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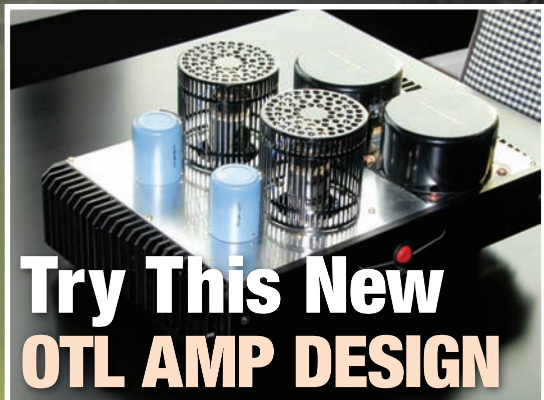

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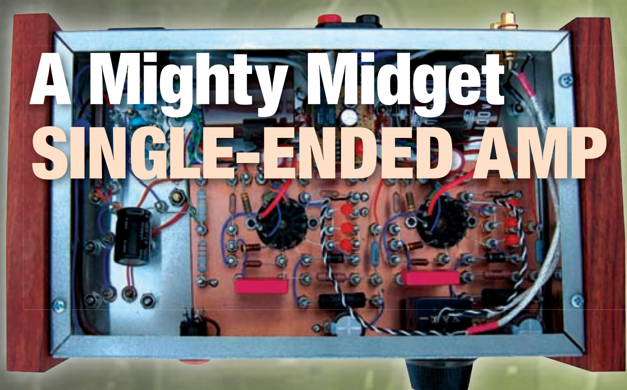

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Of Drugs, Constitutions & Reviews

Human nature, regardless of whatever else about the world changes, remains statically the same. *The New York Times*, as well as PBS, carried stories recently about unpublished studies of the effectiveness of two antidepressants (1/17/08). The published tests reported 60% effectiveness, with 40% of the testees receiving a placebo. However, when the unpublished tests were added to the totals, the drugs were only slightly better than the placebo for depression sufferers.

The New England Journal of Medicine investigated why only some of the trials were not being published. Other previous studies of this issue found much the same reluctance to publish unfavorable test results. There is more to the story. In one, 74 trials of 12 drugs, 90% of the favorable results made it into print, while only 14% of the poor results did. Both the pharmaceutical industry and Congress have recently taken steps to require manufacturers to make full information public, and for several reasons: safety of patients and protection of investors, among others.

This news sounded rather like an echo to me. The main characters in this episode seemed similar to those who participated in an AES forum in the eighties which was billed as "Are Equipment Reviews Accurate?" The participants included all but one of the major reviewers from the audio consumer magazines of the day. The attendees rapidly shifted the subject to "Why aren't bad reviews ever published?" The most popular rejoinder at that time was, "Nobody wants to read bad reviews." It turns out that, on later queries I made to those who were then reviewers, the *publishers* did not want to publish bad reviews.

The reasons are not hard to find. American publishers did not then, or now, wish to upset advertisers, and the reasons for that are certainly not difficult to deduce. At the time I queried John Crabbe, editor of *Hi-Fi News & Record Review* (as it was then), about the situation. He suggested that British reviews were sometimes critical of certain equipment, and that this might result in modifications. British reviewers present at the forum were visibly and vocally scornful of audio review practice in the US.

We've published less than favorable reviews. Two cases had sharply contrasting results. In one, we reluctantly published a report by two of our most competent reviewers who were unhappy to find their negative result but were confident that they were correct. Nor could they discover what the basis of the problem with the equipment was. The manufacturer was not happy at all and thought we were mistaken—and canceled his advertising. In another case our reviewer found a shortcoming in a tube tester made by a British firm. Not too long afterward, we received a revised sample which was right on spec, news of which we happily published.

It seems strange that bad test results of any magnitude in the drug industry do not seem to send the makers back to the drawing board. Doubtless this does happen in some companies. This question has become extremely personal for me. A close member of my family took a diabetes medication—one pill a day—for 422 days. On advice of his physician, he stopped, and two months later suffered congestive heart failure. After a two-week hospital stay and three months intensive rehabilitation, he is largely recovered. However, the pharmaceutical company is facing a class-action lawsuit.

By a strange coincidence, just as news of the prevalence of unpublished unfavorable tests appeared in the media, I was reading "The Genius of America," by Eric Lane and Michael Oreskes (Bloomsbury, 252pp, $23.95, 2007). The subtitle reads "How the Constitution Saved Our Country and Why It Can Again." The authors, one a lawyer, the other a distinguished editor, have one main point: the Constitution, unlike the prior organizing document of the United States, the Articles of Confederation, is firmly and clearly based on a belief in the basic selfishness of every person. The earlier document assumed that everyone would act in a gentlemanly and polite manner, following the rules of honor.

The 13 years between 1776 and 1789 conclusively demonstrated that such a high-minded based government was a disastrous failure. Although most of the delegates to what turned out to be the constitutional convention in the summer of '89 were deists, they knew enough about Puritan Calvinism to recognize that human beings are usually self-centered and will act in their own self-interest.

The Constitution is constructed with three balanced powers. In order for anything to be accomplished, cooperation is absolutely required between these. In any given issue, no participant should go away from a final decision completely happy with the result. Most would agree, I think, that we have just lived through seven years which graphically and tragically demonstrate the truth of what the delegates of 1789 assumed. The Constitution has been seriously violated in deed and spirit and the health of this country, because we have not honored its precepts, has suffered serious damage, whether in the drug business, governments, or in the audio publishing business.—E.T.D.

CONTRIBUTORS

Edward T. Dell, Jr. (Editorial—"Of Drugs, Constitutions & Reviews," p. 6) is editor/publisher of *audioXpress*, *Voice Coil*, and *Multi Media Manufacturer* magazines.

Nelson Pass ("The F5 Power Amplifier," p. 8) is the proprietor of Pass Laboratories and a frequent contributor to *audioXpress*.

Pete Millett ("The Mighty Midget," p. 17) 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.

Born and educated in England and now living in New Zealand for the past 30 plus years, **Terry Bicknell** ("Valve (Tube) Regulated Power Supplies," p. 26) trained in the British Telecom system. He has been in electronics all his life and closely connected with audio for most of it. He taught electronics in the NZ polytechnic system for a number of years. Because he developed an interest in older technologies, he formed Renascent Audio, specializing in refurbishing and designing (mainly) valved audio, studio, and test equipment.

AJ van Doorn ("Build a Hybrid SE OTL Amp," p. 34) resides in The Netherlands.

After 40 years in engineering and product management in the consumer, industrial, and telecommunications field, **Ken Bird** ("The Sidewoofer Line Array," p. 41) retired to his native Missouri, where he pursues his interest in building speakers and tube amplifiers, along with model railroading, amateur radio, and digital photography.

By Nelson Pass

The F5 Power Amplifier

This minimalist amp design is so simple, well, even a caveman can do it.

As many of you may know, First Watt is dedicated to exploring the performance quality of small, simple power amplifiers. Over the past four years, five such amplifiers have been designed as concept pieces and produced in limited quantities.

The F1 and F2 explored the possibilities of current source operation with single-stage class-A circuits and no feedback. The Aleph J used JFET devices for the front end of a two-stage single-ended class-A amplifier. The F3 achieved very low distortion using power JFETs in a single-stage, single-ended class-A circuit. The F4 demonstrated that an amplifier did not necessarily require voltage gain to be useful.

You can follow the progression at www.firstwatt.com.

Here is the F5. We want to further push the performance boundaries of simple little amplifiers with a FET two-stage class-A push-pull design.

A QUICKIE TUTORIAL ON FET AMPLIFIERS

One of the aims of these articles is to encourage people to build amplifiers, so here is some tutorial material to get beginners going. I have written up some of this material before ("The A75," *Audio Amateur* 4/92), but that was 15 years ago, and maybe it will be helpful to repeat bits of it. All 31 years' worth can be found at www.passdiy.com and related links.

I'm assuming that you understand the concepts of voltage, current, and resistance. If you already know how a FET works, you can skip ahead.

Figure 1 shows an N channel FET, a quantum mechanical black box with three connections. This device is meant to function as a valve, a little bit like a water faucet. In this illustration, the drain (D) of the FET is attached to an electrical power supply, analogous to the pressurized water supply on the other side of the faucet. You can imagine the pipe as wire, and the tank of water as a battery or even a charged-up capacitor.

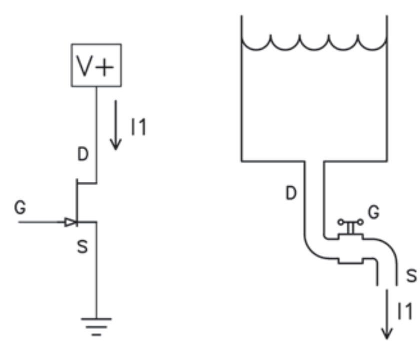

FIGURE 1: FET transistor vs. water faucet.

