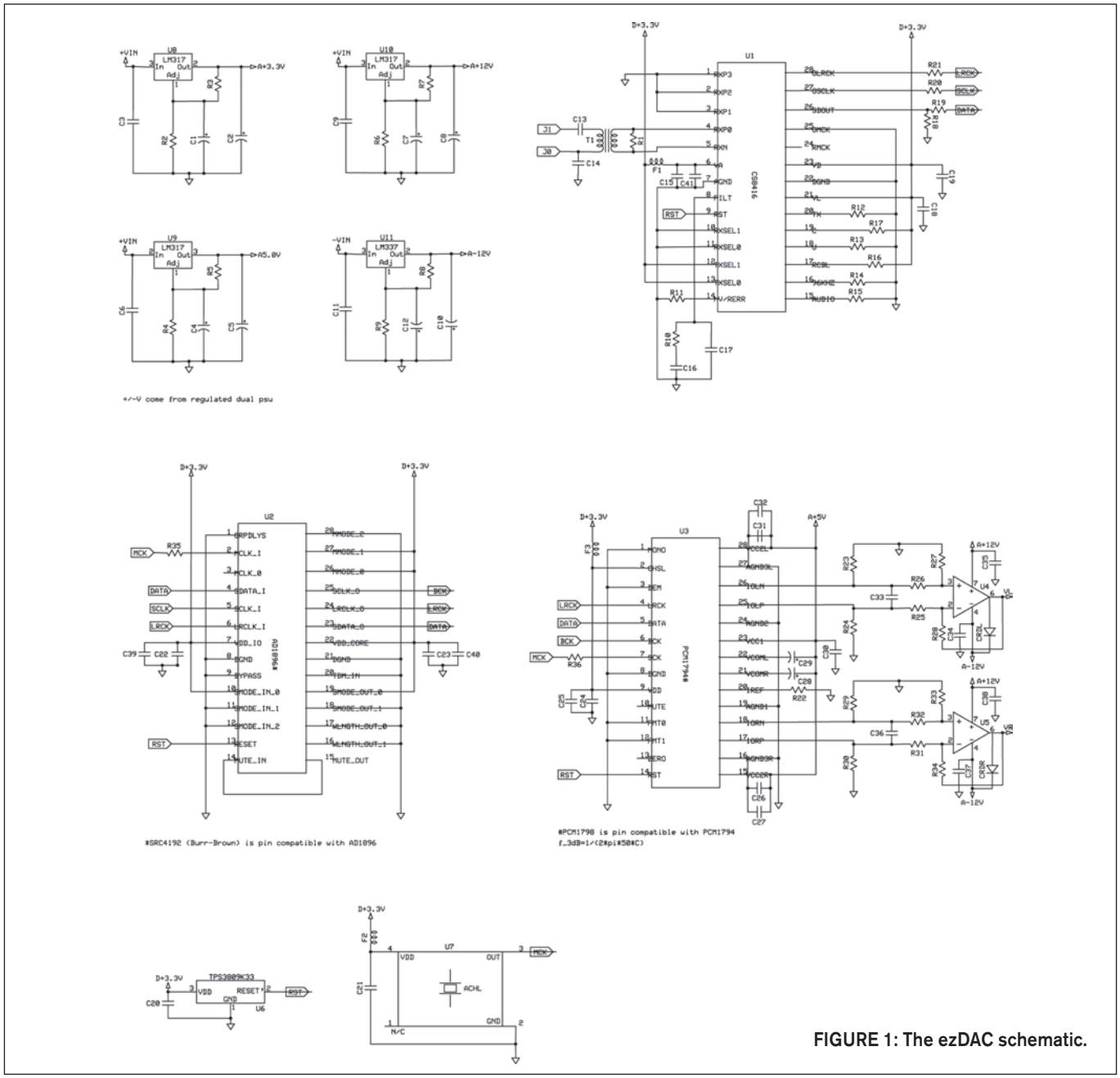


FIGURE 1: The ezDAC schematic.

believe silver solder makes better-sounding devices, the real aim of the silver is to facilitate the soldering of SMT parts, because these components come with silver-plated leads. I used the Sn96.5/Ag3.5 eutectic alloy.

After finishing with the surface-mount parts, it's time for the standard through-hole ones, starting with the resistors, followed by the capacitors, the quartz, and the pulse transformer, and ending with the four voltage regulators. You are free to choose the most exotic parts, if the dimensions and application are correct. For example, I was careful with the ±12V DC regulators adjusting resistors (R6-7, R8-9) which I matched to get the same voltage output. For the adjustment pin bypass capacitors, I used low ESR (Equivalent Series Resistance) ones such as the Sanyo SH and the Panasonic FC, while the output caps (C2, C5, C8, C10) are generic with a significant ESR, because this resistance is used as a filter to prevent resonance. Finally, I preferred a polystyrene capacitor in the signal path (C33-36). The finished ezDAC board is seen in **Photo 2**.

THE POWER SUPPLY

The schematic of the power supply I used is in **Fig. 2**. The 230V AC is filtered by a varistor, for surge suppression, and by an X2 class capacitor for RFI (Radio Frequency Interference). The MOV (metal oxide varistor) is a pretty useful component, because it protects the power supply from quick transient voltages by absorbing those dangerous pulses.

It comes in a variety of clamping voltages, with 275V AC the common value used in Europe, and 150V AC in the US. Its case is button style, ranging from 5 to 20mm diameter, with the bigger one capable of absorbing more energy. The main disadvantage of the MOV is that when it fails, due to a too long overvoltage, it doesn't show up, leaving the user in doubt whether or not it is protecting. Moreover, in extreme situations, the varistor can ignite, as reported

by the *National Fire Protection Association*, producing a dangerous situation.

My suggestion is to use the Littlefuse TMOV, which integrates a thermally activated fuse, designed to open in case of overheating, thus preventing it from catching fire and failing. When the temperature goes down again, the TMOV reactivates itself. Littlefuse produces an even more advanced type, named iTMOV, which allows you to monitor the functionality of the varistor through an LED. Mouser (www.mouser.com), Digi-Key, and Farnell carry these Littlefuse varistors.

Two transformers are utilized to supply the analog and digital section; when choosing the secondary output keep in mind that the LM3x7 regulators need an input voltage at least 2V DC higher than the desired output to work properly. I placed a film capacitor between the transformer secondary and the bridge rectifier, which reduces the voltage transient due to the diodes switching. Each bridge diode is bypassed by a ceramic cap (X7R type), whose function is to minimize diode noise.

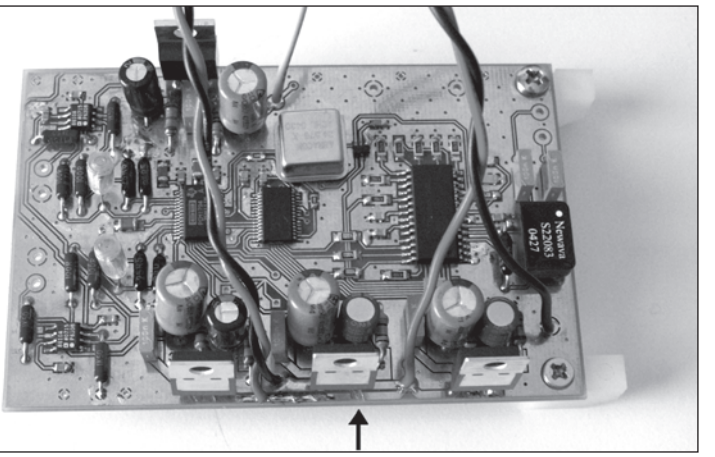
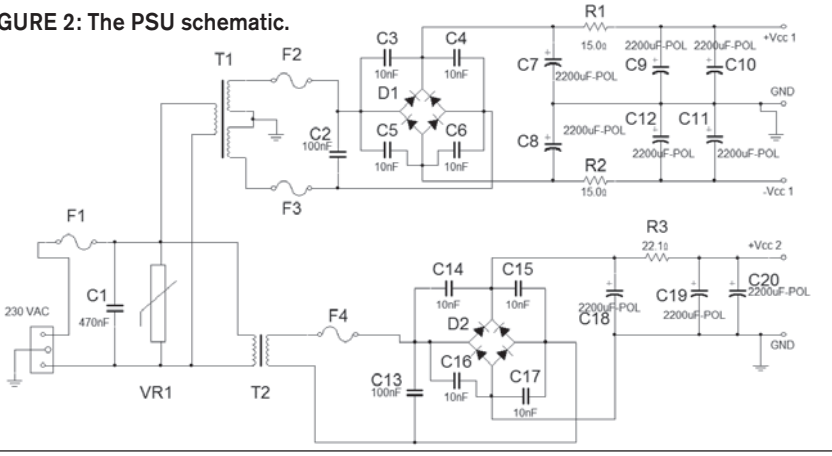


PHOTO 2: The ezDAC completed PCB; the arrow indicates the cut trace to separate the analog/digital supply.

FIGURE 2: The PSU schematic.



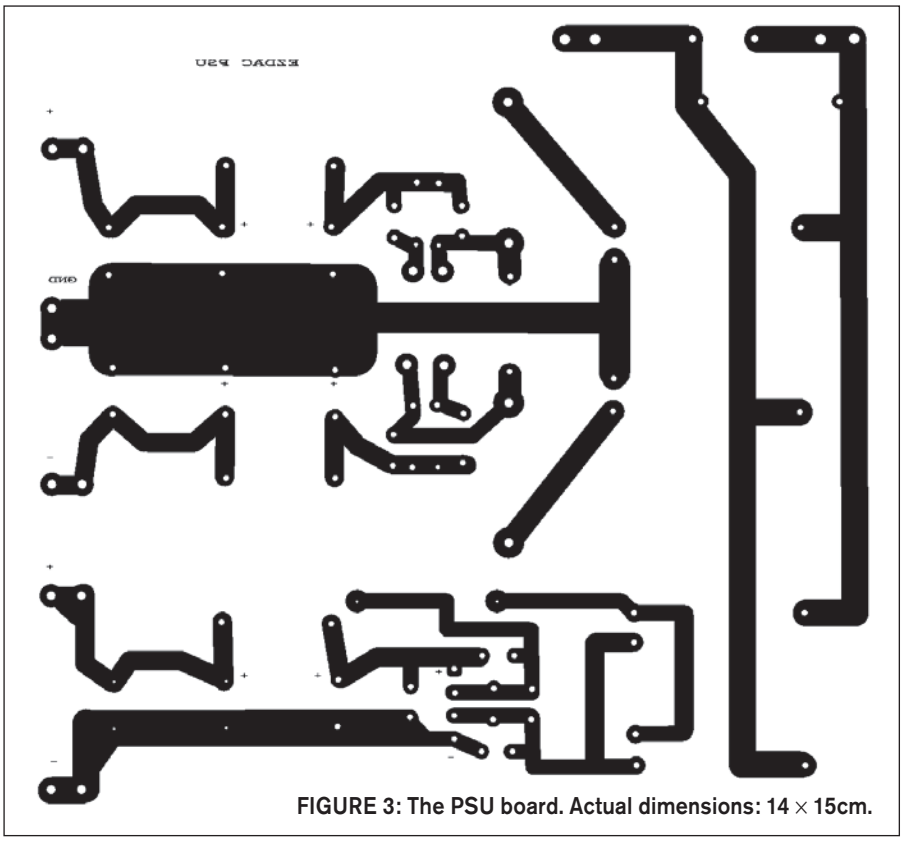


FIGURE 3: The PSU board. Actual dimensions: 14 × 15cm.

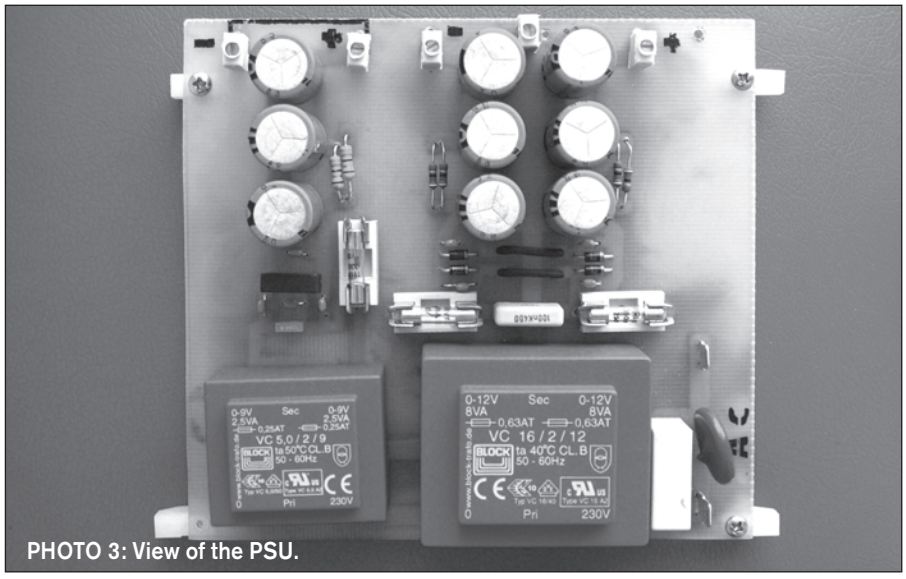


PHOTO 3: View of the PSU.

TABLE 1

REFERENCE	DESCRIPTION	RS-Comp. PN	Farnell PN
R1, R2	Res 15Ω, 2W, 1%	214-1715	9474943
R3	Res 22Ω, 2W, 1%	214-1737	9475109
F1	fuse, 0.5A, 250V, fast	541-4347	1271612
F2-4	fuse, 0.3A, 250V, fast	541-4325	1123182
VR1	MOV, 275V AC		4804697
D1	diodes, 1N4937, 600V, 1A	652-7365	4502358
D2	diode bridge, 100V, 2A, case D44	629-6077	2416189
C1	cap X2 class, 470nF, 275V, P22.5	208-8030	1198299
C2	film cap 100nF, 100V, P15	298-0368	9752820
C3-6, 14-17	ceramic cap, X7R, 10nF, 100V, P5	211-5081	1200397
C7-12, 18-20	elec. cap, 2200µF. 25V. diam. 16mm, P7.5	628-3914	8813027
C13	film cap 100nF, 100V, P5	108-2700	1572134
T1	power transformer, 220/12-0-12V AC, 8VA	347-2688	
T2	power transformer, 220/9V AC, 5VA	347-2537	
3 PCB fuse holder	5 × 20mm, P23	417-098	1162740
PCB terminal block		548-7250	
PCB male faston	6.3mm	534-834	

The filtering section is made by paralleling three 2200µF electrolytic capacitors, or 6600µF per rail. Using multiple caps instead of a single one helps to reduce the total ESR. The resistors R1-2-3 form an RC circuit to further improve the filtering capabilities of the power supply (I used two resistors in parallel because I already had them). The PCB copper side is depicted in **Fig. 3**, while the completed supply is illustrated in **Photo 3**. The used components are listed in **Table 1**: the parts suppliers are Farnell (www.farnell.com) and RS-Components (www.rs-components.com).

ASSEMBLING THE DAC

The case is the GX287 by HIFI-2000 (www.hifi2000.it), an Italian producer that sells abroad, too. The model I used has aluminum sides that can be used to dissipate the regulator heat. Using the inner cave and some screws, I blocked a clip that pushes the three LM317s to the case side, through an aluminum oxide insulator (Fischer Elektronik AOS247), which I preferred in order to get some distance between the PCB and the panel. The dissipation capacity of the case is overkill, with the 3.3V DC regulator (the one that has the hardest job) always cool after many hours of use.

Photo 4 shows the DAC blocked to the case. I then took care of the power supply. Because there are some rails that carry the line voltage, I painted the PCB with transparent paint to achieve some electric isolation.

Moreover, I glued a white plastic sheet to the case bottom. I found both the plastic sheet and the screw clip (used to block the regulators) in an old computer PSU, which I cannibalized for useful components. Once both the PCBs were fixed to the case, I connected the power cables first and the signal cable after. For S/PDIF cable I utilized the coaxial RG179, which has an impedance of 75Ω as required by the Sony Philips Digital Interconnect Format.

The completed DAC and its back panel are shown in **Photos 5-6**. Note that I didn't use any power switch, nor an IEC power connector, but feel free to install them if you like.

Continued on p. 33.

Continued from p. 16.

MEASUREMENT PERFORMANCE

The output DC offset is 2 and 5.5mV for the right and left channel, respectively. The frequency response is represented in **Fig. 4**, showing an almost perfectly flat line in the audio band, with the -3dB frequency located over 40kHz. The channel matching was 0.06dB.

Figure 5 is the spectrum distribution of a 1kHz tone. The harmonics are mainly of the odd type, but of very low value (the third harmonic is at about -120dB). The 200Hz tone, and its harmonics, belongs to the measurement chain, and is present in all the shown spectrums.

The intermodulation spectrum is what you can see in **Fig. 6**, relative to a 19 + 20kHz tone. The resulting IMD is very low, with the tones products staying at an insignificant -102dB or lower.

SUMMING UP

I am happy with the results obtained by this little DAC, which sounds pretty good. I compared the ezDAC to a cheap DVD player and medium-priced CD player, and the audible differences were like night and day, in favor of my last project. The total cost was about 140 euro, which sounds good to me in this time of world economy potential recess. *aX*

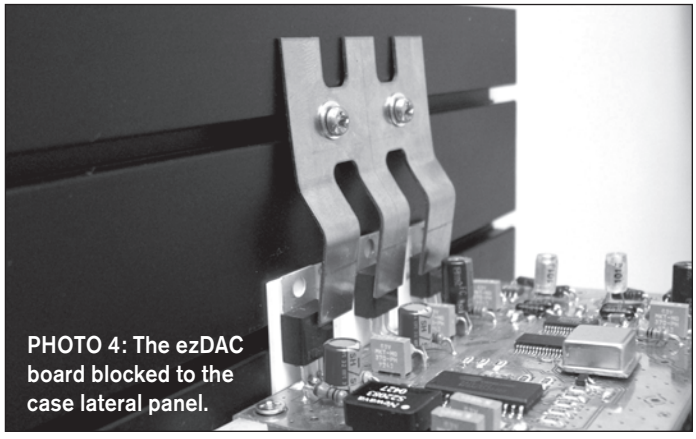


PHOTO 4: The ezDAC board blocked to the case lateral panel.

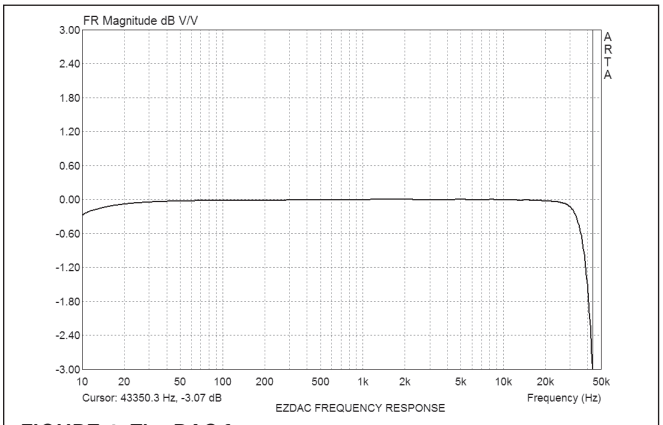


FIGURE 4: The DAC frequency response.

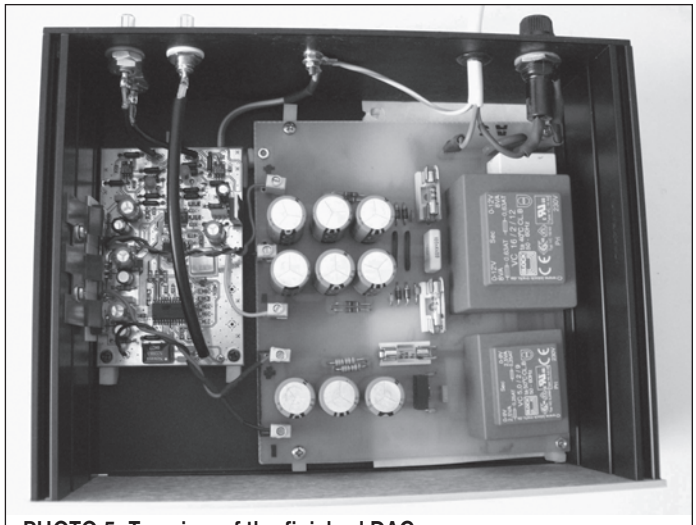


PHOTO 5: Top view of the finished DAC.

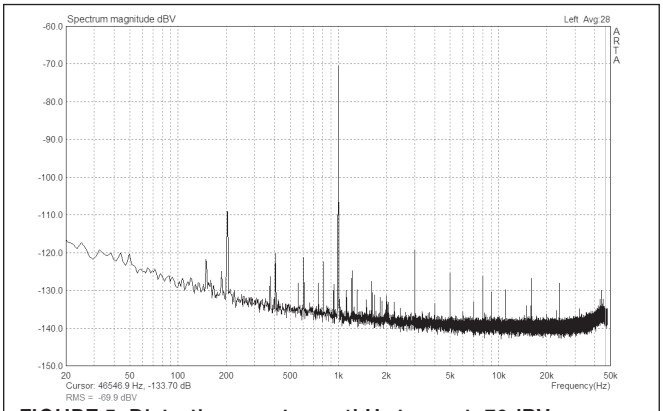
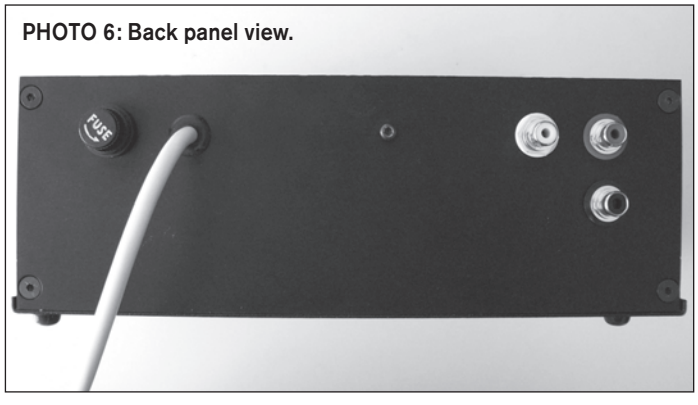


FIGURE 5: Distortion spectrum: 1kHz tone, at -70dBV.

PHOTO 6: Back panel view.



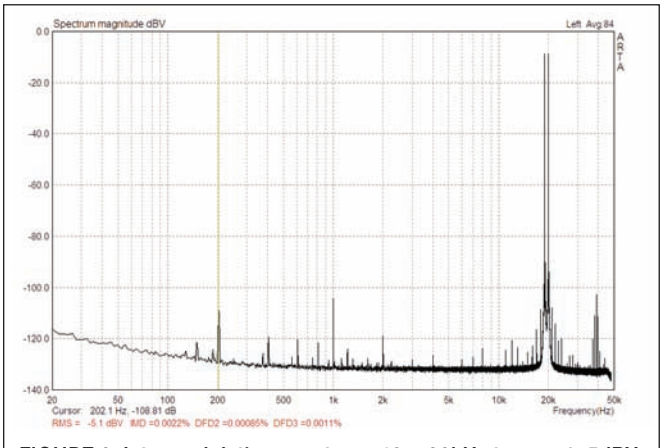


FIGURE 6: Intermodulation spectrum: 19 + 20kHz tones, at -5dBV.

GLASS AUDIO

Nardeux Stereomatic 602 S

This author's first tube amp project is a new version of the Nardeux Stereomatic 302 S Amplifier.

By John Badalamenti

First and foremost, I want to extend my sincerest thanks to Nick Soudas, an *aX* project author himself, for helping me with my first tube amplifier project from beginning to end. I could have not done this alone, and Nick has helped me with schematics, parts, advice, and with all the obstacles I have come across in the building and testing process.

My idea was to build a tube amplifier similar to my recollections of the original Nardeux Stereomatic 302 S (**Photo 1**). The 302 S is a PA amplifier unit containing two power supplies and two monoblock amps using 807 pentodes, a preamp with push-switch type bass and treble controls, and two inputs—one for a Shure microphone for public address, and one for a cartridge pick-up for the dual 1009 turntable with a Shure M3D stereo cartridge. It was a full-range system consisting of four speaker cabinets.

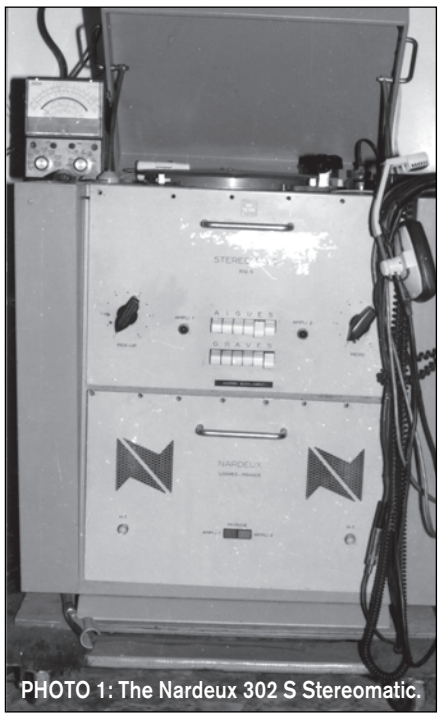


PHOTO 1: The Nardeux 302 S Stereomatic.

PHOTO 2: Author's Nardeux Stereomatic 602 S amp.

The bass cabinet had an 18″ bass speaker and two 7.5″ paper cone tweeters. The midrange cabinet consisted of four 7.5″ full-range speakers with whizzer cones and alnico magnets. This unit was original equipment from Loches, France, that was a part of the Himalaya amusement ride in Coney Island from about 1967 to 1975. Although I had no access to the Nardeux 302 S schematics, I remembered that each amp section power supply consisted of two 866A mercury rectifier tubes and the amplifier section used 807s in the output stage.

After reading Nick's featured article in *aX* May 2001 ("A 211 SE Triode Amp," p. 32), I decided to contact Nick and ask him whether he would be willing to help me with a good design for my project. He said yes, and has been very pleasant and patient throughout. After numerous e-mails over a span of about one year, and many troubleshooting tests, I can finally say that I have an excellent and powerful amplifier (**Photo 2**). There is nothing more exciting than listening for the first time to your own first-built vacuum tube amplifier, in this case, the "Nardeux Stereomatic 602 S" (30W per channel × 2 channels in stereo).

Considering I had very limited electronic test equipment available, I was able to work my way through troubleshooting and testing via the trial-and-error method for the most part. I had no audio generator or oscilloscope at the time to test properly the input and output section of the amplifier. Please keep in mind this was my first tube amplifier project, and I am writing this article from notes based on Nick's help, my own trial-and-error testing, various articles I have read in *aX*, Internet articles, and *Beginner's Guide to Tube Audio* by Bruce Rozenbilt. I actually completed this project in 2002.

The only change I needed to make was to remove two 807s from each channel. Instead of having the desired quad of 807s, I now have only two 807s per channel because it was the only way I could stabilize the amplifier. I was hoping to get about 60W per channel, but settled for about half of that. I believe the original Nardeux 302 S was rated for 100W per channel. I don't have a way of measuring the output power using just the two 807s.

I would appreciate feedback on ideas for possible modifications or improvements. For example, improving parts such as the output transformer, or possibly modifying the circuitry so that a quad of 807s is stable, or using other tubes to output at least 60 to 100W per channel.

THE NARDEUX STEREOMATIC 602 S

This amplifier is based on the Williamson ultralinear push-pull parallel amplifier. Each channel uses two 6SN7s for input, phase splitter, and driver section, and two 807s in the amplifier output section. The power supply uses two 866As mercury vapor rectifier tubes for a full-wave rectifier.

I decided to build two separate dedicated power supplies on one chassis and two mono amplifiers for stereo on a separate chassis. This worked out nicely, but I encountered a lot of trouble stabilizing the amplifier, mostly when trying to use four 807s and the feedback loop resistor/capacitor values. The instabilities included pulsing DC, hum, and noise. By using a resistance decade box and a capacitance decade box, I was able to dial in different value resistors and capacitors simultaneously until the amplifier stabilized with little extraneous noise, hum, or power supply pulsing DC.

I learned the hard way that I needed to use shielded cable from the input (from RCA jack input to input of V1) and shielded cable for the feedback loop itself. The shield being grounded on the input end only helps prevent ground loops. One modification I implemented was to connect a .001µF capacitor from feedback loop to ground to reduce feedback, which resulted in opening the amplifier with more gain and open-airy sound.

Even after all this trial and error, troubleshooting took most of the time in order to get this amplifier stable. This amplifier now sounds very powerful and produces very satisfying music output. I'm sure this amp can probably be improved and tweaked even further with a sine- and square-wave generator using an oscilloscope to optimize the amplifier's sonic characteristics and power output.

WHAT IF. . .

If I had to build this amplifier again, I would build two separate monoblocks, each with its own power supply and amplifier on one chassis instead of the two chassis I used here. I did this because I wanted to basically build a layout similar to the original Nardeux 302 S. I would also use small diameter microphone-shielded cable for all audio signal paths, add volume controls for each channel, and use a choke in the power supply to reduce the high voltage hash that is slightly audible at the speakers. The AC power cables tightly twisted away from signal cables. Matched tubes in the output section also may improve stability and provide equal power output in each output channel. This schematic uses a fixed bias on the 807s.

I chose to keep the integrity of the original Nardeux 302 S by using the 866As in the power supply. Any input from readers is most appreciated, since I plan to build another version of this amplifier in the future.

This project also brings back fond memories of my brother Andrew Badalamenti, who also worked on the Reverchon Himalaya ride in Coney Island and worked the Nardeux Stereomatic 302 S PA music system and the ride during the summer months for many years. It was my brother who got me interested in the Nardeux system and stereo equipment. We both have great memories when we reminisce about the good old days testing, experimenting, and just enjoying classic stereo equipment such as the Nardeux. *aX*

PARTS LIST

REFERENCE	Part
C1, C2, C4	20µF
C3, C11-13	100N
C5-8	220N
C9, C10	240µF
C14, C15	1N
C16, C17	40µF
F1	2A
F2, F3	0.5A
J1	RCA input
J2	120V AC
L1, L2	Lamp
R1, R9, R10	470k
R2	10k
R3, R11, R16, R17	1k
R4, R5, R12, R13	47k
R6	3k9
R7, R8	22k
R14, R15, R23	100k
R18, R19	300
R20, R21	100
R22	150
R24	50
R25	LSK
S5	Power DPDT
TDR1	Relay 45 sec.
T1	Hammond
T1	HT 375V AC
T2	2.5V
T3	Filament 6.3V CT
V2, V1	6SN7
V3, V4	807
V6, V5	866A

VENDORS

Antique Electronic Supply
6221 S. Maple Ave.
Tempe, AZ 85283
480-820-5411
800-706-6789
info@tubesandmore.com

Mouser Electronics
(800) 346-6873
www.mouser.com

Parts Connexion
2885 Sherwood Heights Drive
Unit # 72
Oakville, Ontario
CANADA L6J 7H1
866-681-9602
www.partsconneXion.com

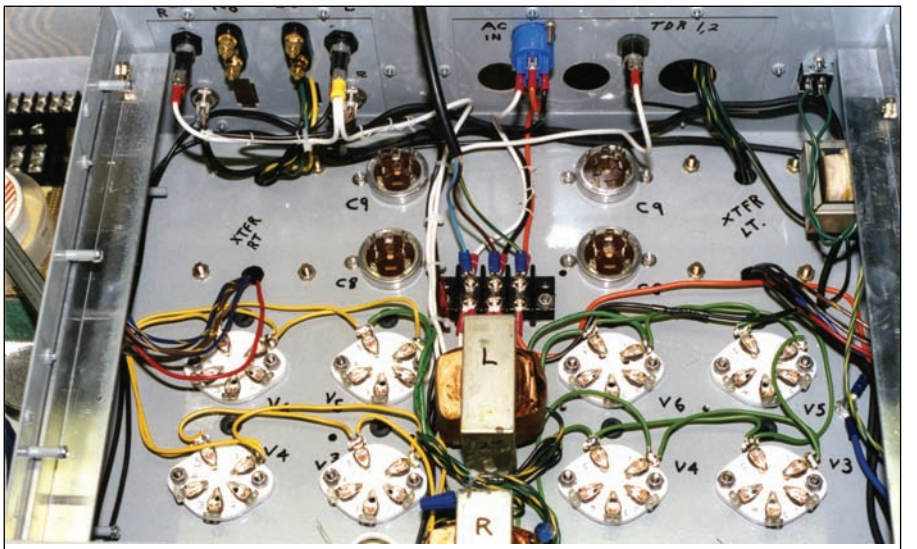


PHOTO 3: Inside the amp.