To continue the metaphor, the voltage of the supply is the water pressure, and the water flowing from the supply is the electrical current. The source (S) connection of the FET is the output of the faucet. The gate (G) of the FET is the control pin, and like the handle on the faucet, it controls the amount of electrical current through the FET from the drain to the source. For the FET, this control is a function of the relative voltage between the gate and source pins. For an N channel FET, raising the gate positively with respect to source increases the current flow.

Yes, I know some of you are thinking that maybe the source should be on top and the drain on the bottom, but they're not. You might not want to call a quantum mechanic if your plumbing stops up.

The idea that the current going through the FET transistor is controlled by the voltage between the gate and the source pins remains the key idea, and if you have that firmly fixed, you can leave the waterworks metaphor behind.

FETs come in different types. There are two polarities, N channel and P channel. There are different voltage, current, and power ratings, and different semiconductor processes resulting in JFETs and MOSFETs. In all of them, the current from the drain to the source is controlled by the voltage between the gate and the source.

A FET is a three pin device, and there are three ways to amplify with them. **Figure 2** illustrates these with an N channel FET:

- Common-Source (CS) is the connection that can give both voltage and current gain in a circuit. The input voltage (shown as a little graphical sine wave) goes to the gate, and the output is taken from the drain and appears across a resistor between the drain and the supply voltage. The source is grounded, and doesn't show a signal voltage, and that's why it's called common-source. Note that the output voltage is inverted in phase from the input voltage.

- Common-Drain (CD) gives current gain only, and is also known as source follower, because the output voltage across the source resistor is nearly identical to the input voltage at the gate. While the drain is usually attached to a DC voltage value, the AC voltage is ideally zero, and so it is called common-drain.
- Common-Gate gives non-inverted voltage gain only, with the input signal going into the source and coming out the drain. The gate is grounded.

Figure 2 shows what happens only to AC signals, but it doesn't illustrate the DC voltage and current values that the FETs need in order to operate. These DC values are often referred to as the bias of the device, and you will hear this term a lot with respect to amplifiers.

The optimum bias values vary by the device and the needs of the circuit. In general, to function as an amplifier, the FET needs to have at least a few volts between the drain and the source. If the FET is an N channel type, the drain must be at least a few volts positive with respect to the source. If it's a P channel type, the drain must be negative with respect to the source.

In addition, the gate of the FET must be placed at a DC value relative to the source so that the current and voltage of the FET is positioned in the linear region between the extremes of voltage and current—somewhere between all the way on and all the way off. It is in the middle ground where the distortion is low. Generally for N channel JFETs, it is with the gate voltage at 0 or slightly negative with respect to the source, and for N channel MOSFETs the gate voltage is a couple volts positive. One of the important functions of a circuit is to set up the DC conditions of gain stages so that the devices have stable operation in this region. For every amplifying circuit, there will be a "sweet spot" of voltage

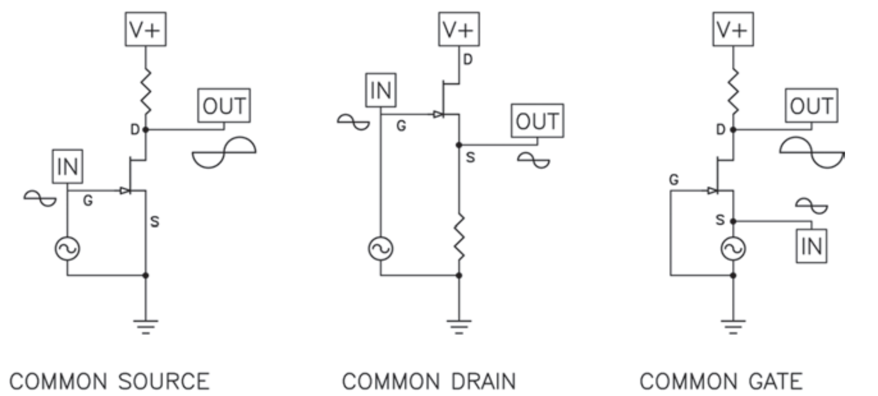

FIGURE 2: The three operating modes.

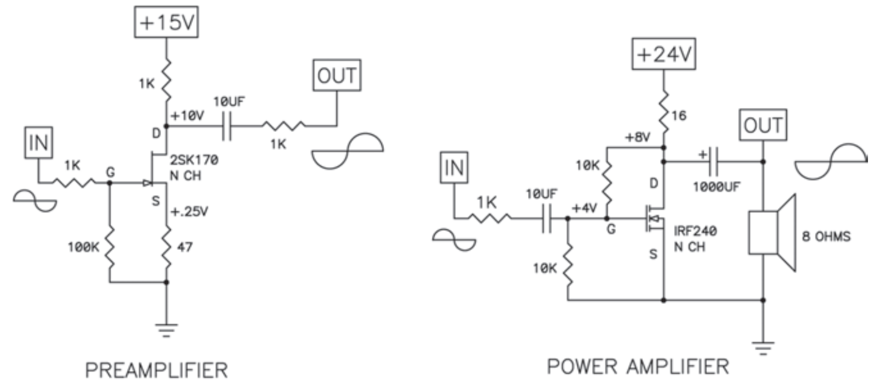

FIGURE 3: Common-source examples.

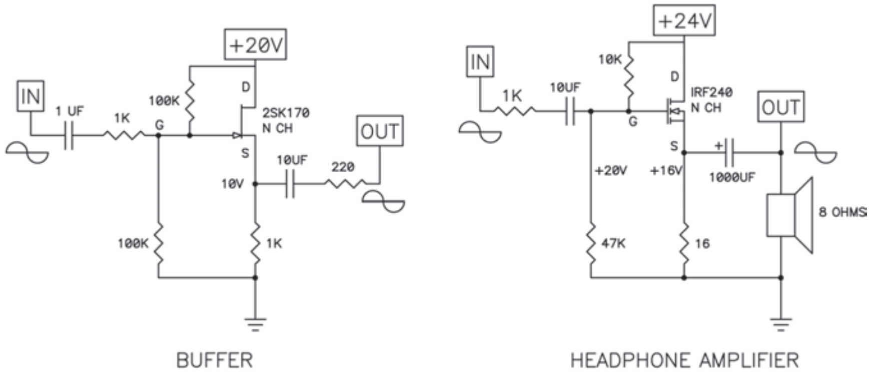

FIGURE 4: Common-drain examples.

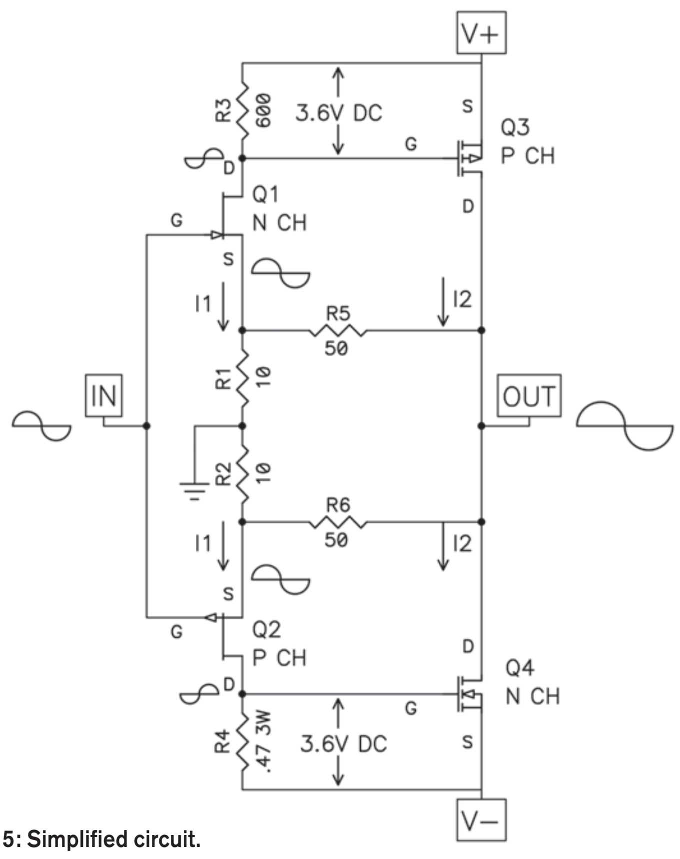

FIGURE 5: Simplified circuit.

and current which will give the best overall performance.

Figure 3 shows some examples of real circuits that illustrate simple common-source amplifiers and the bias voltages and currents that would be typical for them. You can build both of these circuits, which will work as shown.

On the left you see an example of a simple preamplifier stage with a gain of roughly 10 times (20dB). The JFET is self-biased in this circuit: With 5mA going through the JFET, the gate needs to be at about -0.25V with respect to the source. Notice 47Ω in series with the source, raising its voltage to +0.25V, which conveniently allows you to bias the gate at 0V DC, or ground.

On the right is a power MOSFET set up as a simple power amplifier, intended to deliver about 1W into an 8Ω loudspeaker. The 16Ω power resistor performs as a current source to the circuit, and the two 10kΩ resistors on the input set the DC value for the gate to about 4V to get it to conduct a 1A bias current. For a more powerful version of this, see Zen Variations #1 (*audioXpress*, March 2002).

Figure 4 shows some real common-drain amplifiers, a line-level buffer that you can use in active filters, and a headphone amplifier. In both cases there is no voltage gain in the amplifier, but there is current gain. They both have high input impedance and low output impedance.

Common-gate operation is encountered less often, usually found in a "cascode" connection with another device. Discussions of cascoding with good examples are found in Zen Variations 8 and 9 (*audioXpress*, January 2006, May 2006) and also the article "Cascode Amplifiers" originally published in *Audio Magazine*, March 1978.

SIMPLIFIED F5 CIRCUIT

Figure 5 shows the simplest imaginable version possible of the F5. The topology is familiar: a two-stage conjugate-complementary circuit using two JFET transistors for the input amplification and two power MOSFET devices for the output.

JFETs Q1 and Q2 form a complementary common-source stage, with the input appearing at their gates and output at their drains. The JFETs are class A self-biased at about 6mA, and so a current I1 comes out of their drain connections and produces a voltage of about 3.6V across R3 and R4, whose values are about 600Ω. This 3.6V DC value is necessary to bias the power MOSFETs Q3 and Q4 into conduction. The voltage gain of Q1 and Q2 appears across each of these resistors for a gain of about 10 each.

The gain of 10 for the input stage comes from the ratio of the 600Ω divided by the apparent resistance from the source to ground, which is roughly 60Ω. This 60Ω number comes from the inverse of the JFET's transconductance plus the 10Ω actual resistor. The transconductance for a JFET is the ratio of current change against input voltage change. The gain of this part is typically 0.02 siemens, or 0.02A per control volt, and if you invert that you get 50Ω (R = V/I). So it looks as though there is 50Ω inside the part (although there isn't), and to that you add the 10Ω of the real resistor to get 60Ω.

Q3 and Q4 do the heavy lifting in this circuit, providing the large output current needed to drive the loudspeaker. The DC voltage appearing from the gate to the source of these devices is about 3.6V, and this biases the MOSFETs to about 1.3A.

For this sort of circuit, a 1.3A bias means that the amplifier will operate class A to 2.6A of output current. To understand this, imagine a condition in which Q3 and Q4 are idling at 1.3A, so that all the current is going from the V+ voltage rail to the V- voltage rail, and none is going through the loudspeaker.

When a positive voltage appears at the gates of Q1 and Q2, it makes the current through Q1 increase and the current through Q2 decrease. The resulting voltages across R3 and R4 make the current through Q3 increase and the current through Q4 decrease. This makes the output voltage go positive. As the positive input voltage increases, you approach the point at which Q3 is conducting 2.6A and Q4 is conducting 0 amps—and all of the 2.6A goes through the loudspeaker.

The power of 2.6A into 8Ω is $I^2 * R$, or 2.6 * 2.6 * 8 = 54W. This is the peak value, and the nature of an undistorted sine wave is that the peak wattage is twice the average, so this circuit would operate 27W average class A into 8Ω. At currents above 2.6A one of the transistors will shut off, leaving the other to continue to increase beyond the 2.6A in what is known as class AB.

This circuit employs feedback to improve the performance in a number of ways. Feedback sets the gain, lowers the distortion, improves the bandwidth, and produces the output impedance (damping factor) of the amplifier.

The feedback mechanism for this amplifier is R3 through R6, a dual pair of low impedance voltage dividers which feed the output to the source pins of Q1 and Q2. Low impedance feedback has been (incorrectly) referred to as "current feedback," and it is popular in simple high-speed linear circuits. One of the charms of this arrangement is that unlike the classic two transistor differential pair, the drive current available exceeds the bias of the input stage.

Something different about this example is that each JFET has its own feedback—there are two separate feedback loops to this amplifier, so that the loop of Q1/Q3 is independent of the loop for Q2/Q4.

The circuit of **Fig. 5** illustrates the basics and does in fact work, but has a tendency toward instability. Local parasitic oscillation is often seen, and the bias drifts with temperature, requiring a lengthy adjustment period. This simplified circuit also has no provision for adjusting against the variations to be seen between real FETs that you will encounter.

ACTUAL WORKING CIRCUIT

Figure 6 shows a completely functional version of the circuit, with useful enhancements: the addition of R7 and R8 in parallel with R5 and R6 to share the dissipation, allowing inexpensive 3W resistors to do the job.

You never know what the amplifier will be attached to, so the input of the amplifier now sports R9 to avoid parasitic oscillation of the input JFETs, and R10 to ensure that the input has a default reference to ground when there is nothing connected.

R11 and R12 are 3W power resistors added to the source pins of the MOSFETs to increase thermal stability and serve as convenient current-sensing elements. R13 and R14 are placed in series with the gates of the MOSFETs to prevent parasitic oscillations, just as you did with R9 on the input stage.

R2 and R4 have had their values increased and then placed in parallel with 5kΩ trim potentiometers P1 and P2. This allows adjustment of the output stage bias current and also the output DC offset.

You can stop here, and the amplifier will be fully functional. The remaining additions will enhance temperature tracking and provide for output current limiting.

Th1 and Th2 are small 4.7kΩ thermistors placed in series with R15 and R16, respectively. The resistance of the thermistor declines with temperature, and if placed close to the output transistors will help compensate for thermal drift. You can build the amplifier without them, but will have a longer warm-up time and you will spend more time adjusting the bias.

On the positive half of the output stage, Q5, R17, R19, and R21 will arbitrarily limit the output of the amplifier in the event that you accidentally set the bias too high or short the amplifier's output. Q6, R18, R20, and R22 do the same for the negative half of the amplifier. The output MOSFETs are easily capable of passing very high currents at the request of Q1 and Q2, and it is wise to set a limit in case of accident.

Understanding the limiting circuit is easy. Q5 and Q6 look at the voltages across R11 and R12, and will start to conduct when their base to emitter voltage exceeds 0.4V or so. At 0.4V Q5 and Q6 draw off enough drive current from Q1 and Q2 to start producing measured distortion, and hard limiting occurs when the voltage driving Q5 or Q6 is increased to 0.6V.

Because the 1.3A bias through the 0.47Ω source resistors provides 0.6V already, you must divide that voltage down so that limiting occurs at a higher current. You do this for Q5 with divider resistors R17 and R19, and for Q6 with R18 and R20. R21 and R22 allow you to also adjust the limit point based on some information from the