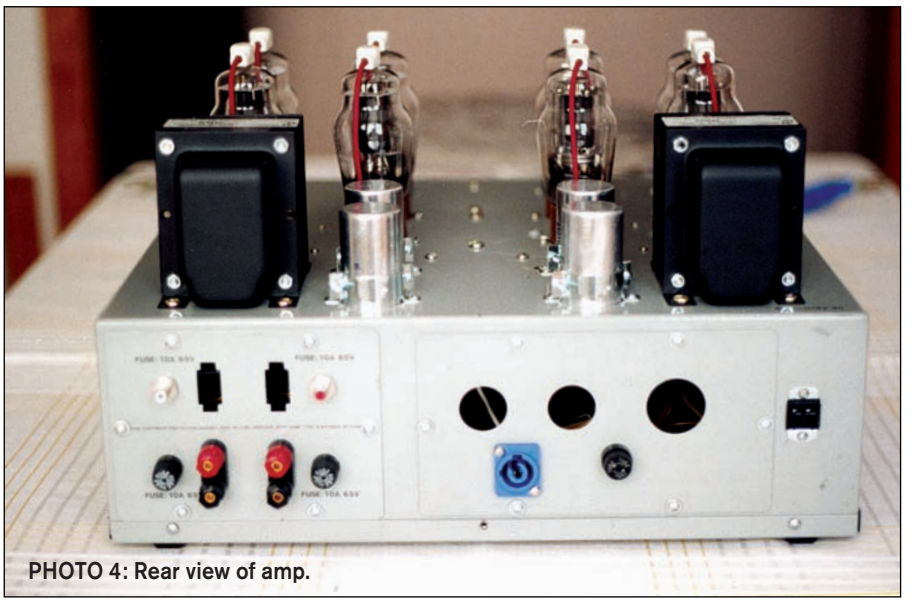


PHOTO 4: Rear view of amp.

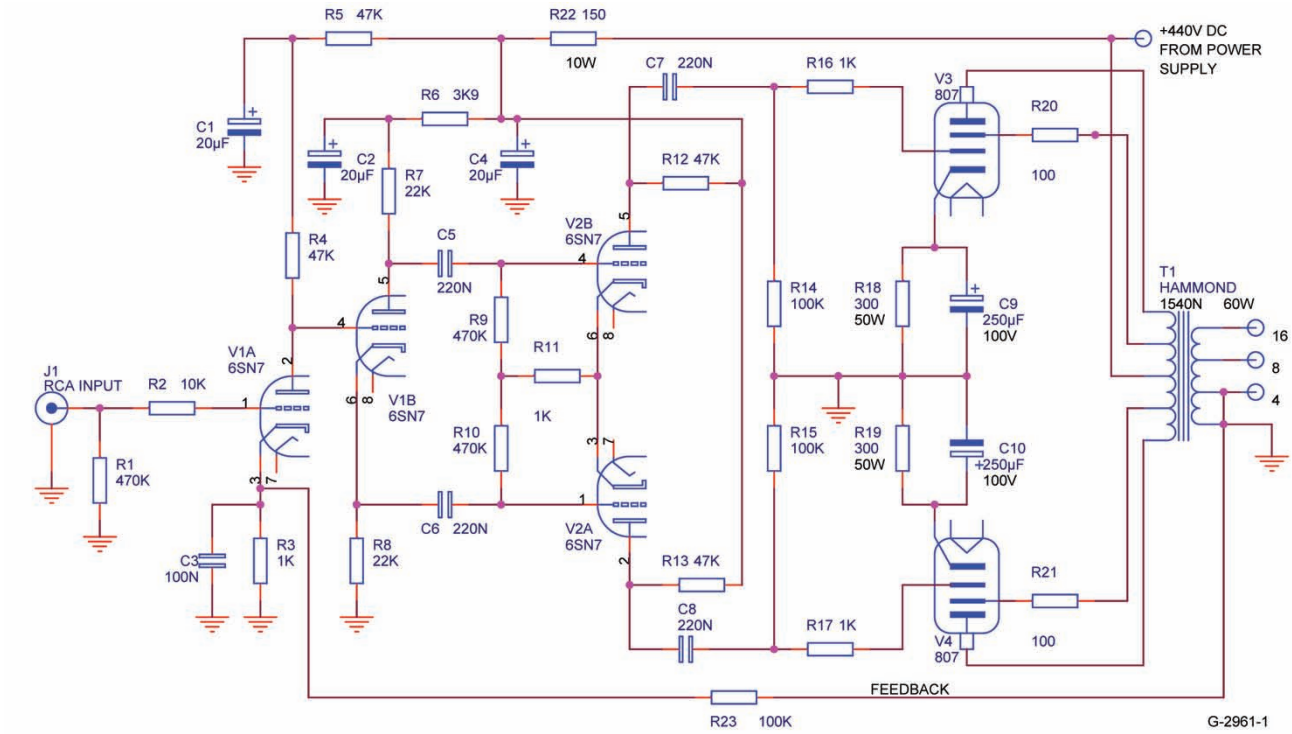


FIGURE 1: Power amp circuit (one channel).

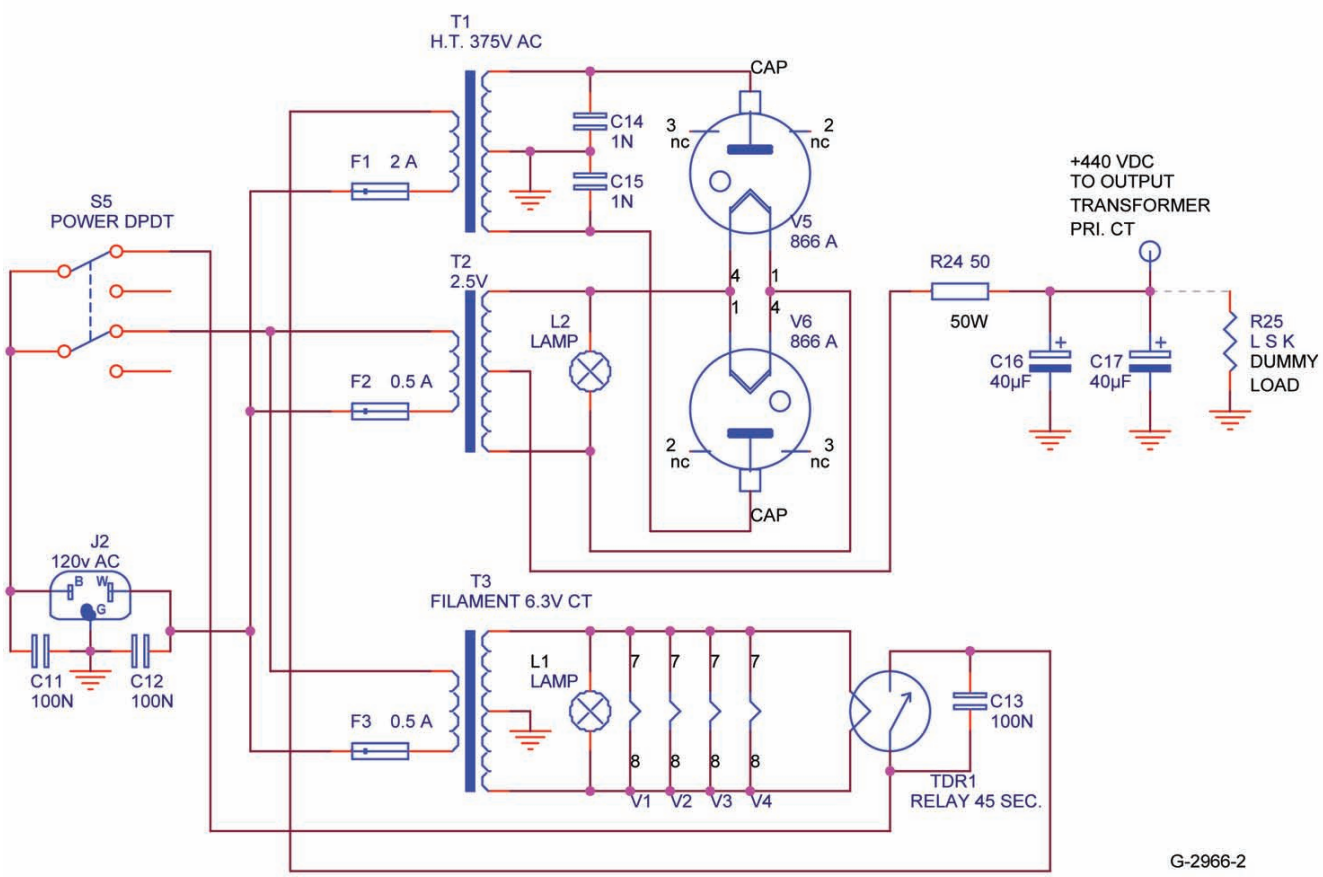


FIGURE 2: Power supply circuit (one channel).

The Unnecessarily Complex 300B Amplifier

Rube Goldberg meets the directly heated triode.

By Pete Millett

PHOTO 1: The amp.

I have a confession to make.

Even though I've been experimenting with and designing with tubes for many years, I've never designed or built a single-ended directly heated triode amplifier. Until now, that is.

Single-ended DHT amps have always seemed too fashionable. I've always been more interested in doing something out of the ordinary. The high price of triodes such as the 300B and the fact that I hadn't heard many of these amps that really sounded great to me were also major contributors to my prejudice against these amps.

But all along I had this nagging question in my mind: was I missing something? After a fortuitous trade netted me a pair of new 300B tubes, I decided that I really needed to give it a go.

DESIGN GOALS

If I were to design and build a single-ended 300B amplifier, I decided that I should incorporate as many of the current design details for this amp that I could. After all, how better to find out whether any of them were worthwhile?

To this end, I deemed the amp needed to use interstage transformer coupling, fixed bias, DC and/or CCS filament supplies, and mercury vapor rectification with a choke-input filter. Some of these design details have clear technical advantages associated with them—the advantages of others were not so clear to me. In any case, even if I wasn't sure what the amp would wind up sounding like, I *was* certain that it was going to *look cool*!

Being the engineer that I am, of course, this also meant that I needed to include circuitry to delay application of B+ until the mercury vapor tubes had sufficient time to heat. And fixed bias meant I needed to design a very clean bias supply, and provide protection so that if the bias supply were to fail, the output stage would be protected from excessive plate current.

The result (**Photo 1**) is an amplifier that I call the "unnecessarily complex" 300B amplifier, because you really don't need all of this complexity to build a good single-ended 300B amp. You could build the same amplifier circuit with a more conventional power supply with a lot fewer components. But, this is a hobby, not a business, and who says we need to be practical?

THE AMPLIFIER

The amplifier circuitry itself—that is, the part that the audio signal passes through—is really not complex at all. The schematic is shown in **Fig. 1**. The 300B output tube is coupled to the speaker via a Tango XE-20S output transformer.

The 300B's grid is driven through a 1:1 interstage transformer (I used a Tango NC-14) by a two-stage amplifier using two sections of a 5687 or 7044 dual triode, DC-coupled to each other. I found both the 7044 and 5687 worked equally well, as should the 7119/E182CC tube. Power to the first stage needs to be decoupled with an RC filter; without it, I found that there is a considerable gain peak at low frequencies. Note that a large (and relatively high voltage) cathode bypass capacitor is required on the second stage to provide enough gain, and to keep the Zout low enough to drive the interstage transformer, especially at high frequencies.

The negative grid bias voltage for the 300B is delivered through the interstage transformer. The bias voltage is adjustable via a pot, and is bypassed to ground with a 3.3µF polypropylene film capacitor.

My initial design used a single-stage driver, using a triode-connected EL802 tube, a very high-Gm pentode designed for video amplifier service. This design was simpler and actually performed slightly better than the two-stage design

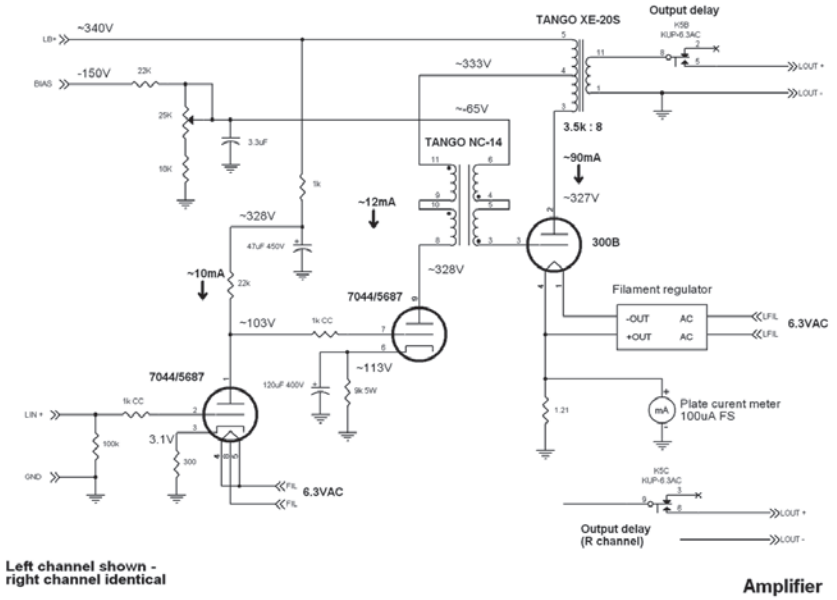


FIGURE 1: Amplifier schematic.

shown here. Unfortunately, it suffered from slightly lower than ideal gain, and high input (Miller) capacitance that made it problematic to use with a passive attenuator directly from a CD player. If you're interested in this design (and don't mind using a preamp), you can find the schematic on my website www.pmillett.com.

I chose to operate the 300B tube (I used the TJ meshplate 300B, shown close-up in **Photo 2**) at a plate voltage of about 325V and a current of around 90mA, partly based on what I could conveniently generate using available power transformers. This is just under 30W of plate dissipation, well under the 40W maximum that most 300B tubes carry.

Plate current (actually cathode current) is metered to facilitate bias adjustment. I used some lovely little surplus meters (no, I'm not revealing my source!) that were graduated from 0 to 100. With a 1.2Ω resistor in parallel with the meter, this equated to a 0-100mA scale, exactly what I needed.

The only thing a little "different" in the amplifier circuit is that there is feedback taken from the ultralinear tap of the output transformer, applied to the primary of the interstage transformer. I did this because I found the amount of distortion in the amplifier without it was a little too high for my taste.

The 300B filament is supplied with regulated DC provided by a circuit I designed and built on a small PCB. The schematic of the filament supply (**Fig. 2**) consists of a bridge rectifier (using Schottky rectifiers) with a large capacitive filter, followed by an LM1085 (or LT1085) 3A linear voltage regulator IC. The output voltage is adjusted to 5V (measured at the tube socket under load). A common-mode choke such as those used in EMI filters is used at the output.

The PCB layout is shown in **Fig. 3**. You can download more details on the filament supply, including the PCB layout, from my website.

I designed the form factor of the filament supply so that I could also substitute a constant-current filament supply sold by TentLabs (www.tentlabs.com). If you don't want to build your own, I definitely recommend the TentLabs supplies.

Another little complexity that you'll see on the amplifier schematic is that there is a relay in series with the speaker outputs. I found that if you delay application of B+ (which you must do with mercury vapor rectifiers), you get quite a "thump" in the speaker as the B+ rises. So, this relay mutes the speakers for a few seconds after the application of B+. More on this later.