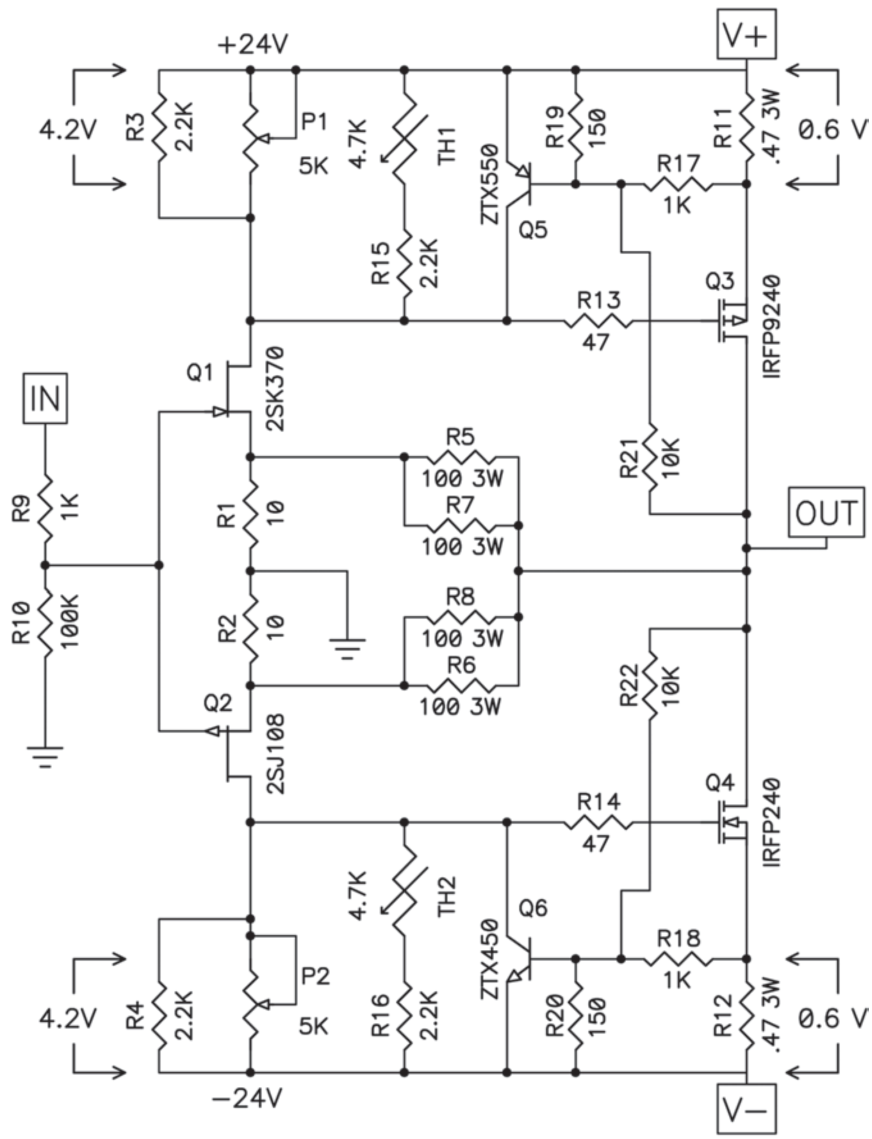

FIGURE 6: F5 amplifier.

output voltage.

Where do you want to limit? The amplifier with 24 supply rails should be good for a 50W peak into 8Ω, or about 2.5A. For a 4Ω load you would want 5A, and for a 2Ω load, you want 10A. Because you only have two output devices, you probably are best off stopping there.

Calculate the values as follows, starting with arbitrary values of 1kΩ for R17 and R19. At 10A peak, the voltage across R11 or R12 will be about 4.7V. Choose R19 and R21 to divide that down to 0.6V, and solve the formula

0.6V/4.7V = R19/(R17 + R19)
Because R17 is already 1kΩ, you simplify to 0.127 = R19/(1000 + R19)
And you get R19 = 150Ω

10A into a dead short is about 200W dissipation on an output transistor, which will probably exceed its rating. R21 adds a "foldback" characteristic to this so that the current value will be lower into a shorted output. To ensure against damage into dead shorts, you want to limit the zero crossing current to about 5A, keeping the dissipation just this side of destruction.

Current limiting gets a bad rap in general, but I think it's a matter of where and how the limits are set. If you build this amplifier, you are, of course, welcome to delete this portion of the circuit, and, of course, you will not complain if a shorted output lets the smoke out of your amplifier. As it is, you will still be taking some chances with a shorted output.

PARTS SELECTION

The input JFETs used are 2SK170 or 2SK370 for the N channel parts (Q1), and 2SJ74 or 2SJ108 for the P channel parts (Q2). In these cases the Idss selection code is BL, although V and GR types can be made to work with adjustments to their drain resistances. The primary thing about these particular parts is the transconductance figure of 20ms—many of the potential substitutes are much lower at 4 to 10ms.

The output MOSFETs are the IRF240 N channel type (Q3) and the IRD9240 P channel type (Q4). They will need a voltage rating in excess of 50V, a current rating of about 15 to 20A, and a dissipation of about 150W. Comparable parts are widely available, and here I used Fairchild FQA19N20C and FQA12P20.

The remaining parts can be generic or special as you see fit. I used Dale RN55D type ¼W resistors and the 3W Panasonic power resistors from Digi-Key, where you can also get the 4.7kΩ thermistors and trimpots.

Audiophiles are often concerned about the effects of capacitors in audio circuits, but a quick examination of the schematic will put your concerns to rest.

POWER SUPPLY

The power supply of the tested amplifier is ±24V and should be rated at 6A continuous duty, and more than 10A peak per channel. Ordinary unregulated supplies will work fine, and if you need suggestions the Zen Variations #5 (*audioXpress*, October 2003) and Zen Variations #3 (*audioXpress*, August 2002) both have good examples of regu-

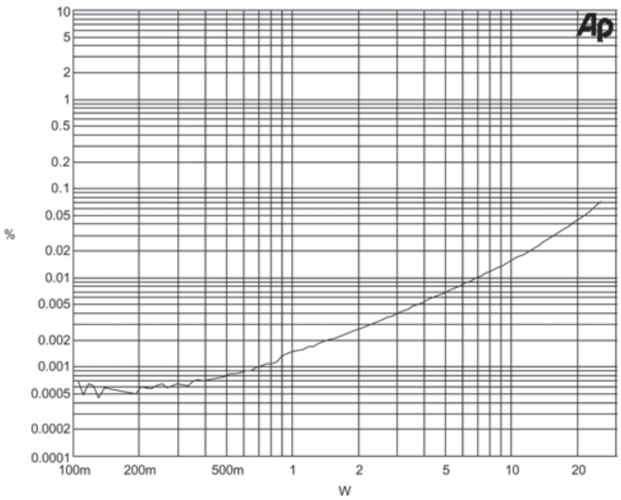

FIGURE 7: Distortion vs. power at 8Ω at 1kHz.

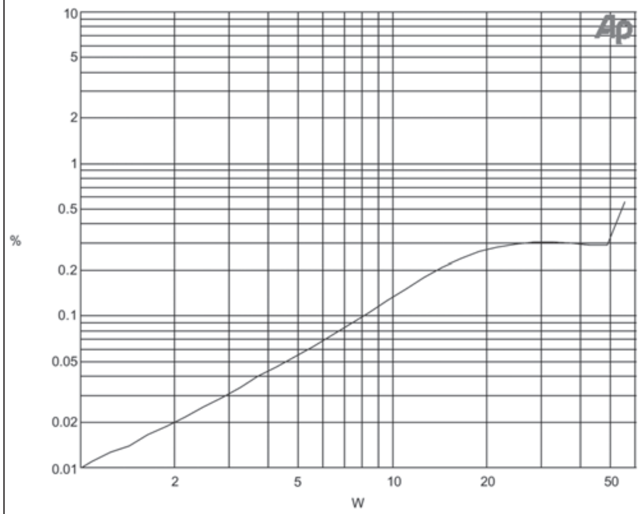

FIGURE 10: Distortion vs. power at 2Ω at 1kHz.

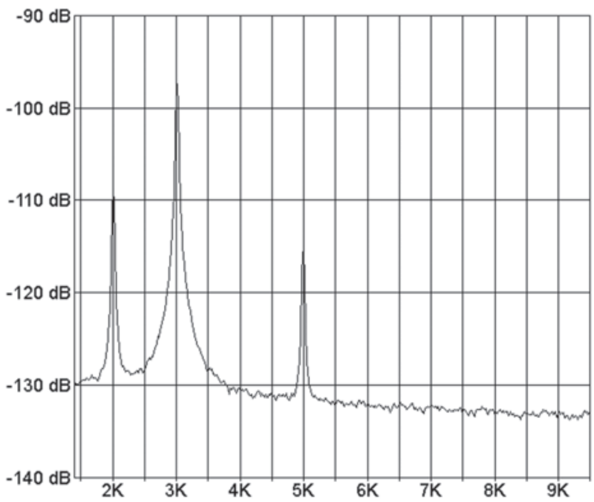

FIGURE 8: Distortion spectrum at 1W at 1kHz.

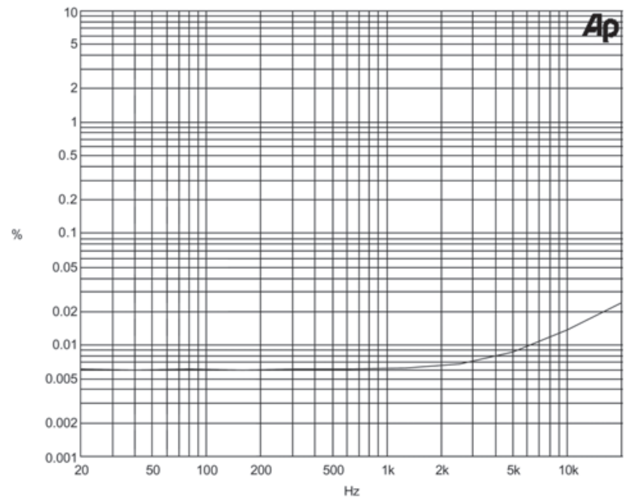

FIGURE 11: Distortion vs. frequency at 8Ω at 2W.

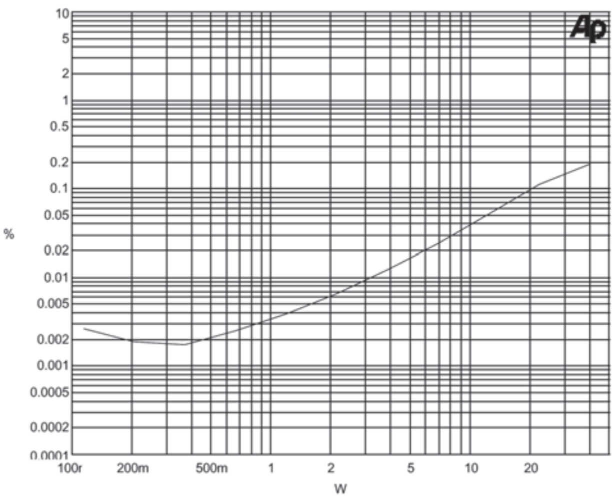

FIGURE 9: Distortion vs. power at 4Ω at 1kHz.

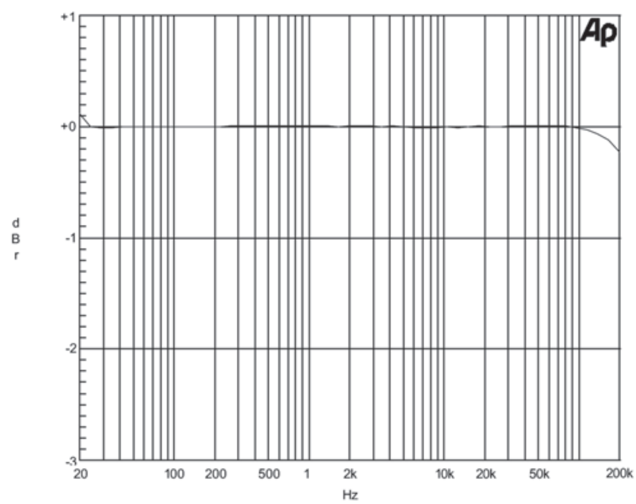

FIGURE 12: Frequency response.

lators and the raw supplies that feed them. I recommend rail voltages from 23 to 25V.

HEATSINKING

At 1.3A per channel, you will see idle heat dissipation of 62W. To keep the temperature rise of the heatsink to 20° C above the ambient temperature, you will need a heatsink rated at about 0.6° C/watt for each transistor. An example of this would be a chunk of finned aluminum, with a series of 2″ fins attached to an 8″ × 6″ base. You will need two per channel.

The output devices need to be intimately attached to the heatsink. The mounting surface on the heatsink should be at least ¼″ thick and smooth and shiny. You can use silicone pads for insulation, but Mica and silicone grease are the best.

INITIAL ADJUSTMENT

Before applying power to the amplifier, you must set the values of P1 and P2 to their minimum. Verify this with an ohmmeter. When it comes time to "fire" up the amp for the first time, if you have a Variac, use it, fusing the AC line to the amplifier with a 1A fast blow fuse. Turn the Variac up slowly, and if you haven't popped the fuse, then go ahead and confirm the rail voltages to the channels.

Each channel does not need to be attached to a load in order to adjust it. If the only load you have is the loudspeaker, I advise against using it during adjustment. For each channel you will be adjusting P1 and P2 alternately in order to achieve 0 volts DC at the output and 0.59V across R11 and R12. Each time you adjust P1 you will probably need to go back and adjust P2 again, so I recommend adjusting the pots in half-measures, alternately setting the pots half-way to their voltage goals and measuring the DC values. Unless there is something very wrong, when the output is at 0V DC, the values across R11 and R12 will be equal.

In spite of the thermal compensation in the circuit, you should assume that there will be drift as the heatsink temperature rises, and you will need to readjust the values over the course of an hour or two. Usually it is best to start out bias adjustment low, at maybe 0.40V across R11 and R12 until the amp is warmed up a bit.

You should be able to get the output DC offset down to 10mV or so, and I consider 50mV the highest acceptable figure for this amplifier when warmed up. After you have operated the amplifier for a few weeks, it is a good idea to check and adjust the offset again after the parts have been burned in.

MEASURED PERFORMANCE

The gain of the circuit is 15dB. The input impedance is 101kΩ, and the output impedance is 0.1Ω, for a damping factor of 80.

The noise of a properly laid out circuit is about 30µV with a quiet supply, and raw supply with two 29,000µF capacitors and 70mV ripple will give about 100µV noise (measured in the band from 20 to 20,000Hz).

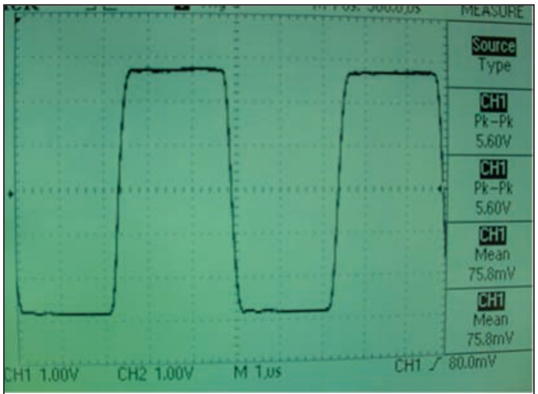

FIGURE 13: 200kHz square wave at 1W.

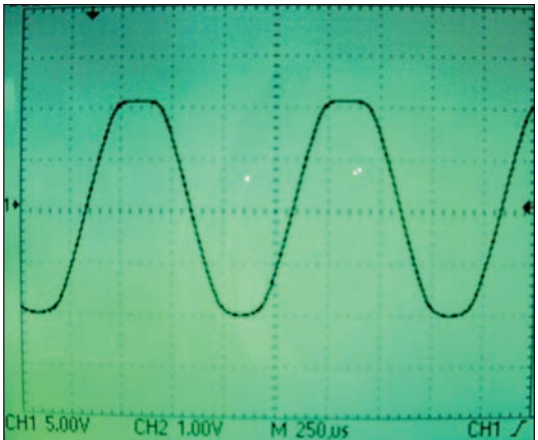

FIGURE 14: Clipping into 1Ω.

Figure 7 shows the harmonic distortion and noise plotted against output power, taken at 1kHz into 8Ω. This is the lowest distortion yet achieved by either a Zen or First Watt amplifier, descending to .001% below 1W.

In **Fig. 8**, spectrum distribution of the distortion at 1W, you can see the dominant 3rd harmonic, as well as 2nd, 4th and 5th. The vertical scale is in decibels relative to 1W. You can also see the noise floor of the system and amplifier at -135dB or so below 1W.

Figure 9 illustrates the harmonic distortion + noise plotted against output power 4Ω load, reaching 0.2% at 40W.

Figure 10 shows the harmonic distortion and noise plotted against output power into 2Ω, where the amplifier begins to clip at 0.3% at 50W.

Figure 11 depicts the distortion and noise versus frequency at 2W into 8Ω.

Figure 12 displays the frequency response, flat to DC and down -0.25dB at 200kHz.

Figure 13 is the square wave response at 200kHz at 1W. The amplifier is flat to DC, and -0.25dB at 200kHz.

If you have the equipment to see it, you may find that different gain devices will give you a small peak somewhere just below 1MHz. I have found that you can trim this by playing with the values of R13 and R14, but as both values approach 0 ohms, you are likely to see parasitic oscillations.

You can additionally limit frequency response with an input capacitor across R10, and also in the feedback loop with capacitance across R5 and/or R8. As you can see, I have chosen not to.

Figure 14 shows the behavior of the amplifier clipping into 1Ω at ±10A.

CONCLUSION

So there you have it, a nice little amplifier without too many parts that sounds great. *aX*

The **Mighty Midget**

This tiny stereo amp puts out over 3W per channel using only two tubes.

By Pete Millett

PHOTO 1: The midget SE amp.

Funny how some projects get started.

I was wandering around the little stalls that sell electronic parts under the train tracks in Akihabara. I often go to Japan on business, and usually find time to go to Akihabara to look around, see what's new, and pick up those few parts that are actually cheaper in Japan than in the US (such as Alps volume control pots).

In a stall that sells transformers (and nothing else), I spotted a cool-looking little transformer in an enclosed black metal case. About the size of a 20W filament transformer, it stood out from the dozens of other power transformers. I took a closer look, and it turned out to be a single-ended output transformer. The label said 8W, which seemed a little hard to believe for its size.

OK, sure, this was no Tango or Tamura piece of iron. But for only $20? And it was just so...*cute*. So I picked up a pair, not expecting much.

When I returned home and took a closer look, the transformers really did look pretty nice. The primary and secondary were tapped so you could set it up for a number of impedances between 2.5kΩ and 7kΩ. A few casual measurements looked encouraging. So what to do?

Design an amplifier around them, of course.

MINIMUM TUBE COUNT AMPLIFIER

Because these output transformers were so small, I thought I would design a physically small amp to go along with them. I've also always been fascinated by the complex, multi-element tubes that were designed near the end of the "vacuum age"; I figured I could probably come up with something using one of them.