POWER SUPPLY

In the power supply design (**Fig. 4**), the complexity is increased just a little. Be-

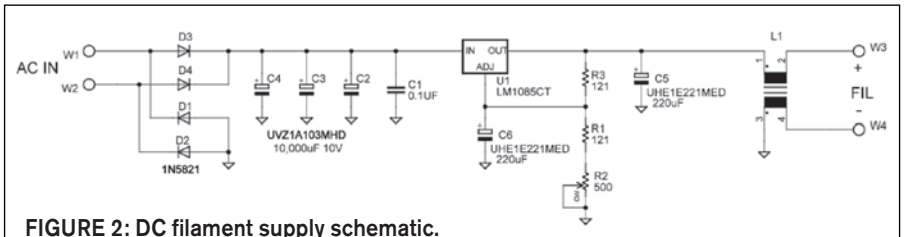


FIGURE 2: DC filament supply schematic.

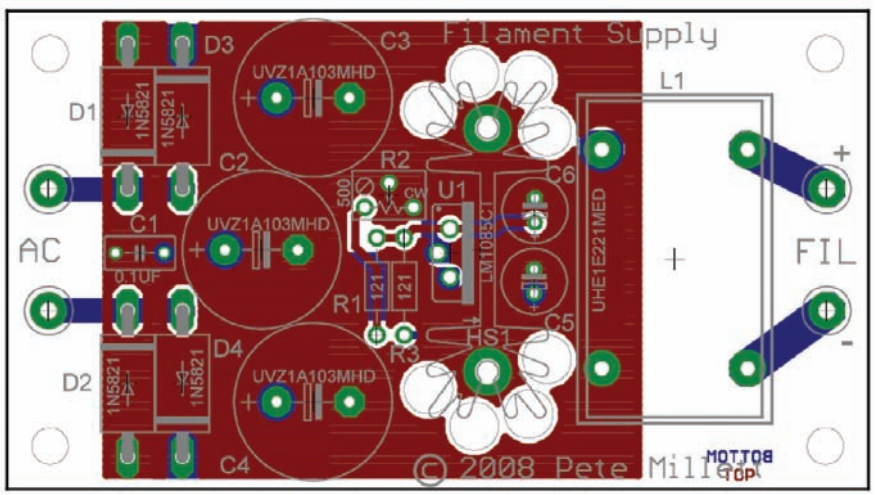


FIGURE 3: Filament supply PCB.

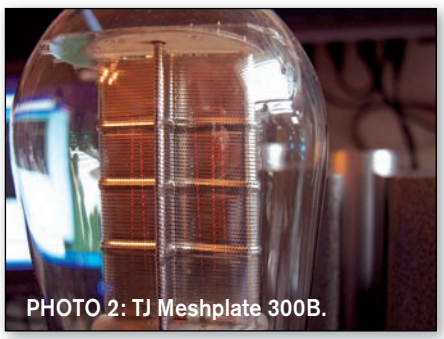


PHOTO 2: TJ Meshplate 300B.

cause I wanted to use fixed bias in this design, I needed a negative bias supply. A sane person would just use a half-wave solid-state rectifier, but where's the fun in that?

I used an inexpensive small plate/filament transformer (from Allied Electronics) and a tube rectifier using a 5AZ4 tube (which is essentially a loctal 5Y3). Filtering is via an RCLC filter, which delivers about -314V. Watch the capacitor polarity here—the output is a negative voltage! This raw supply is regulated by a shunt regulation circuit that I'll visit in a moment.

For the main B+ supply, I used 866AX mercury-vapor rectifiers. These are lovely, tall, slender tubes, which are interchangeable with "normal" 866 rectifier tubes and many other mercury vapor and gas rectifier tubes. They certainly add some visual appeal with their bright blue glow (**Photo 3**).

These tubes require a 2.5V, 5A filament supply for each tube. I happened to have a pair of 2.5V 5A filament transformers sitting around, so rather than buying a new 10A transformer, I used the two smaller units I had. This actually worked out well as far as the chassis layout is concerned—the transformers were small enough that I mounted them on standoffs directly under the 866AX tube sockets, which allowed for very short and neat wiring.

Here's a tip if you plan on using mercury vapor rectifiers: Because they can have condensed mercury inside, MV tubes can only be operated in an upright position. If you want to be able to run your amp on its side, to access the guts to make measurements or debug things, get a pair of 3B28 gas rectifiers. You can use these in place of the MV tube, and they have very similar characteristics and allow you to operate the amp in any position.

To supply the main B+ and all the amplifier filaments, I used a big multi-winding transformer from James, the JS9612. This transformer was designed for a conventional stereo 300B amp, so it has several filament windings in addition to the high voltage winding. Because the 866AXs have their own dedicated filament transformers, I used the 5V rectifier winding for the bias supply rectifier. Three other independent filament windings are used for the driver tubes, and the two 300B filament supplies.

Because I wanted the bias supply to be up and running and the 300B filaments to be preheated (along with the 866AXs) before the application of B+, I used a relay to switch the high voltage secondary going to the 866AX plates. This requires a relay that can withstand a fairly high voltage. I used a three-pole relay with a 6V AC coil that carried a contact rating of 600V AC, which is more than adequate for the job, and provides an extra pole that is useful for the time delay circuitry you'll see shortly.

Though the 866AX tubes can withstand relatively high peak currents, they are normally used with choke-input filters. Because choke-input filters are also in fashion, that's the route I took. I used a small film capacitor (0.1µF) to ground directly on the output of the rectifier, but its function is more as a filter for HF noise than ripple.

Unfortunately, it's pretty hard to find chokes that work well as an input choke; most chokes available these days are not designed to have a large AC voltage applied across them. Normal filter (also called "smoothing") chokes tend to become hot and make loud buzzing noises when used as input chokes.

The only off-the-shelf chokes I've found in the US that are up to the challenge are actually sold as plate chokes for parallel-feed amps. If you think about it, it makes sense—a plate choke is also subject to large AC voltages, so must be designed a little differently than a standard filter choke. The choke I used is a One-Electron PRC-1, with its two windings connected in parallel to give 5H at 200mA DC. Following the choke is a 40µF polypropylene-in-oil motor run capacitor, followed by separate LC filter sections for each channel, using conventional 10H chokes and 70µF PP-in-oil capacitors.

One other odd bit you see in the power supply is the arrangement of series/anti-parallel diodes that connect chassis ground to signal ground. This is a ground-loop-breaker circuit. The 5A-rated diodes will conduct if there is ever a potential difference of more than about 1.2V between the chassis ground (which is also the safety ground connected to the ground of the AC mains plug) and the amplifier's power and signal ground. This breaks the path of ground loop currents caused by small differences in ground voltages between different components. In the event of a potentially hazardous condition such as an insulation breakdown

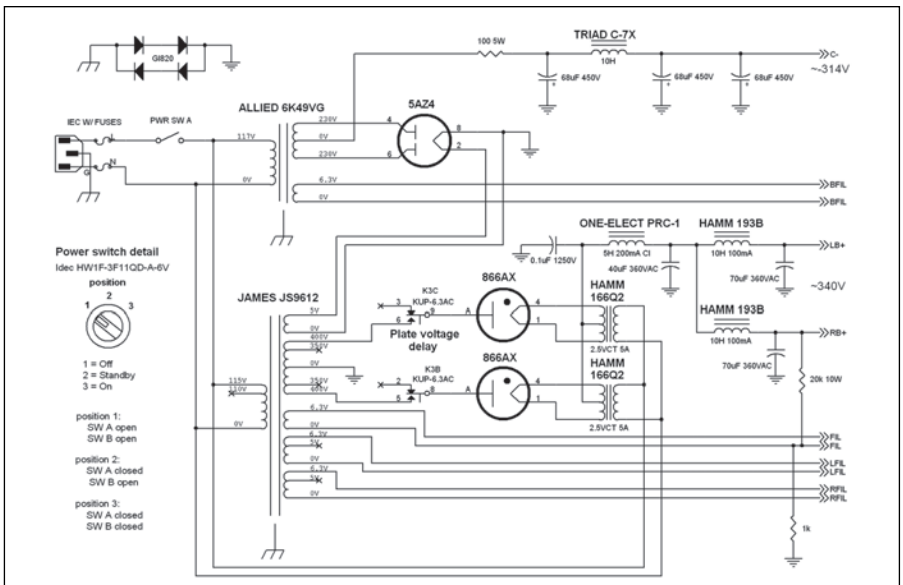


FIGURE 4: Power supply schematic.

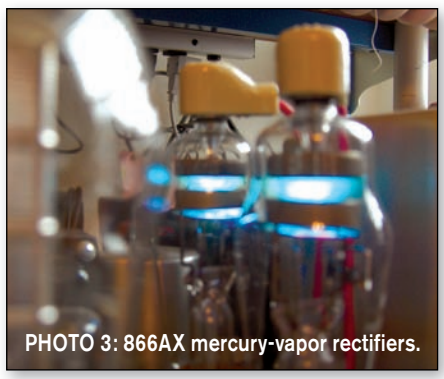


PHOTO 3: 866AX mercury-vapor rectifiers.

in the power supply, current will flow through the diodes to the safety ground, preventing the chassis from becoming energized.

THE BIAS REGULATOR CIRCUIT

The upper portion of **Fig. 5** shows the schematic of the bias voltage regulator circuitry. A 0D3/VR150 tube is used as a shunt regulator. Instead of a series dropping resistor to limit current through the VR tube, it's fed from a constant-current source (CCS) using a 7AD7 tube, which is a high-Gm loctal pentode. It's connected in pentode mode (note the capacitor from screen grid to cathode) and is configured to source about 25mA.

My reasoning for using a CCS here instead of a resistor was twofold: 1. The CCS provides noise rejection, acting like a big inductor. 2. I didn't want to dissipate any more power inside the chassis than I had to. Using a tube CCS put the heat on top of the chassis where it belongs.

Between the plate of the 7AD7 and the cathode of the VR tube (remember, I'm talking about a *negative* voltage here), you will see the coil of a relay, K1. This relay is an old-fashioned 2500Ω plate circuit relay. (I was surprised to find that these relays are still manufactured!) This relay, which activates at 10mA coil current, is used as a bias interlock; unless this relay is activated (which implies that there is current flowing through the VR tube, so the bias voltage is in regulation), the B+ relay cannot be energized. Similarly, if the bias supply fails, K1 drops out, immediately removing B+.

Because the 7AD7 cathode is sitting near -300V, its heater needs to be supplied from a separate transformer winding so it can be biased to avoid breaking down the heater-cathode insulation. The bias transformer I used has a 6.3V winding that I used for this. One side of the heater is connected to the 7AD7 cathode.

The performance of the bias circuit is exceptional. I measured only tens of microvolts of AC noise (unweighted) at the 300B grid (with the driver disconnected). That's at the threshold of my measurement equipment.

CONTROL CIRCUIT

The lower part of the schematic in **Fig. 5** shows the control circuitry to implement the delays and interlocks I mentioned. Here you find the truly unnecessary complexity that would make Rube Goldberg proud!

I wanted to do something more elegant than to use a couple of toggle switches to control power. I ran across a very nice three-position lighted switch, intended for industrial control use. Then I designed a control circuit around it.

The power switch is a two-pole switch. One pole is used to switch the AC line power; in the fully counterclockwise "off" position, this contact is open, and in the

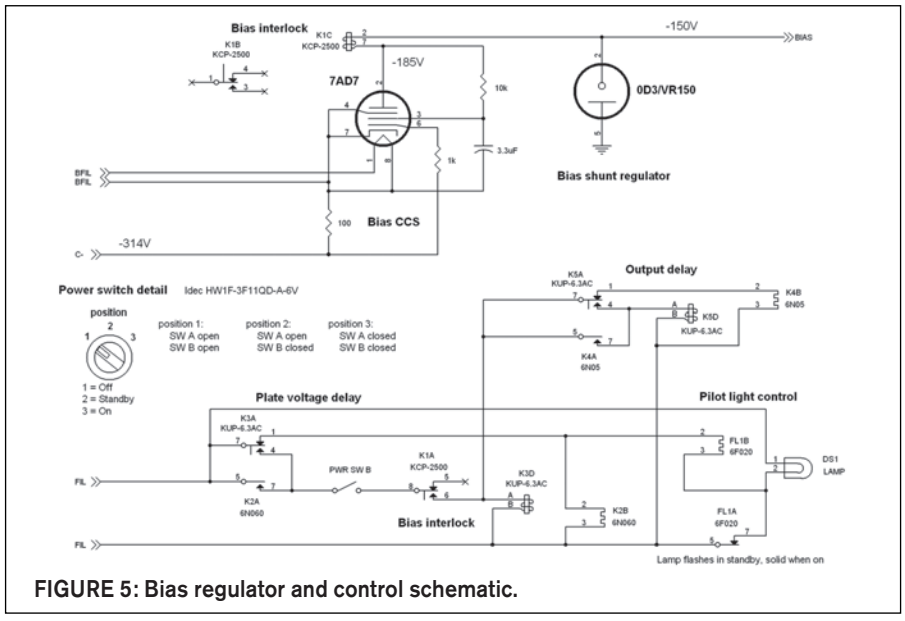


FIGURE 5: Bias regulator and control schematic.

other two positions, it's closed, applying AC power to the transformer primaries. The second pole is closed only in the fully clockwise "on" position, and it's used to enable the B+.

The application of high voltage to the 866AX rectifiers needs to be delayed until sufficient time has passed to properly warm up the rectifiers, which requires at least 30 seconds, preferably 60 seconds. I wanted this to be an "automatic" feature, so you could turn the power switch all the way on without having to wait for the warmup time to pass and enable the B+ manually. I also wanted to make sure that B+ could not be activated unless the bias supply was operating, to prevent damage to the output stage.

I was tempted to implement the delays in modern fashion with a microcontroller, but I've done that before—so I went with an old-school no-silicon approach using thermal time delay relays. Following is an explanation of how this relay logic works:

When AC power is applied (by turning the power switch to the middle "standby" or fully clockwise "on" positions), the 6.3V filament voltage is applied to the pilot light (part of the switch assembly) through a flasher relay (FL1). The flasher relay is basically a thermal delay relay with a normally closed contact—it heats up and opens the switch, which turns off the pilot light and allows the relay to cool, until the contact closes and the cycle repeats. The result is that the pilot light flashes in standby mode, or when waiting for the rectifier tubes to heat. This is no doubt the most unnecessary complexity in the design!

At the same time, power is applied to the coil of a 60-second thermal time delay relay (K2), which is used to delay the application of high voltage until the 866AXs have had 60 seconds to heat. Once the 60 seconds are up, the contacts of K2 close.

At this point, if either the power switch is in the "standby" position or the bias relay is not closed, nothing more happens. The amp happily sits there with the pilot light flashing and the tubes lit.

If the power switch is in the "on" position and the bias interlock relay is pulled in (which it should be unless the bias supply has failed), then the coil of K3 is energized. K3 is a three-pole relay—two poles are used to apply the high voltage to the 866AX plates, and the third is used to de-energize the coil of thermal relay K2, as well as to latch K3 in the on state. This is done so that the thermal delay relay has a chance to cool, preventing it from turning on quickly if power is cycled off then back on.

K3 stays in the energized state until either the power switch is turned to "off" (removing all AC power) or to "standby," or until the bias interlock relay opens (which it should not do unless there is a failure in the bias supply).

When K3 is energized and B+ is applied, power is removed from the flasher coil, which is wired such that the pilot light is lit solid in the "on" state through the normally closed contact. At the same time, a second delay and latching relay circuit consisting of delay relay K4 and three-pole relay K5 is activated, which keeps the speaker outputs muted for five seconds after the application of B+. After the delay, K5 latches on, enabling the output and removing power from the coil of K4.

CONSTRUCTION DETAILS

I built the amplifier chassis from flat 3mm thick sheets of aluminum (for the top and bottom panels), and solid-surface countertop material (DuPont Corian™). I had the aluminum fabricated for me by Front Panel Express (www.frontpanelexpress.com), which adds some cost to the project, but they can certainly do a much better job of it than I ever could—they do beautiful work.

I fabricated the body of the chassis myself using a scrap of ½" thick Corian I found on eBay. The countertop material cuts and drills pretty easily, more or less like a moderately hard wood with a very fine grain. I glued it together using a "slightly retarded cyanoacrylate adhesive" thanks to the advice of my friend Dr. Polymer—that's slow-cure superglue to you and me—and polished it to a nice finish. I used threaded inserts (knurled brass) that press into drilled holes in the Corian to provide screw threads to attach the chassis top and bottom plates, as well as a few of the components inside.

I used single-point terminals and the tube sockets to support the components inside. The tube sockets I used are the "wavy washer" type that mounts in a round hole without screws. Ground connections are to a bus-bar system that is also supported by single-point terminals.

Photo 4 shows a top view of the chassis. The input choke is in the back corner of the chassis behind the main power trans-

PHOTO 4: Top view of chassis.

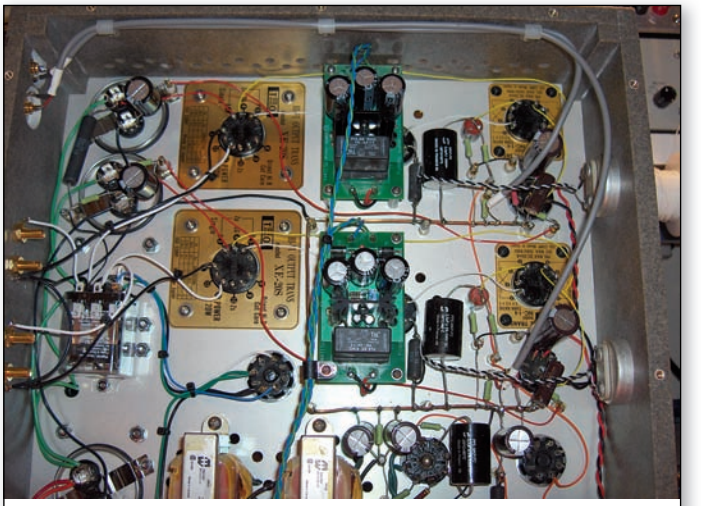


PHOTO 5: Inside wiring (left).

former, located as far away from the interstage transformers as possible to avoid any noise coupling. The B+ filter caps and chokes are lined up along the rear of the chassis. The bias and control circuitry is located in the front right corner, and the amplifier circuitry itself is located at the front left corner.

Figure 6 shows a mechanical drawing of the chassis and the overall dimensions of the amp. You can download details of the mechanical layout, including CAD mechanical drawings and Front Panel Express files, from my website.

In **Photo 5**, a view of the left-hand side of the underside of the chassis, you can see the input wiring, routed via shielded cable from the RCA input jacks mounted on back of the chassis to the input stages at the front. Also visible are the filament power supply circuit PCBs, which are mounted on standoffs directly underneath the 300B sockets. You can see the muting relay at the left side near the speaker binding posts.

Photo 6 shows the other side of the chassis. At the bottom of the photo is a small terminal board from Keystone Electronics that I used to consolidate the AC line wiring between the power switch and the multiple transformer primaries. The two transformers in the center are the rectifier filament transformers, mounted on standoffs directly under the rectifier sockets. The filter choke for the bias supply is located to their right, mounted in a similar fashion, and you can see the power switch in the lower right corner. The HV relay is at the lower left, near the AC inlet—the thick red wires you see are silicone-insulated test lead wire, which passes through the chassis to the top caps on the plates of the 866AX tubes.

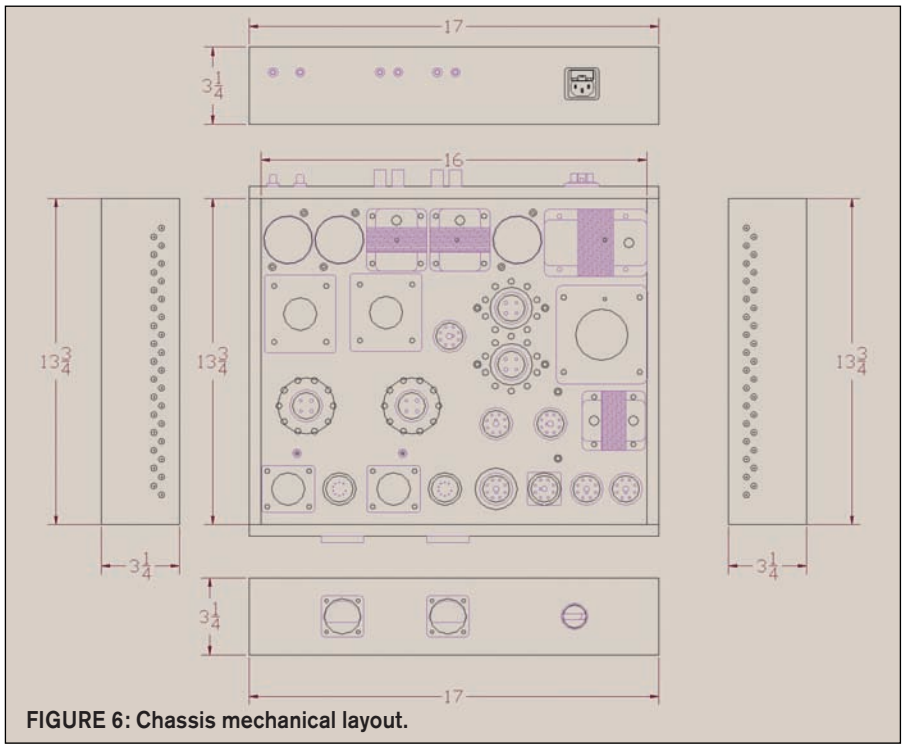


FIGURE 6: Chassis mechanical layout.

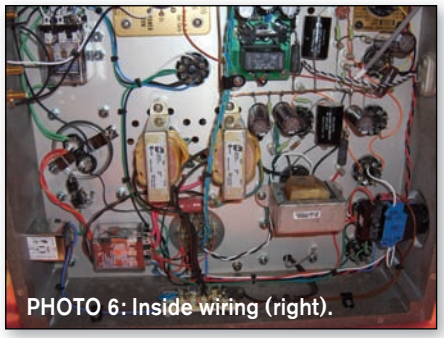


PHOTO 6: Inside wiring (right).

MEASUREMENTS

Figure 7 shows the frequency response of the finished amplifier at 1W out. Due (I assume) to phase shifts through the transformers at low frequency and the negative feedback path from the OPT to the interstage transformer, there is a small rise (<1dB) in response between 10Hz and about 30Hz. The amplitude of this peak is affected by the size of the cathode by-

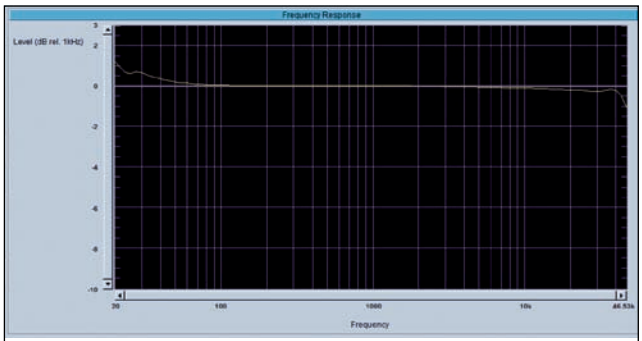


FIGURE 7: Frequency response.

pass capacitor on the first stage, increasing if a smaller capacitance is used.

HF response extends beyond 40kHz, where resonances and phase shifts produce some ups and downs in the frequency response, which eventually drops off at over 80kHz. This ragged ultrasonic frequency response is commonplace among single-ended transformer coupled amps and is a subject of some controversy. This response also manifests itself as ringing if a square-wave input signal with very high slew rate is applied, as seen in the oscilloscope capture shown in **Fig. 8**.

Every attempt I made to tame this high-frequency nastiness resulted in an increase in HF distortion or an unacceptable drop in HF response, so I left it alone. As long as the input signal doesn't contain much energy at these frequencies—which normal, well-behaved audio sources should not—I don't see any reason why this response should cause any problems.

Figure 9 shows a plot of THD + noise versus frequency at 1W output into 8Ω. As you normally see in this type of amp, THD rises at both low frequency (due to transformer nonlinearities) and at high frequencies.

Figure 10 shows an FFT spectrogram of the amplifier output at 1kHz and 1W. As expected of a single-ended amp, the distortion spectrum is dominated by the second and fourth harmonic. The third harmonic is over 90dB below the fundamental.

Finally, **Fig. 11** shows a plot of THD + noise versus output power. Due to the well-filtered power supplies and DC filament operation, this amplifier is very quiet; not much noise appears at the low end of the plot. THD increases as expected with increasing power. The onset of clipping occurs at approximately 7W, and the THD rises rapidly past 5% above about 8.5W. Note that this is with the output transformer connected with a 3.5kΩ load impedance on the 300B.

LISTENING IMPRESSIONS

What does it sound like? Well, it sounds like a single-ended 300B amplifier!

Connected to my Jordan JX-92S speakers, I found the sound to be somewhat muddy, like many other SET 300B amps I've heard in the past. I think this is partly due to the Jordans' inefficiency—I think driving them with an amp that puts out 7 or 8W is asking too much. The Jordans also don't respond well to a high source impedance, which probably contributed some to the lackluster performance as well.

I had much better results listening with my higher-efficiency Fostex-based speakers. Initially, I found the sound of this combo a little too wild—somewhat underdamped bass and a little rough on the high end. I re-wired the output transformers for a 5k plate load instead of 3.5k, which results in a little less power, but lower distortion and lower output impedance. This eliminated the problems, so I listened extensively to this combination.

As you might expect, some types of music sounded better on this system than others. Vocals, both female and male, were very special. There is a sense of realism that I don't get with many of my other amplifiers. The soundstage was also extended, and I noticed small details

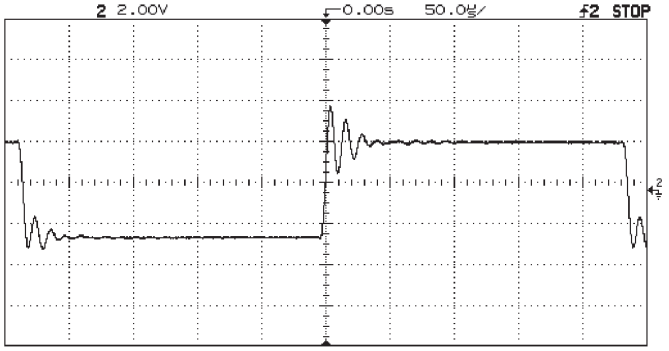


FIGURE 8: Square-wave ringing.

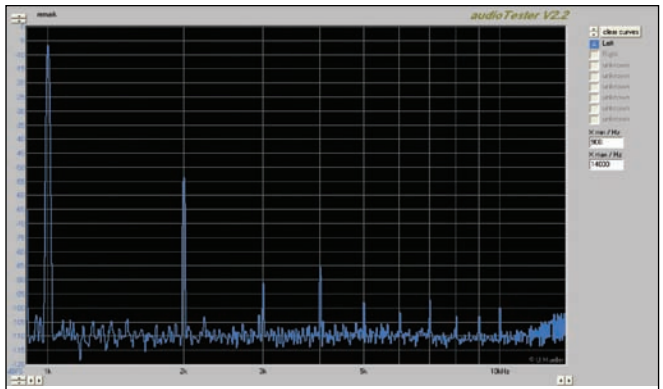


FIGURE 10: FFT spectrogram of amp output at 1kHz and 1W.

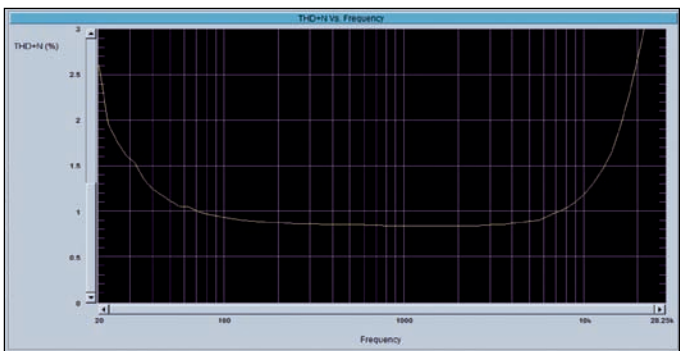


FIGURE 9: Distortion vs. frequency.

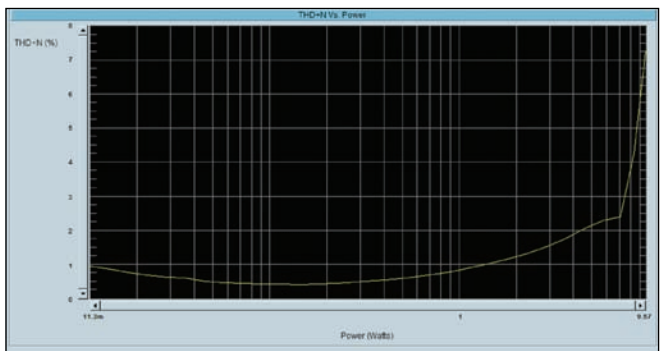


FIGURE 11: THD + noise vs. output power.

that I don't always catch with other systems. Simple arrangements of piano and acoustic guitar attained a high level of realism—no small feat, especially when it comes to piano.

On the down side, things became a little muddled when listening to complex, dynamic music. I've found this to be true in the past when listening to amplifiers with large amounts of even-order distortion.

CONCLUSION

For me, this amp did not produce any revelations—no "sonic nirvana." Don't get me wrong—this amplifier sounds very, very good. But for me, it does not sound significantly better than other amplifiers I've built—just different.

The overall sound of an audio system has much to do with matching the pieces together to get the sound you like with the music you listen to. Arguably, my system—and my speakers, in particular—may not be the most synergistic with a single-ended 300B amp.

As for the audiophile fashion trends I included in this amp design, including interstage transformer coupling and mercury-vapor rectification, I cannot confirm or deny for certain that they had *any* impact—positive or negative. They certainly performed well, but I remain unconvinced that they're necessarily superior to other implementations.

But I haven't given up. I'm now shopping for some new high-efficiency speakers.

Some Notes on Components

Whenever I publish a design, I receive many questions about component selection and sourcing. Unfortunately, with a design like this, there are often parts that I used that may be difficult or even impossible for potential builders to source. Following are a few suggestions:

Output and interstage transformers

Tango transformers, such as the XE-20S and NC-14 I used, are expensive and difficult to obtain in the US (actually, in most of the world outside Japan). You can order them directly from Japan from EIFL export (www.eifl.co.jp), but shipping is expensive. So unless you are willing to fly to Japan to buy them in person (which I heartily recommend, by the way), you may want to substitute something else.

The output transformer can be replaced by pretty much any single-ended OPT with an ultralinear tap, using an impedance ratio of 3.5k:8 or 5k:8, whichever you prefer. The interstage transformer is a little more problematic, because not many vendors manufacture good interstage transformers. About the only vendors I can recommend (other than Tango and Tamura of Japan) are Lundahl from Sweden, which is well distributed throughout the world, and Electra-Print in the US. Electra-Print (www.electra-print.com) builds custom transformers to order, and can, no doubt, build appropriate transformers for this design at a reasonable cost.

Note that the performance of the amplifier can be affected by the choice of transformers. Especially when there is feedback involved, as there is in this design, some experimentation may be needed if different transformers are used.

Power transformers

Unfortunately, James, which makes the main power transformer that I used, no longer has a US distributor. You can order directly from James in Taiwan—I have had reports that this is not too expensive, and shipping is reasonably quick. If you don't want to go that route, there are several other options. You can use multiple transformers and assemble an equivalent of the James unit using stock transformers from Hammond or other makers (for example, using a plate transformer with 5V and 6.3V filament windings plus two separate 6.3V filament transformers for the 300Bs). You could also have a power transformer custom wound, which can be accomplished by many companies, including Electra-Print.

The bias supply transformer I used came from Allied Electronics (www.alliedelec.com), and there are other models from Hammond or other sources that you could use here as well. The same can be said of the rectifier filament transformers—the ones I used came from Hammond.

Chokes

As mentioned, the main B+ input choke needs to be rated for choke-input service. The One-Electron PRC-1 choke, available from Antique Electronics (www.tubesandmore.com), works well. Another option is to find a military surplus choke, either on eBay or at a surplus dealer

such as Surplus Sales of Nebraska (www.surplussales.com). If you have access to Japanese iron, Tango and Tamura make choke-input chokes as well, though they are expensive.

The second-stage filter chokes are not as problematic—I used some custom-wound chokes I had on hand, which you can replace with chokes from Hammond or other manufacturers. The bias choke is not critical—you can use an inexpensive choke from Triad Magnetics (available through the usual distributors) or from Hammond.