My first thought was to use a triode/pentode tube, such as the ECL86/6GW8, that was designed for audio amp applications. The ECL86 is—more or less—a 12AX7 triode section together with a power pentode similar to the EL84. But a quick look on the web showed that they were too expensive, going for $30 each!

The next candidate I considered was a dual dissimilar triode, such as the 6EM7. I've heard from many happy 6EM7 builders, who used the bigger 10W triode as a SE output stage and the smaller triode as a driver. This seemed promising, but when I put one on a breadboard, I wasn't very happy about the large amount of distortion and low output power that occurred.

OK, time to become more imaginative and dig around in my boxes of tubes. Whenever I feel "creative," I seem to gravitate toward oddball tubes, such as loctals and compactrons. Surely, there must be some multi-element compactron tubes that could work?

Between some archaeological explorations around my lab and some Internet searching, I came up with a list of compactron tubes that contain a power section plus a small-signal section. There were dual triodes such as the 6EM7, and some triode/beam tube combos, both of which were designed for TV vertical deflection oscillator/amplifier use. There were also triode/high Gm pentode combos, used as a color blanker or AGC amp plus video output amp in color TVs. Then there were some tubes with a dual-control pentode for FM detector use along with a beam power tube for an audio output, again meant for TV use.

Looking over datasheets and checking prices (I imposed an upper limit of $5 each—hey, I have two kids to send to college!) and availability online produced a short list: the 6T10, 6G11, 6AG9, 6AH9, 6AL11, and 6LR8. All had promise, and I think any of them could be made into a respectable audio amplifier.

To make a long story short, after doing some experiments I settled on the 6T10 tube for this amplifier design. The 6T10 has a beam-power section somewhat similar to a 6V6 (but with higher Gm) and a "dual-control" pentode in one 12-pin compactron envelope. It was designed for TV sound use, providing up to 4.2W out in SE pentode mode. It's affordable as well—I bought a handful from Antique Electronic Supply for $1 each.

And in the right circuit, it makes a

fine single-ended audio amplifier.

DUAL-CONTROL PENTODE?

So, what's a dual-control pentode, you ask?

Normal pentodes use the suppressor grid, or G3, to repel electrons that become dislodged from the plate back to the plate (secondary emission). To perform this function without getting in the way of the desired electron stream from cathode to plate, G3 is typically a widely-spaced wire grid that is held at ground (or cathode) potential.

A dual-control pentode is different in that G3 is designed to be used as a second control input, so that varying its voltage has an effect on plate current. To accomplish this, G3 in a dual-control pentode is wound more closely than in a normal pentode. Because of its increased density, it intercepts some of the electrons on their way to the plate. This causes it to draw current (sometimes even more than the plate), so it must have a more substantial structure that can dissipate some power.

Dual-control pentodes were intended mostly for detector and mixer applications, in which one RF signal is applied to G1 and a second to G3.

If you look at the plate curves for a dual-control pentode (the 6T10 curves are shown in **Fig. 1**), you'll see that the G1 curves look basically like a normal pentode, but are a bit rounded, somewhat nonlinear, and show a fairly low plate resistance (a bad thing for a pentode). If you look at the G3 curves, you see a strange characteristic somewhere between a pentode and a triode, which is not particularly linear. So, how do you make a decent audio amp out of *that*?

There are a couple of possibilities. One is a triode connection; several choices will work and make a decent triode, including connecting G2, G3, and the plate together, or using G2 and G3 as the plate and not connecting the "real plate" at all. But I derived better results using the tube as a pentode, by biasing G3 a few volts positive, which has a dramatic shape on the G1 plate curves.

You can see this by looking at the plate curves. There is one curve on each of the graphs that corresponds to G1 = G3 = 0V; these curves are identical (note, though, that the vertical scale on the graphs are different). If you look at the G3 graph, notice how the *shape* of the curve changes as you move to G3 = +3V—it becomes more horizontal with a squarer knee, which is what you want a pentode curve to look like to make a linear amplifier. Now imagine the entire family of curves in the G1 graph "squared up" like that. That's the resultant family of plate curves you wind up with by biasing G3 to a positive DC voltage.

PENTODES (AND BEAM TUBES) AS POWER AMPS

As most of you know, there are several ways to use a pentode or beam power tube as an audio amplifier output stage. You can operate it in pentode mode, where G2 is held at a constant voltage; in triode mode, with G2 tied to the plate to emulate a triode; or in ultralinear mode, with G2 fed a portion of the output signal to produce a characteristic somewhere between a triode and a pentode.

In triode mode, the circuit has a low enough plate resistance and low enough distortion that often no feedback is needed to produce a good output stage. The downside is that the amount of power available in triode mode is substantially less than that in pentode mode—in other words, the plate efficiency is not very high. On the other hand, in pentode mode, more output power is available, but the output impedance is very high (and often distortion is as well), so some form of negative feedback is usually needed to produce acceptably low distortion and adequately high damping factor.

A common way to deal with the problems of pentode operation was to put a lot of "extra" gain into the amplifier, then feed a portion of the output (taken from the speaker terminals) back into the input stage.

This is referred to as "global" or "loop" feedback. I've never been a big fan of global feedback, especially where transformers are involved. Achieving good stability can be difficult, and it can produce higher-order distortion artifacts that sound unpleasant. And I'm just not ambitious enough to make it work well.

Another solution is to use local feedback. By "local," I mean feedback around only one amplifying stage—in this case, the output stage. There are several ways to implement local feedback in a pentode stage, because you have three grids and the cathode that you can play with:

- The ultralinear connection is a form of local feedback using G2, usually employing a tapped output transformer to give characteristics midway between a pentode and triode connection.
- Cathode feedback can be used by providing a winding on the transformer connected to the cathode. Even an un-bypassed cathode resistor is a form of cathode feedback.
- In some tubes, G3 can be used to control plate current (à la the dual control pentode), but it isn't very practical with most output tubes (especially beam tubes).
- Signal can be coupled from the plate to G1 ("plate-to-grid" feedback).

PLATE-TO-GRID FEEDBACK

There is an excellent article written by O. H. Schade of RCA in 1938 entitled "Beam Power Tubes" that discusses the application of beam-power tubes in

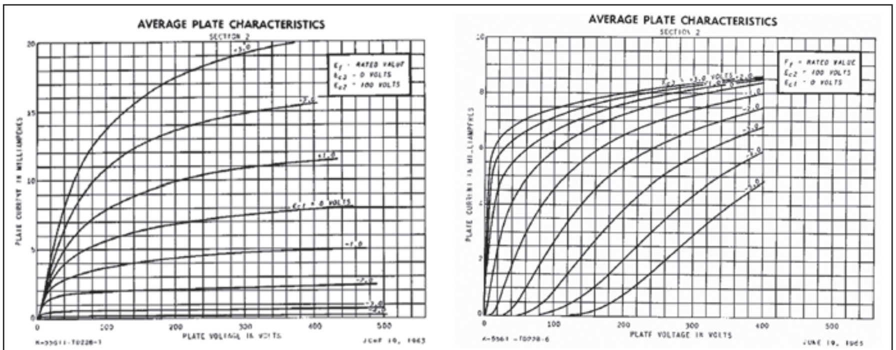

FIGURE 1: 6T10 dual-control pentode plate curves.

audio power amplifiers. (The article is available to download from my website www.pmillett.com). In this article, Mr. Schade discusses the application of local feedback in beam power tubes to produce essentially a "designer triode," having the distortion and output impedance characteristics of a triode, but retaining the power output and plate efficiency of a pentode.

Following Schade's instructions, I plotted plate curves for this "imaginary" triode for some of the tubes I was investigating. **Figure 2** shows these curves for the 6T10 beam power section.

These curves look like those of a pretty good output triode. Note that the curves can be used to show the Rp and mu of the resultant "triode," but the tube is actually running in pentode mode; the grid voltage never really goes outside the range shown on the pentode curves.

I wound up with an operating point just slightly hotter than the one recommended on the 6T10 datasheet for pentode operation. I'm running a plate voltage of about 270V and a plate current of 35mA. This requires a cathode bias voltage of around 8V.

CIRCUIT DETAILS

The amplifier schematic (one channel only) is shown in **Fig. 3**. The driver stage bias is established by a string of LEDs and a zener diode. The cathode sits on one red LED connected to ground, providing about 1.8V of cathode bias. Two more LEDs bias G3 to about 3.6V above the cathode. A 100V 5W zener diode puts G2 at about 104V above the cathode, and a 30k 5W resistor provides current to this string from filtered B+. (I tried to bypass a cap on G2, but it didn't seem to provide any significant improvement, so I left it out in the interest of simplicity.)