Relays

The relays I used are all still in production, though some of them are best bought from surplus dealers. The delay relays and flashers are made by Amperite, and are available from tube vendors, Antique Electronics, or eBay for less money than from mainline electronics distributors.

The plate circuit relay used for the bias interlock is a Potter & Brumfield (now owned by Tyco Electronics) KCP-11-2500. This is a 2.5kΩ coil relay in an octal-based enclosure. The B+ and speaker muting relays are P&B KUP-series 3PDT relays with a 6V AC coil, part number KUP-14A55-6. The contact voltage ratings of this relay are a little confusing, but the label on the relay reads 3A, 600V AC, which is more than adequate. I found both of these relays in stock at Newark (www.newark.com) and at Master Distributors (www.masterdistributors.com). You can also find equivalent relays at surplus dealers.

Capacitors

The main filter capacitors used are polypropylene-in-oil AC-rated capacitors normally used for motor run applications. These are made by companies such as ASC, GE, or Commonwealth Sprague, and I've found the best deals for these are on eBay. The bias supply filter capacitors, input stage B+ filter caps, and cathode bypass cap on the second driver stage are electrolytic. I like to use low-ESR caps from Nichicon, but you can use whatever cap you like here. You may want to experiment with paralleling a small film capacitor across these—that is one tweak I haven't yet experimented with.

Plate current meters

Analog meters are becoming hard to find, especially small, high-quality units. They are still manufactured, but can be very expensive. Your options include surplus—if you can find what you need—or new production from companies such as Simpson or Tripplett. There are also meters sold specifically for tube amps that are very nice, though I don't know of a US source—for example, Yamamoto in Japan (the seller of uber-expensive Teflon tube sockets) sells them (expensive, of course).

The best bet is a tube amp parts vendor in Belgium called Aquablue (www.diyparadiso.com). These very attractive and reasonably priced meters are small enough that postage is not an issue. Tell Benny I sent you.

The power switch

The power switch I used is made by Idec, part number HW1F-3F11QD-A-6V. This modular switch can be assembled a zillion different ways. I found that Master Distributors seems to stock virtually every combination. Study the datasheets found on the Idec website (www.idec.com). You can even pick your own pilot light and trim ring colors. *aX*

Glass Audio Marketplace

Balanced Cable Hum: Cause and Cure

Here are some solutions to the problem of 60Hz hum induced into balanced-line microphone cables.

I used to think I knew a lot about interface design for audio systems. I've certainly enjoyed reading articles by experts who agree with my beliefs about grounding, shielding, and balanced lines[1]. But recently I've stumbled onto something that throws into doubt everything I've learned about audio interfacing.

HUM SOURCES

In the lifelong quest for high fidelity, we always seek more dynamic range, pushing headroom up and noise floor down. And what do we meet near the noise floor but our old nemesis, 60Hz hum. Our environment is filled with 60Hz signals, both electric and magnetic, unless you happen to live off the grid and can power everything from batteries and solar cells. Most hum and buzz can be addressed with shielding and grounding, but to deal with stray 60Hz magnetic fields from power supplies, fans, and motors, we must use balanced-line cables, right?

Wrong! Try this experiment today while I have your attention. Plug a low-impedance microphone into a preamp using a typical balanced-line mike cable with XLR connectors on both ends. Set gain high enough to hear the noise floor while listening in headphones, and drape the mike cable across a wall outlet adapter, the kind that puts out 9V DC for stomp boxes and accessories. If you aren't used to listening near the noise floor, please note that there is a danger of unexpected inputs causing very loud outputs. You will now be listening to 60Hz hum plus harmonics, coupled from the outlet adapter's transformer into the mike cable. But that's impossible, isn't it?

Now find a microphone with a single-ended cable, having just one center conductor surrounded by a shield. In my studio I happened to have a tie-tac microphone with a single-ended cable to the battery box, and balanced-line from there to the mixer. I was stunned to find that the balanced-line section easily picked up hum from my little power transformer, while the single-ended section suffered no hum at all! What's going on here? You can also try this experiment with a musical instrument having a single-ended output connected to a balanced input through a direct interface (DI) box. You'll find the same thing: the single-ended cable rejects hum, while the balanced-line cable lets hum in.

I don't mean to pick on wall outlet adapters, but you must admit that they are a common source of stray 60Hz magnetic fields in many small studios and listening rooms. Most of us have at least one outlet strip full of them, stuck in a corner or behind an equipment rack, along with our excess lengths of balanced-line cable that connect everything together. This is a recipe for disaster, unless you can figure out what's wrong and how to correct it. And why have I been buying XLR cables all these years if they don't protect me from the sources of hum I'm most likely to encounter? I've thought about this a lot, and have concluded that the environment has changed since the days when XLR cables first became popular.

It seems that balanced-line cabling must have originally been meant to eliminate hum pickup due to differing ground potentials between the various parts of an auditorium, recording studio, or broadcast station[2]. And it must have done this job very well, because no one would suggest building such a facility today using single-ended cabling. But I don't think there were many wall outlet adapters sitting on the floor back then. The first outlet adapter I ever saw was for an AT&T Princess Phone, but now they are everywhere. So why do balanced-line cables allow hum to enter from this new source?

MAGNETIC INDUCTION

Magnetic fields induce a voltage in a circuit that is proportional to the time rate of change of magnetic flux through the surface area outlined by the circuit[3]. This is known as Faraday's Law.

$$E = \oint_C \vec{E} \cdot d\vec{l} = -\frac{d\phi_m}{dt}$$

Notice that DC magnetic fields—from permanent magnets, for instance—have no effect because their rate of change is zero. Magnetic flux through a surface is defined as the integral over the surface of the magnetic field perpendicular to the surface.

$$\phi_m = \int_S \vec{B} \cdot \hat{n}\, dA = \int_S B_n\, dA$$

For the simple case of a uniform field passing through a flat surface, the flux is just the product of field times area, with a cosine factor to account for the field's angle with respect to perpendicular.

$$\phi_m = B\, A \cos\theta$$

You can make a couple of assumptions to simplify this analysis. First, assume that all the induced voltage appears across the preamp input. In other words, the preamp's input impedance is high compared to the microphone source impedance. Then assume that the 60Hz magnetic field is acting in a relatively small region—say, over 6″ of cable or less. This turns out to be important, because it is about the same length scale as the twists in twisted pairs.

TWISTED PAIR CABLE

Now consider the input circuit, including a microphone capsule, the two center conductors in the balanced-line cable, and a preamp input (**Fig. 1**). You can ignore the braided shield, because it is likely made of copper and has little effect on magnetic fields. The surface area outlined by the circuit is the small gap maintained between the two center conductors by the insulation on each of them. The center conductors are twisted together so that a uniform magnetic field over a length of cable will cross the gap in alternating up and down directions.

Induced voltage will then alternate plus and minus, adding up to approximately zero if the field is the same over a length containing many twists. The reason this fails in this case is that the wall outlet adapter's external magnetic field is localized to a small region of the cable, perhaps affecting just one twist or so. You could get lucky and find that the cable lies so that induced voltages cancel, but it is more likely that they will add up to be non-zero.

If this model is correct, then induced hum should depend on rotation of the cable, with one positive maximum and one negative maximum per complete rotation. When I rolled the mike cable between my fingers near the wall outlet adapter, this is exactly what I observed (**Photo 1**). You can get the same effect by sliding the cable length-wise along the outlet adapter. In fact, this is one way to find out how close the twists are in a twisted pair.

I also found that if you unplug the microphone but leave the cable connected to the preamp input, the circuit is broken and the hum goes away. If you short pins 2 and 3 at the microphone end, the circuit is complete again and the hum comes back. None of these observations hold, however, for the single-ended mike cable. Rather, it ignores magnetic flux because it doesn't define a circuit whose enclosed area the flux can penetrate.

FIXING THE PROBLEM

Now that you have a good description of the problem, what can be done to minimize it? Luckily, stray fields from wall outlet adapters are highly localized. If you think of it as a magnetic dipole, the field strength will fall with the third power of distance along the dipole's axis. So the first step is to keep wall outlet adapters far away from mike cables, preferably up off the floor.

Next, you see that for a given field strength, the flux enclosed is proportional to the area between the conductors. You should be able to reduce induced hum

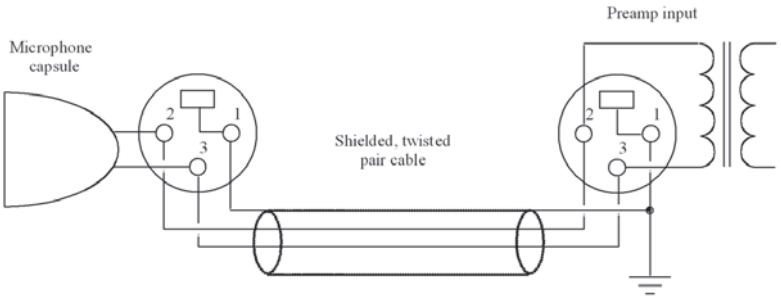


FIGURE 1A: Schematic diagram of a conventional balanced-line mike cable, made with shielded twisted-pair wiring. The mike cable itself and the XLR connectors on both ends are antennas for induced hum.

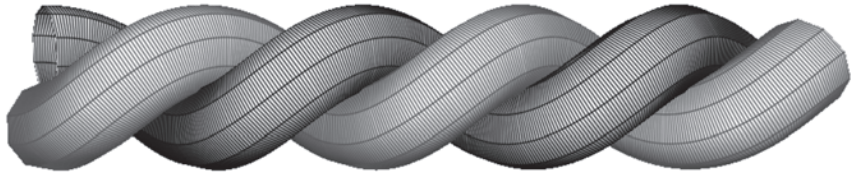


FIGURE 1B: Twisting the inner conductors together limits the gap between them, where magnetic flux can enter, to the combined thickness of their insulation.

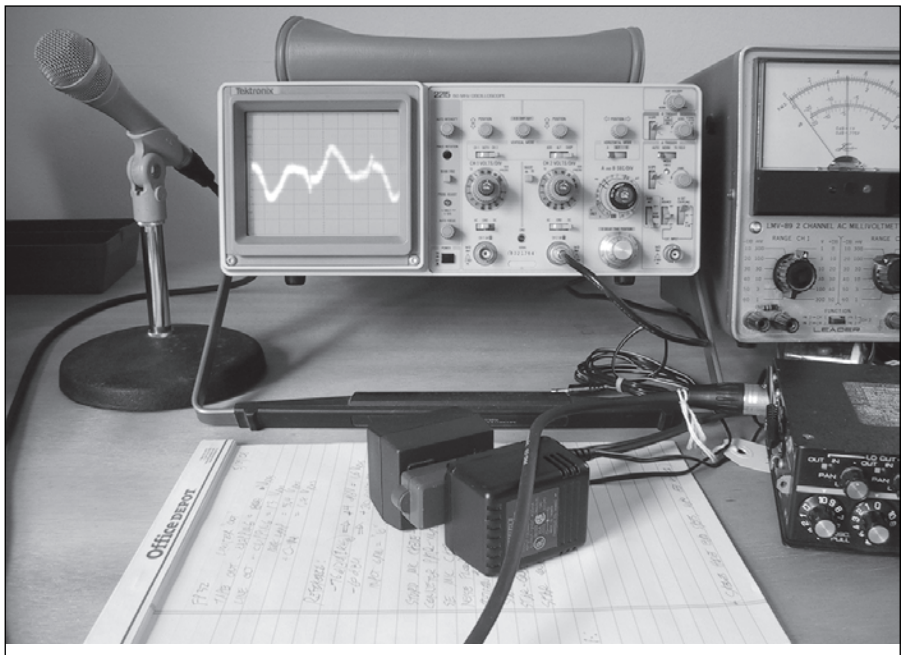


PHOTO 1: Bench setup for measuring induced hum. Note that instruments have been pulled together for the snapshot, but were farther apart for actual measurements. This was necessary because each instrument has its own power transformer and stray magnetic field. Microphone shown was replaced with a shorted plug for actual measurements.

by choosing a cable with smaller conductors having thinner insulation separating them. I happened to have some Belden #8640 instrumentation cable on hand which consists of two strands of 28 gauge magnet wire with enamel insulation, twisted and covered with a foil shield. So I made a mike cable out of it, and measured a 12dB improvement in hum rejection (**Table 1**) compared to a generic balanced mike cable. Unfortunately, this cable is too fragile for field use, but you could certainly choose more conventional cables that have a reduced insulation thickness on the inner conductors.

At this point it occurred to me that there is a very weak link in the chain—the XLR connectors themselves. Inside both plug and socket, the two center conductors are forced to a separation of about ¼″, over a length of almost 1″. This represents a huge area which is essentially an antenna for hum. I confirmed this by connecting two mike cables in series and placing the junction near a wall outlet adapter.

Sure enough, the XLR connectors picked up a huge hum signal, more than 20dB greater than the cable alone. But don't the connector shells help us out here? Unfortunately, they are typically made of non-ferrous metal and have no effect on magnetic field. You can work to eliminate inline junctions by having cables long enough for every application, but that still leaves the connectors in patch bays and stage boxes. Until new connector designs come along, you'll just

TABLE 1 Hum induced in microphone cables and connectors, measured at the input of a Shure FP32 mixer. Two different wall outlet adapters are shown for comparison: #1 from a big-name manufacturer, #2 an unknown import. All values are relative, since the details of the magnetic fields are not known. All measurements are audio bandwidth (20Hz-20kHz), average AC volts, in units of RMS voltage. Microphone end of each cable was terminated in a short circuit.

Induced hum from wall outlet adapters, referenced to input	Wall outlet adapter 1	adapter 2
XLR connector pair	-81dBV (89μV)	-87dBV (45μV)
Standard mike cable	-104dBV (6.3μV)	-114dBV (2.0μV)
Star-quad connector pair	-114dBV (2.0μV)	-122dBV (0.8μV)
Belden #8640 cable	-116dBV (1.6μV)	-126dBV (0.5μV)
single-ended cable	-125dBV (0.6μV)	-127dBV (0.4μV)
Star-quad cable	-126dBV (0.5μV)	-128dBV (0.4μV)
noise floor	-130dBV (0.3μV)	

need to make sure that stray fields don't encroach on your connectors.

MORE SOLUTIONS

You won't believe what I did next. Based on my experiment with the tie-tac microphone, I made a single-ended mike cable, with some single conductor plus shield coax and XLR connectors on both ends (**Fig. 2**). The incredible thing is that it worked! Good clean sound and very little hum from wall outlet adapters.

But I can't recommend this as anything other than a laboratory curiosity. The reason is that you'd be giving up the protection from circulating ground currents that you enjoy with balanced-line cables. Also, phantom power wouldn't work as intended. Avoid single-ended solutions, for now.

Now isn't there some way you can have the best of both worlds? Rejection of stray magnetic fields and protection from circulating ground currents in one cable? While researching this article I learned of a cable type I hadn't heard of before, the "star-quad" configuration. Star-quad is defined by Wikipedia as "a cable with four conductors in a square formation, in which opposite conductors are connected in parallel at both ends to form a single, magnetically-concentric balanced pair, in order to minimize magnetic induction.[4]"

I had to try this, so I cut the ends off a 4-conductor shielded MIDI cable and put XLR connectors on each end, carefully connecting opposing conductors in parallel (**Fig. 3**). For this to work, I believe that all four conductors must be twisted about a common center. Two twisted pairs twisted around each other probably won't help. I must say that star-quad is the answer.

I measured a 20dB reduction in induced hum as compared to a generic balanced mike cable. Yet, pins 2 and 3 are still isolated from ground, so protection against circulating ground current is maintained and phantom power continues to work normally. It's tempting to throw out all my old mike cables and replace them with star-quad units, but what about the connectors? As long as there is an XLR connector on each end, even the best cable can still be a hum antenna.

In high-resolution applications, it may be desirable to use a non-standard connector, one that maintains a magnetically-concentric balanced pair along its length when connected to star-quad cable. But would the performance improvement justify breaking with the long-standing tradition of XLR connectors?

By now you should know what my next experiment was. I cut my home-made star-quad cable in half and inserted a pair of computer connectors, using five adjacent pins to connect the shield and four separate inner conductors (**Photo 2**). I wired the connectors so that opposite conductors would maintain their formation as they entered and exited the connectors. My first attempt yielded better than 30dB improvement in hum rejection, compared to a standard XLR connector pair.

CONCLUSION

It seems clear to me that for high-resolution applications, you should replace conventional two-conductor plus shield microphone cables with four-conductor plus shield cables wired in a star-quad configuration. But it doesn't make sense to do this until you have a connector that maintains the hum rejection

of star-quad cable, while retaining the field-proven robustness and ease of use you've come to expect from XLR connectors. Such a connector would need to be available as plugs and sockets, in both cable ends and chassis mount versions. It should be available with RF suppression built-in, not tacked on as an afterthought. And it should be suitable for designing into microphones, with appropriate strain relief for hand-held applications.

Some of you may be old enough to remember a time before XLR connectors were standard on microphones, when every mike included a cable with stripped and tinned ends waiting for you to attach your connector of choice. Such a time may be approaching again. Meanwhile, you can continue to experiment with new configurations, and report your results in journals such as *audioXpress*.

High-end mike preamps would have one more requirement. It now seems likely that to push hum below the noise floor, point-to-point wiring and circuit board traces inside a microphone preamp would also need to maintain a magnetically-concentric balanced pair configuration, at least until reaching the first gain stage. This will be necessary because—whether your gear is rack-mounted or just sitting on a table—it will be part of a system containing a variety of power transformers, each having its own stray magnetic field.

What about switching power supplies, the kind that are used with laptop computers and that don't have an iron transformer operating at 60Hz? I tried my various test cables draped over an adapter powering the computer I use to type, and heard a raspy buzz at a level similar to iron transformers, with some 60Hz and other components as well. So that's not the answer either.

One author[5] has designed a rack-mounted power supply for replacing "wall warts," which could be a good improvement in some applications. Alert readers will notice that I haven't mentioned triaxial cable, which has one center conductor and two concentric shields, all of which are electrically isolated. I guess I just didn't get around to it. If any of you have tried using triax in audio applications, I'd love to hear about it. aX

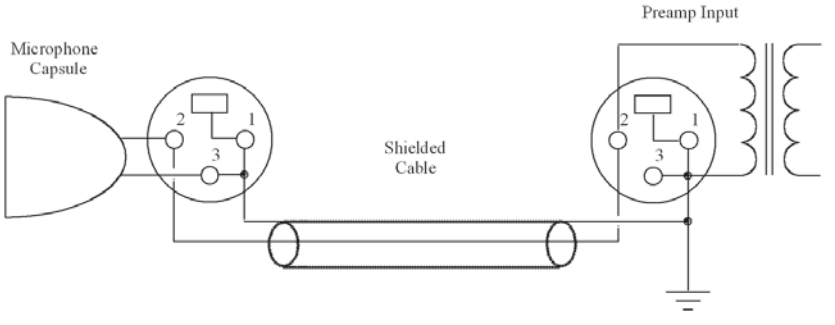


FIGURE 2: Single-ended mike cable, made with single conductor plus shield wiring. This laboratory curiosity has surprisingly low induced hum, but cannot be recommended for real-world applications.

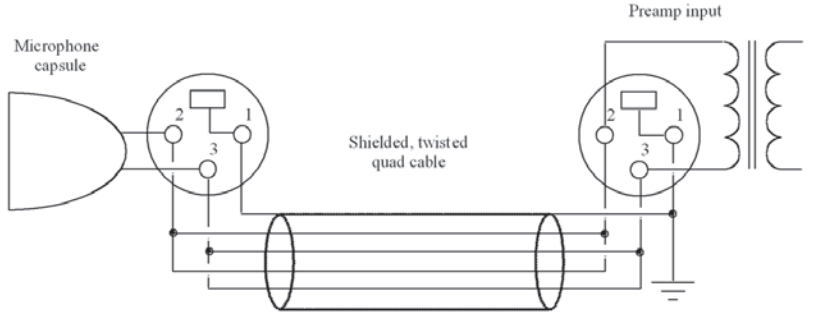


FIGURE 3A: Star-quad mike cable made with four conductors twisted about a common center, then covered with an overall shield. This cable has very low induced hum and is well suited for general applications, but see text for discussion of why you shouldn't use it with XLR connectors.

FIGURE 3B: Four inner conductors are twisted about a common center to provide a magnetically concentric balanced pair. Opposite conductors are connected in parallel at both ends, so red and black are one side of the balanced line, while blue and yellow are the other side.

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1. Gary Galo, "Grounding and System Interfacing," Jan. '07 *audioXpress*, pp. 26-33.

2. Bill Whitlock, *Answers to Common Questions About Audio Transformers,* Jensen Transformers, Inc. Application Note AN-002, 1995, available online at www.jensentransformers.com/apps_wp.html.

3. Paul Tipler, *Physics for Scientists and Engineers, 4th ed.*, W.H. Freeman and Company, New York, 1999.

4. See Wikipedia, XLR connector, http://en.wikipedia.org/wiki/XLR_connector (as of May 30, 2008, 15:00 GMT).

5. Ron Tipton, "A Multiple "Wall-Wart" Power Supply," June '08 *audioXpress*, pp. 20-23.

PHOTO 2: Closeup of computer connectors wired in star-quad configuration. This connector pair is inserted in the middle of a star-quad mike cable, which was made by cutting the ends off a shielded MIDI cable and installing XLR connectors.

VENDORS

Yard Sale

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

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Ad Index

The iPanda is the latest in **Speakal**'s line of audio systems with a built-in iPod docking station. Designed with five speakers which deliver 25W of premium sound, the iPanda can produce 360° sound, reaching the entire room. Features include an onboard hi-fi amplifier, dynamic airflow optimizer, iPod dock with standard auxiliary 3.5mm input jack, as well as a remote control. To learn more, please go to www.speakal.com.

Dayton Audio has expanded its line of HDMI cables. Its new line of flat cables allow cable pairs to separate even further, reducing the effects of inter-pair crosstalk. The cables are ATC certified, guaranteeing v.1.3b compliance, giving color up to 48-bit deep, and faster refresh rates. Authorized retailer in the US is Parts Express (parts-express.com).

New products from **Pass Laboratories, Inc**. include the XP-10 and XP-20 preamplifiers, which employ 64 volume steps and present a superior noise floor and fine detail retrieval and furnish two balanced and three single-ended inputs. The XP-20 includes an external power supply which conveys musical bass with more impact, and a more refined sound stage. Also new from Pass Labs, the SR1 Loudspeaker is a four-way floor standing system, which uses dual enclosures, and was designed to harness the mass controlled piston model. Available as a bi-amplified or quad-amplified system, the SR1's sensitiviy is approximately 90dB for a 2.8V input. To learn more, visit info@passlabs.com.

OTTO Engineering has introduced the Traveler, a noise-canceling headset with a built-in 1GB MP3 player. Featuring padded leather ear cups, the Traveler contains Active Noise Reduction technology which reduces 85% of surrounding noise. The MP3 player can store 250 songs or 1000 minutes of audio books, and it comes with a detachable cable. For more information, please visit www.hearotto.com.

The Stibbert Tube CD Player by **Goldenote**, from **Koetsu USA** employs a newly designed double chassis completely isolated by four conical springs. Also containing Dual Speed technology which reduces voltage and speed and Zero Clock which eliminates jitter and clock errors, the Stibbert features a new analog output stage based on two 6922 tubes. To learn more, go to www.goldenote.it or www.koetsuusa.com.

Harman Kardon, Inc.'s 7.2-channel AVR 7550HD receiver won the 2009 International Consumer Electronics Show (CES) Best of Innovations 2009 Design and Engineering award in the Audio Components category (www.harman.com). The 7550HD provides HDMI capability and can pass video signals to 1080p, and employs Faroudja DCDi Cinema digital video processing, providing enhanced image quality from all video sources. It is the first receiver to utilize **Texas Instruments**' Aureus DA710 audio digital-signal processor in a dual-DSP implementation, and offers a built-in USB port, an Internet radio tuner, and automated room-equalization and system calibration.

New from **Audio Technica**, the BP4025 X/Y Stereo Field Recording Microphone includes large diaphragm capsules and phantom power operation, for broadcast and professional recording. The AT8022 X/Y Stereo Microphone is compact and lightweight, and operates on battery or phantom power for use with pro or consumer equipment. For more, contact audio-technica.com.

The NL-27, from **Scantek**, is a sound level meter that can measure fast or slow maximum, peak sound levels, and has manual storage for 199 measurements. Allowing internal and external calibration, the NL-27 is small (11mm) and achieves automatically timed averages and DC and AC output. For more, go to www.scanteckinc.com.

Designed for portable listening applications, the HFI-15G, from **Ultrasone Inc.**, is an open-back headphone that allows you to experience natural sound sensations without the use of digital processing. Featuring dynamic principle and a 40mm mylar driver, the HFI-15G weighs 85g without a cord, has 32Ω impedance, and a frequency range of 20-20kHz. To learn more, please visit www.ultrasone.com.

Sennheiser introduces the IE6, IE7, and IE8 ear-canal headphones, designed with dynamic transducers that cover the entire frequency range with one system. The IE6 offers accurate and deep bass and has passive attenuation up to 20dB and frequency response of 10-18000Hz. The IE7's frequency response ranges from 10-19000Hz and produces a more balanced, natural sound, while the IE8 contains a low-profile dial on each earphone, enabling fine-tuning of the bass response. For more information, go to www.sennheiserusa.com.

New from **Trends Audio**, the PA-10 Tube Headphone/Preamplifier is only 0.5kg, and contains a pure class-A circuit and warm tube sound, as well as excellent high fidelity and sufficient driving power. The PA-10 comes equipped with a bi-triode 6DJ8/6922 or 12AU7 tube, two sets of switching audio inputs, and a specially designed AC to 24V AC adaptor with extra low ripple and noise. To learn more, visit www.TrendsAudio.com.

The PerfectWave Digital Suite, from **PS Audio**, utilizes the company's exclusive Global Net Service built on a Cloud Computing platform. The PerfectWave Transport is an optical disc transport for CD and DVD that rips all optical data first to a memory, then outputs that data through a built-in digital lens to eliminate jitter and timing artifacts. The PerfectWave DAC can be configured as a high-resolution server with the addition of an internal network bridge; both are included in the Suite. For more information, contact Dave Kakenmaster at dave@psaudio.com or (720) 406-8946.

Prism Sounds' dScope Series III analyzer has been reduced in price by 24% to $9,850. The analyzer platform supports digital audio carrier testing, quasi-anechoic microphone/loudspeaker analysis, and more. The Series III is suited to production line testing applications, and can easily be automated using its built-in VBScript interpreter. To learn more, please go to www.prismsound.com.

Designed by Joe D'Appolito, **Snell Acoustics**' Phantom is a new loudspeaker that has dual 4.5″ machined XL magnesium midrange drivers mounted over and under a low-resonance 1″ **SEAS** XL silk dome tweeter with neo-magnet. This "D'Appolito Array" configuration is focused on making a coordinated wave-front with smooth and consistent axial dispersion. For additional information, go to www.snellacoustics.com.

McIntosh Laboratory has reissued its MC75 monoblock tube amplifier and C22 preamplifier in honor of its 60th anniversary. The company also introduces its MX150 A/V Control Center, featuring five HDMI 1.3 audio/video inputs, engineered to decode **Dolby** Pro Logic II, Dolby Digital EX, DTS ES Matrix 6.1, DTS Neo:6, and more. Containing an RS232 input, the MX150 can control an entire Mcintosh system from a single hand-held remote. Other products from McIntosh include the XR200, a 51″ tall, three-way floor-standing loudspeaker with a 19-element midrange/tweeter array. It contains three custom-made 8″ bass drivers, each with a 12 lb magnet structure. To learn more, please visit www.mcintoshlabs.com.

The Feature loudspeaker, from **MartinLogan**, is internally powered with a 150W switching amp, and includes an RCA line-level input to minimize wire clutter by connecting directly to on-wall televisions without the need for external amplifiers. Also included are standard speaker-level connections, and a Vojtko crossover that seamlessly blends the Feature's electrostatic transducer, neodymium soft-dome tweeter, and more, across the entire frequency range. Go to www.martinlogan.com for more information.

SANYO's Internet Radio provides instant access to numerous free stations and podcasts worldwide, without the need for a computer. It features eight Internet station presets, 90mm diameter stereo speakers, 4W total output (2W RMS), a stereo input jack for MP3/WMA players, an ethernet and stereo headphone jacks, and can access shared music files from computers on the same network. For more, please visit us.sanyo.com.

CONTRIBUTORS

Bill Fitzmaurice ("The XF410," p. 5) has been a professional musician since 1966 and has been constructing instruments, amplifiers, and speakers for just as long. Vice president of DeltaSounds Loudspeakers Inc., Fitzmaurice is the author of *Speaker Builder's Loudspeakers for Musicians* and over 30 magazine articles dealing with speakers and electric instruments, as well as the action-adventure novel *Operation: Sergeant York*. Bill and his wife of over 30 years reside in Laconia, N.H.

Claudio Negro ("ezDAC," p. 12) is a retired commercial airplane pilot, and has been interested in speaker building since the age of 13 when he built a zero offset three-way speaker with a self-made tweeter acoustic lens. He also became interested in electronics, building a mixer based on a Radford preamp and a Marantz MM pre-schematics, with separate PSU enclosure. In 1985 he was classified sixth in an Italian magazine DIY speaker contest, after which he took a long break from DIY audio. Recently, he has studied speaker measurement software, publishing a tutorial on Speaker Workshop. His other interests are listening and playing hard rock music, photography, computers, and online car racing.

John Badalamenti ("Nardeux Stereomatic 602 S," p. 17) is a certified communications engineer and technician. Licensed by the FCC and certified by NARTE (National Association of Radio and Telecommunications Engineers), he is also a licensed FAA airframe and powerplant technician. He graduated from the Academy of Aeronautics in 1984 and received his associate's degree in Airframe and Avionics Technology. He has an FCC general radiotelephone license with ship radar endorsement, and is NARTE Certified Engineer Class III, NARTE Certified Technician Class II. He is currently employed at ASPECT SOFTWARE, "Communications Engineer Field Services." He enjoys setting up and playing with professional high-end stereo equipment, home theater, radio-control model airplanes, high-power model rockets, and O-gauge model trains. He also enjoys swimming and reading audio- and video-related magazines and books. John can be contacted via e-mail at j.badalamenti@att.net.

Pete Millett ("The Unnecessarily Complex 300B Amplifier," p. 22) has worked as an electrical engineer for about 25 years, doing mostly board-level design of computer and consumer equipment. He's been designing and building tube audio projects since the mid-90s. Pete currently lives in Texas, where he defines integrated circuits for consumer electronics for a large semiconductor company. You can see more of Pete's projects, and contact him, via his website www.pmillett.com.

Mark Williamsen ("Balanced Cable Hum: Cause and Cure," p. 34) is a graduate student in Physics at the University of Wisconsin in Milwaukee. He is active in Milwaukee's busy music festival scene, and has recorded several album projects for local musicians. More info is available at his home page, http://my.execpc.com/~williamm/.

Chuck Hansen (Book Review: *The Guitar Player Repair Guide,* by Dan Erlewine, p. 43) is an electrical engineer and holds five patents in his field of engineering. He began building vacuum-tube audio equipment in college. He plays jazz guitar and enjoys modifying guitar amplifiers and effects to reduce noise and distortion, as well as building and restoring audio test equipment. He enjoys sailing and has over 150 magazine articles to his credit.

John Sunier ("Super Fidelity," p. 46) is a CD reviewer for *Australian HiFi and Home Theatre Technology.* His website is www.audaud.com.

BOOK REVIEW

The Guitar Player Repair Guide
By Dan Erlewine

Reviewed by Chuck Hansen

Hal Leonard/Backbeat Books,
321 pages, soft cover
ISBN-13: 978-0-87930-921-3
www.stewmac.com
$30

This book grew out of author Dan Erlewine's popular *Guitar Player* magazine columns. As a jazz guitarist, I bought the original *Guitar Player Repair Guide* back in 1990. I continued to stuff Dan's technical articles and Repairs & Modifications and Mod Squad columns into that book until the covers fairly bulged. I used Dan's chapter on shipping guitars and amps every time I sold a guitar or amplifier on eBay. Every one of them arrived safe and sound and added to my 100% positive feedback score.