The driver plate is connected to filtered B+ with a 64.9kΩ plate load resistor (R2), and also to a 357kΩ resistor that is connected to the output tube plate (R4). R4 provides the plate-to-grid feedback path. OK, technically, it is connected from the plate of the output tube to the plate of the driver tube; but from an AC perspective, the plate of the driver tube is the same as the grid of the output tube, because they are connected by a 0.22µF coupling capacitor. A 240kΩ resistor (R7) provides a DC return for the grid of the output tube.

Setting up this stage is a tricky task of getting the best values for the plate load resistor, feedback resistor, and the grid-to-ground resistor of the power tube. The DC operating point is affected by the plate load resistor as well as the feedback resistor, because it's connected to the plate of the output tube, which is sitting just a tad under B+.

The amount of feedback is set by the voltage divider formed by the feedback resistor working backwards into the output impedance of the driver stage, and into the input resistor of the output stage (R7). The driver output impedance is formed by a parallel combination of R2, R4, and the Rp of the driver tube (which, being a pentode, is quite high). The gain of the driver stage alone is also affected by these same components, so you can see it's not a simple process to figure out ideal values.

I used a combination of engineering, educated guesses, and trial and error to arrive at the values used in the schematic, making distortion and power measurements along the way. I calculated the impedance of the driver plate circuit to be around 30-40kΩ, so the 357kΩ feedback resistor gives somewhere around 10% feedback, which was my target.

The output stage is straightforward, with a 200Ω cathode bias resistor bypassed by a large capacitor. G2 is connected directly to B+ (through a grid stopper resistor). I experimented with different bias resistances, plate voltages, and load impedances, and settled on an operating point close to the datasheet spec. I found that a load impedance of 7kΩ provided a little nicer looking harmonic FFT than 5kΩ at only a small loss of power, so I used the 7kΩ: 8Ω configuration of the baby output transformers.

THE POWER SUPPLY

Because I wanted this amp to be small and inexpensive, a big tube power supply with enormous chokes was out of the question. So I designed a small solid-state supply, using a Hammond 261G6 power transformer I happened to have on the shelf. Strictly for looks, I mounted the transformer inside of a box on top of the chassis.

Figure 4 shows the schematic of the power supply. Following the rectifier is an RC filter stage, then two LC stages. The RC stage (with a small capacitor) limits the peak currents in the rectifiers, which helps reduce noise. The separate LC filters, one for each channel, provide isolation at audio frequency between the two channels in addition to helping to achieve low supply ripple. Splitting the filter also allows the use of small chokes, because each channel draws only around

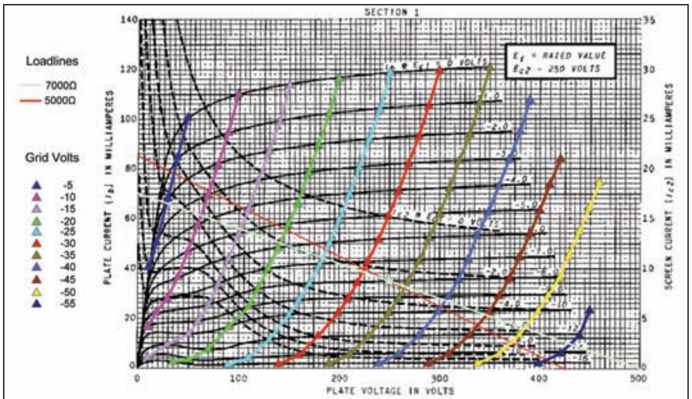

FIGURE 2: 6T10 beam power tube with feedback.

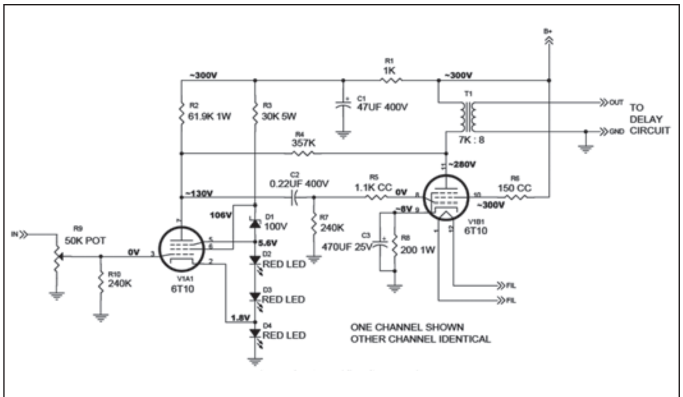

FIGURE 3: Amplifier schematic.

50mA. The chokes I used were surplus potted chokes (from the Danish telephone company!), but any small choke (I suggest the Triad C-3X, which is very inexpensive) will work fine.

Whenever you use solid-state rectifiers, especially inside a small enclosure, you must worry about reverse-recovery noise from the rectifiers getting into the signal path. The use of fast or ultra-fast recovery diodes is a must. I used 600V, 1A ultra-fast recovery MUR160s from On Semiconductor, and added small ceramic capacitors across the diodes.

AN AFTERTHOUGHT: THE MUTING CIRCUIT

After I built the amp and had it running, I discovered that when I turned it off, it made some really nasty noises. I'm not sure exactly why—maybe as the supply voltage dropped one stage was operating in a negative-resistance part of its plate curve—but it emitted a loud squeal for a fraction of a second. It also made a little bit of a thump when power turned on. So I decided to build in a small power-on delay circuit that would keep the speakers disconnected for a while after power-on, and disconnect them immediately when the power switch was turned off.

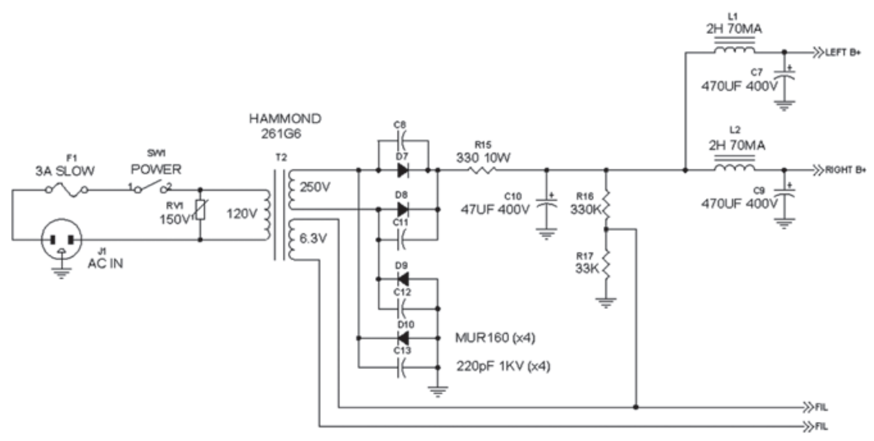

FIGURE 4: Power supply schematic.

The schematic of the circuit is shown in **Fig. 5**. It uses a power-on reset IC, the TL7700, from Texas Instruments, to drive a relay that disconnects the speaker outputs.

The 6.3V filament voltage is rectified and filtered to generate a DC voltage to run the circuit. The filter cap (C4) is not very big, because you want the DC voltage produced to drop very quickly on power-off, and you can tolerate lots of ripple. A dropping resistor (R11) in series reduces the DC voltage so the relay coil receives 5V when it's activated. Note that you may need to use a different value if you use a different relay than the one I used (an Omron G5V-2-H1-DC5).

C6 sets the delay at power-on; with 100µF, the delay is about 20 seconds. R13 and R14 set the threshold voltage at which the circuit starts counting the delay, and the voltage where the relay will open when the power drops. This voltage is set to around 3.8V.

CONSTRUCTION

Wanting to keep the size to a minimum, I built the amp in a 6″ × 10″ × 2″ aluminum chassis (**Photos 1** and **2**). This required a very tight layout. I used single-point terminals screwed to a blank piece of PCB material to hold most of the parts, as I've done in projects I've published in the past. (For details on this construction technique, refer to "A Low-Mu Triode Preamp," *audioXpress* 2/04.) The copper foil of the PCB is used as a ground plane.

I added some oiled bubinga wood to the sides to dress things up a bit, and painted the chassis with aluminum-colored Hammerite™ spray paint.

Photo 3 shows the inside of the chassis. Because space is tight, I temporarily mounted the tube sockets to the PCB material, and wired most of the circuit outside the chassis on the bench. Then I mounted the assembly inside the chassis, using the four tube socket screws to hold the PCB assembly to the chassis. It was so crammed inside that I wound up needing to remove a coupling capacitor to fit the volume control pot in place.

The input wiring is routed along one edge of the chassis, and the AC mains wiring along the opposite edge. I used shielded wire to connect the input RCA jacks to the volume control, though I don't know that it was necessary.

Photo 4 shows a closer view of the terminal board before installation. The black-and-white twisted pairs you see are the input connections; the rest of the wires are power supply connections.