The *Guitar Player Repair Guide* is the most widely used basic reference in repair shops, featuring factory setup specifications for Martin, Gibson, and Fender guitars, in-depth fretting instruction, fixing cracks, braces and pegheads, repairing and replacing pickups, transducers, and wiring components, advice from top repairmen, and much more. This 3rd edition includes lots of recent techniques Dan has discovered over the years while repairing every kind of guitar, from the new and inexpensive to the vintage and priceless. Dan shows you how to evaluate, intonate, and string a guitar. You can print out handy assessment forms for repairs if you own a repair business.

Dan is Director of Technical Operations for Stewart-MacDonald's Guitar Shop Supply in Athens, Ohio. He has also produced a number of acclaimed how-to DVDs. He is the go-to guy for high-quality guitar maintenance and repairs.

The contents of the book are arranged in 19 Chapters (vs. 11 Chapters and 207 pages in the original book): Chapter 1—Cleaning and care; Chapter 2—Installing strings; Chapter 3—Neck evaluation and truss rod adjustment; Chapter 4—Setup and action; Chapter 5—Intonation and compensation; Chapter 6—Tuning machines; Chapter 7—Electric guitar bridges: non-tremolo; Chapter 8—Electric guitar bridges: tremolos; Chapter 9—Acoustic guitar bridges; Chapter 10—Acoustic body repairs; Chapter 11—Necks; Chapter 12—Fretwork; Chapter 13—Nut replacement; Chapter 14—Bindings and pickguards; Chapter 15—Guitar electronics; Chapter 16—Finishing and finish repairs; Chapter 17—Shipping a guitar or amp; Chapter 18—Tools; Chapter 19—Resources and schools.

A free "Guitar Tips" DVD is included in the back cover.

CONCLUSION

I found this to be a must-have book for any guitarist, from casual plinker to serious pro. Signed editions of the book are available from the Stewart-MacDonald website for a limited time: www.stewmac.com. *aX*

CORRECTIONS

In *audioXpress* April 2009 pp. 40-42, my letter looks OK in many regards, but does have some errors needing correction.

1. The redundant "Special Note:" disclaimer in bold should have been deleted.

2. No author is mentioned; it should have been signed as this letter.

3. On P41 top, 1st column, the first sentence should read as: "So, if the DC bias of M2 is for example 2V, the resistor ratio shown increases it to 3x this, or 6V." The general expression for multiplied bias voltage in the previous paragraph portion ending on p 40 is OK.

Walt Jung
waltjung@waltjung.org

The George Danavaras article "The Reflection Loudspeaker" (March '09, p. 5) suggests using the Audax TM025F1 tweeter for the project. As much as I enjoy the sound of this tweeter, it is discontinued.

Alain Price
Alain.Price@sunlife.com

TUBE AMP

My question for Karin Preeda ("A Chip-Driven SE Monoblock," Jan. '09, p. 24): is there a circuit design using a vacuum tube driving the SV811-10? Something like a 6SN7, 6DJ8, 6H30, and so on? I would be interested in building this type of amplifier. I am only a hobbyist, not a circuit designer.

Ed Rusnak
roidua@aol.com

Karin Preeda responds:

Yes, the SV811-10 can be driven by tube. However, I recommend that you use medium power tubes such as the 6A3 or 6B4G because of its low Rp and ability to drive. Electra-Print makes a suitable interstage transformer that's capable of driving a power tube such as SV811-10 to class A2 because it can provide power to its grid. The transformer's spec is 3K: 3K 60mA 3.5W.

Figures 1 and **2** show a two-stage driver design and power supply. Your choices of using pre-driver tubes are as follows:

1. 6SN7 - R1 22K 3W, R2 750Ω 1W. So, the tube will be cathode biased at 4.5V, Ip is 6mA, and Vplate is approximately 160V.

2. 6N30 - R1 10K 10W, R2 500Ω 1W. The tube will be cathode biased at 7.5V, Ip is 15mA, and Vplate is approximately 150V.

For the power supply, you need a separate set. You can use the Hammond 270DX for HT voltage and filament power for 6A3/6B4G. You also need another 6.3V AC transformer for 6SN7 or 6N30s heater.

LITTLE SECRETS

I was pleased to see Mr. Godmaire's efforts in obtaining the quiescent conditions of triode-strapped/ultralinear pentodes when only limited information on the valve is at hand ("Vg1 Correlations," Mar. '09, p. 32). Certainly there are still those useful "little secrets" to be found!

I would like to point out, though, that Mr. Godmaire's empirical method is only necessary if the datasheet provides *only one* set of plate characteristic curves, corresponding to only one screen voltage. However, there are very few valves I have come across for which such little information is available. If two sets of plate characteristics can be obtained, for different screen voltages, then the quiescent voltages for the valve under any other set of conditions may be derived more directly. The following method is quite accurate for anode voltages greater than about 100V.

1. Note the difference in screen voltage between the two sets of characteristics in hand, which is Δvg2.
2. Select any suitable anode current and voltage (they must be the same on both graphs) and note the corresponding bias voltage shown on each graph. Then calculate the difference between these bias voltages, which is Δvg1.

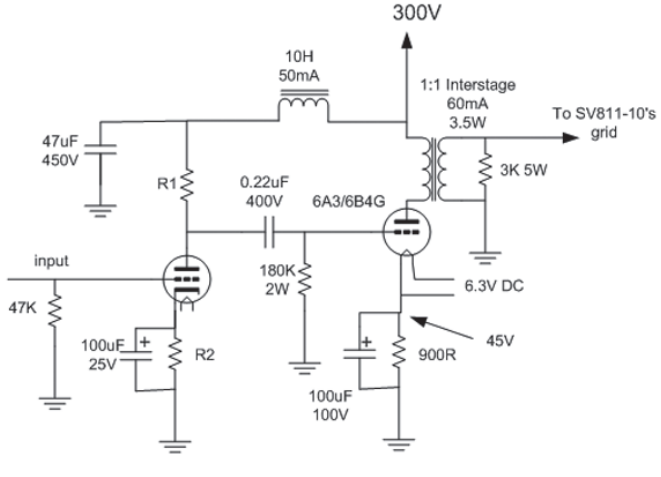

FIGURE 1: Driver circuit.

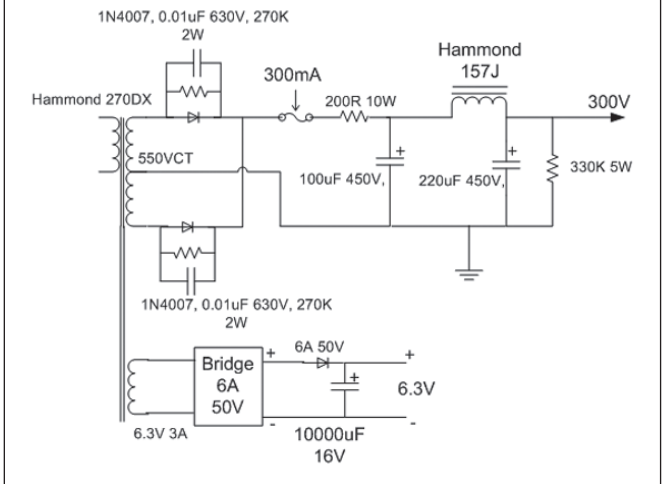

FIGURE 2: Power supply.

3. From these two values, you can find the triode amplification factor, μ_t:

$\mu_t = \Delta vg2/\Delta vg1$

Once you know this, you can readily obtain the quiescent conditions for any screen voltage (and therefore triode/UL mode also):

4. On one plate characteristics graph, note the bias voltage at the desired anode voltage and current that you intend to use in triode/UL mode; this is $Vg1_{(pentode)}$.
5. Take the difference between the screen voltage for this set of plate characteristics, and the desired anode voltage for triode/UL mode. This gives $\Delta vg2$. The required bias voltage is then:

$$Vg1_{(triode/UL)} = Vg1_{(pentode)} + \Delta vg1 = Vg1_{(pentode)} + (\Delta vg2/\mu_t)$$

(Note: Be careful to observe the sign of $\Delta vg2$ in step 5. If the screen voltage for the set of plate characteristics used is *less* than the desired anode voltage in triode/UL mode, then $\Delta vg2$ will be a positive number, and vice versa.)

See also:
Shimmins, A. J. (1950), "The Determination of Quiescent Voltages and Currents in Pentode Amplifiers," *Electronic Engineering*, (September) pp. 386-8.

Merlin Blencowe
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Rudy Godmaire responds:

The letter from Mr. Blencowe exemplifies one of the many merits of a publication such as *audioXpress;* it offers a medium to share knowledge that might be otherwise lost. This is especially important when we think about vacuum tube technology.

I was not aware of the methodology described by Mr. Blencowe and I sincerely thank him for taking the time to share his exposé. I will certainly look forward to trying this very interesting approach and, out of curiosity, compare its results with the "Vg1 Correlations."

I was intrigued by the Rudy Godmaire article "Vg1 Correlations" (March '09, p. 32), because searching correlations between variables of such simple systems which are valves is really an unexpected strategy. In fact, both correlations, established empirically and proposed by Rudy Godmaire in his article, can simply be established analytically, assuming that working points on power valves are always close to their maximum anode power dissipation. This is really an excellent example which demonstrates that, when your knowledge on a system is sufficient, it is always more effective to work on it using this knowledge rather than refusing it and considering the system as a black box.

There is no doubt that both correlations obtained and proposed by Rudy Godmaire will be appreciated by many DIYers. Nevertheless, these DIYers must keep in mind that the generality of such correlations is doubtful and that the bias determination of power valves cannot be reduced to such correlations.

It is, in fact, a difficult problem which takes into account the class of operation of the valve as well as the distortion rate. I thank Rudy Godmaire very much for having proposed his findings to *audioXpress* readers, and I strongly encourage him to continue his personal research.

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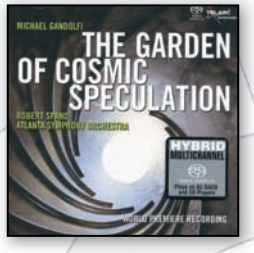
Rudy Godmaire responds:

I was pleased to read the comments of Mr. Touzelet, and I appreciate his observation about the probable pitfall of using such a mathematical model. It is important to clarify that the "Vg1 Correlations" exposed in my article are not meant to substitute for the design of an output stage, which is a completely different matter.

Generally speaking, I find that for a given operating voltage, better performance can be achieved when the bias is set near its maximum value. In this respect, when the plate characteristics for the triode or ultralinear mode are not available, the "Vg1 Correlations" offer an alternate way to facilitate the design of the driver stage before committing to build the amplifier. This is the only purpose of this model, and I hope DIYers will use it this way. *aX*

Super Fidelity By John Sunier

www.audaud.com



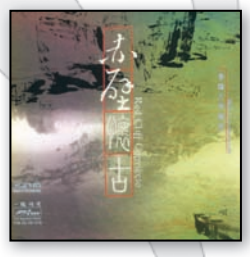
Michael Gandolfi|The Garden of Cosmic Speculation—Orchestral Suites|Telarc MC SACD-60696
This is a world premiere recording and the first recording of a major orchestral work by American composer Michael Gandolfi, who brings some fresh, accessible sounds to symphonic music. The real garden the music references is the creation of US architect Charles Jenicks, who says it's a metaphor for the physics of our universe. One example of a garden feature and the music for it is *The Willowtwist*. It's a shiny metal sculpture something like a Mobius strip. Its music is a twisting, cyclical piece whose surface details are also circular. *The Universe Cascade* is a neo-Baroque suite of dances which quotes Bach. The physical counterpart is a cascade down a stairway, interrupted by landings inscribed with the major events in the history of the universe. The music traces the flow of Western music's history using quick quotations beginning with Gregorian chant and ending with Steve Reich. This one gets my recommendation!

Mahler|Symphony No. 2|Mimi Cortese, sop/Lucretia West, contralto|Orig. Westminster 2-track DVD-R HDDVD137
Mahler's Second Symphony is the one usually chosen to introduce his unique sound world to new listeners. The opening movement introduces a musical struggle between will and death. The slow movement is the simplest and most beautiful Mahler ever wrote, and the short *Urlicht* movement has the contralto singing one of the composer's most inspired songs of faith. The *finale* builds to a gloriously stirring conclusion with the two singers' voices floating over the hushed chorus and ends with a tremendous pealing of bells and sounding of horns. This exciting performance was released on LP and open reel. HDTT processed the sonically superior tape version and transferred it digitally to high-res 96k/24-bit DVD so the final result is about as good as listening to the original tape. Mahler's Second has had good performances by Kaplan, Bernstein, Tilson-Thomas, and Abbado but this powerful Scherchen performance sometimes bests them all in terms of its sheer intensity, and the sonics are superb.

Steve Reich|Music for 18 Musicians|Grand Valley State Uni Ensemble|Innova MC SACD 678 [www.innova.mu]
What an unexpected musical and audiophile treat! The minimalism of Steve Reich has probably been exposed to more listeners via this "trippy" work than any of his others. It's about the only work of his I ever listen to: The interlocking rhythms and melodies build inexorably toward some really swinging jazzy orchestration and it has "watershed" status in Minimalism similar to Terry Riley's "In C." This band of university students and volunteers honed their performance to such a high level they stunned audiences—including the critic for the *New York Times*—at the "Bang on a Can" music festival. They later recorded the work in surround in the St. Cecilia Music Centre, and here it is. Comparison with the original Nonesuch CD shows few musical differences and this SACD puts the players all around you—adding 100% to the musical experience—and the sound is more transparent and impactful. A "New Music" must-have!

Wayne Horvitz Gravitas Quartet|One Dance Alone|Songlines MC SACD SGL SA 1571-2
Many Songlines SACDs make for difficult listening. However, this one's second track—*A Walk in the Rain* —immediately caught my ear. At first, even though the spatiality and realistic presence of each of the four acoustic instruments is uncanny, I felt all the other tracks wandered too far from their melodic and tonal centers to interest me, but by my third repetition I began really enjoying the entire disc, even in ordinary stereo. The Gravitas Quartet plays in a sort of chamber version of what used to be called the "Third Stream." There is no rhythm section and no lead saxophone or trumpet. An adventurous statement from a leading pianist and composer in the alternative jazz area, this album shows how a seemingly classically-oriented chamber ensemble can blend jazz, blues, and creative improvisation and come up with new expressive possibilities without resorting to such strictures as the sonata form or big band swing.

Wei Li|Guzheng Solo|Red Cliff Capriccio|First Impression Music FIMK2HD076
The amazing thing about K2 HD releases is that JVC's engineers have somehow found a way to upsample recordings to 192k/24-bit and then super-bit-map them down to CD standard while retaining most of the enhanced fidelity. This new FIM release was engineered by Reference Recordings' Keith Johnson, renowned for the amazing presence and realism in his recordings. Featuring Wei Li, the world's leading guzheng player, it will naturally appeal to audio buffs of Chinese background, but when you hear the broad range of exquisite string sounds cascading from the speakers you, too, will be captivated. I found the sounds no different—and perhaps offering even a wider variety—than a solo classical guitar recital. The expressive power of this string instrument is astonishing. Its metal strings deliver powerful attack transients yet the quiet lyrical passages are as soft and delicate as if they were played on a koto. The realism of the K2 HD recording is truly astounding.



Lars Erstrand|Swingcerely Yours|Opus 3 MC SACD CD22081
Seventy-two-year-old Lars Erstrand is one of the world's best-known jazz vibists. His style is rooted in the Swing Era and his general approach is close to that of the great Lionel Hampton. This SACD is a compendium of tracks from his past CDs and LPs for Opus 3. All were recorded two-channel with a Blumlein setup, so they've been able to recover the ambient information for the surround channels. The disc's first two sets of excerpts are from 1983 and '84. Erstrand and guitarist Bertil Fernqvist turn in an absolutely lovely interpretation of the slow movement of the *Concerto D'Aranjuez*. The next sets from 1986 pay tribute to the Benny Goodman Quartet, with tunes closely identified with Benny, including a swinging *Stompin' at the Savoy*. Tracks from the "Dream Dancing" album follow, some featuring Kjell Ohman on B-3, so the LFE channel gets some action: one track is actually titled *One for the Woofer*!