I mounted the power supply components directly to the chassis, using single-point terminals and capacitor clamps for the big filter caps. If you can't find appropriate capacitor clamps, you can use a tie wrap that has a screw-mounting hole to hold the can caps to the underside of the chassis.

Photo 5 shows a closer view of the power supply section. The rectifier diodes are hiding under the ceramic caps.

Figure 6 shows the wiring layout. Note that some wiring is left off this drawing

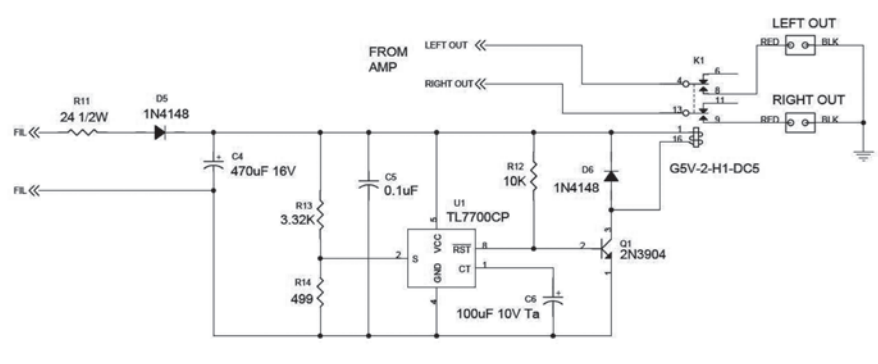

FIGURE 5: Delay circuit.

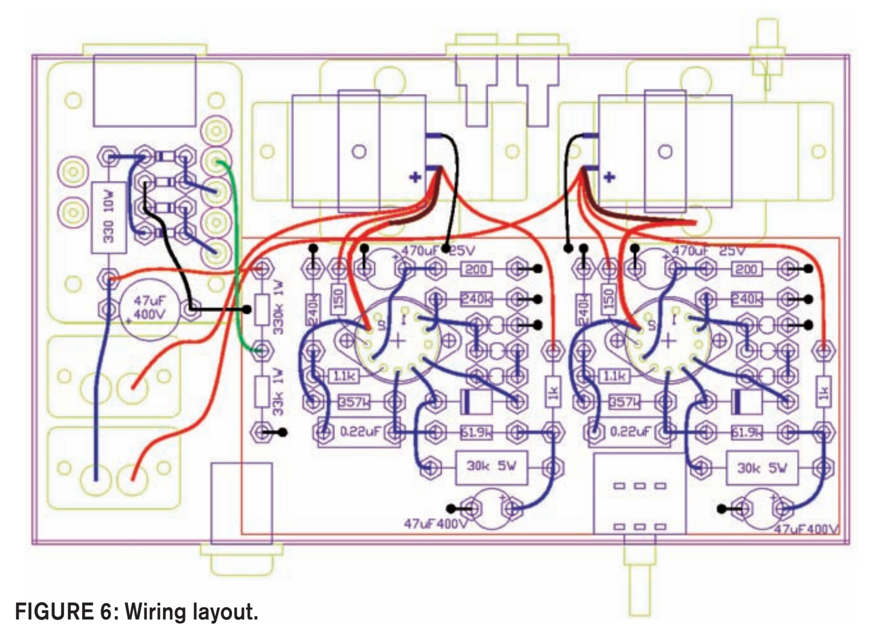

FIGURE 6: Wiring layout.

PHOTO 2: Back view.

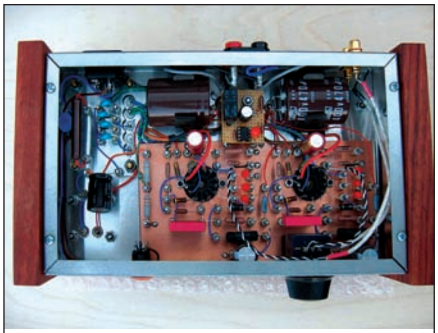

PHOTO 3: Inside the chassis.

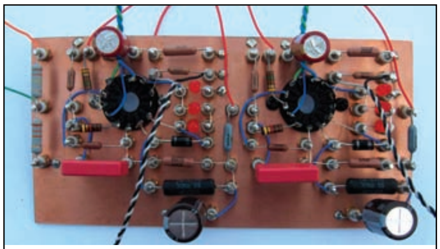

PHOTO 4: Closeup of the terminal board.

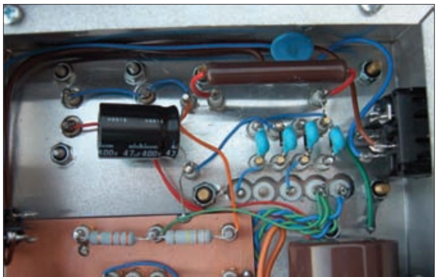

PHOTO 5: Power supply circuitry.

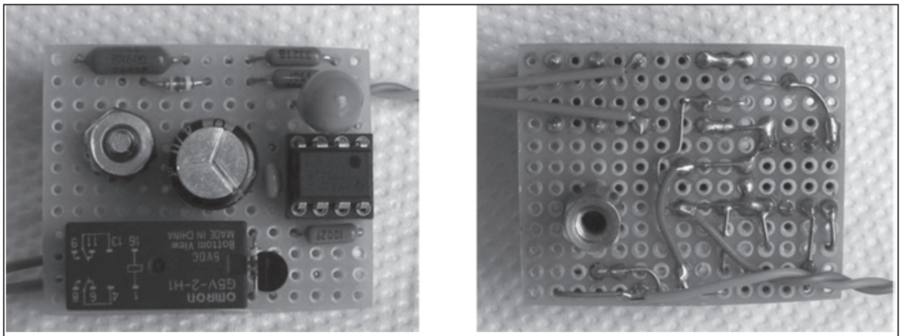

PHOTO 6: The delay circuit.

for readability; the filament, input, output, and AC wiring are not shown.

I assembled the delay circuit on a small piece of perfboard. Luckily (since it wasn't originally part of the design), I found a place to put it—I mounted it to a standoff that was attached by one of the output transformer mounting screws. You can see it in **Photo 3**, located near the output jacks; **Photo 6** shows a detailed view of the assembly. The orange and green wires connect to the 6.3V filament circuit; the gray wires connect to the output terminals. The OPT secondary wires are soldered to the relay next to the gray wires.

MEASUREMENTS

One of my goals in setting up the amplifier was to have a reasonable amount of distortion at 1W output, and, I hoped as much power as possible at the 5% distortion level. I also wanted to make sure that the harmonic content was dominated by the 2nd harmonic, at least at lower powers. **Figure 7**, an FFT of the output at 1W (into 8Ω), shows the result.

You can see from the FFT that the highest harmonic is the 2nd, with the rest following at progressively lower levels. THD+N is about 0.7%. You can also see some "hum intermodulation" on the fundamental (1kHz) and even harmonics—these are the spurs that are ±60Hz relative to the signal. I don't know whether these are audible, as they are 80dB down. They appear to be induced from the AC heaters, since the B+ has very little ripple, and they also appeared when running from a DC bench supply.

The non-harmonic signals you see that are ~90dB or more down (for example, at 4600Hz) are measurement artifacts generated by various pieces of equipment on my bench.

I tried several different 6T10 tubes to make sure things weren't going to vary too much between tubes. I found that in this circuit there was a definite difference between GE 6T10s and Sylvanias—the Sylvanias had less THD, which dropped as the amp warmed up. The GEs had more second-order distortion, and their THD didn't change with time. Though I've only listened with the Sylvanias so far, I'd guess the GEs would sound a little softer, and a bit "tubier."

Maximum output power (at 5% THD) is about 3.5W into 8Ω. Impressive for an output tube whose plate is dissipating around 10W! The distortion products increase smoothly as you raise the output power, with the 3rd harmonic exceeding the second at around 2.5Ws. The frequency response (**Fig. 8**) is good, showing 1.5dB down at about 18Hz and 20kHz.

Another important characteristic of an amp is the damping factor, or output impedance. I measured the output impedance of the amp to be 3.5Ω. Not extremely low, to be sure, but not bad for a pentode amp that doesn't use global feedback.

LISTENING IMPRESSIONS

When I first set this amp up to listen, I connected it to a pair of full-range speakers in MLTQWP cabinets that use the Jordan JX92S driver. It sounded OK, but to be honest I wasn't impressed. The sound was OK, but dull and sort of lifeless. In retrospect, I shouldn't have been surprised—the Jordans really prefer to be driven with a low impedance, and really need more like 10W of power.

Next, I connected a pair of bass-reflex speakers that use Fostex FE164 drivers (you can see them in "An Affordable Full-Range Speaker," *audioXpress*, 6/03). I already knew from earlier experiments that these speakers really sounded best when driven with a higher impedance, so I had high hopes that this amp would make them sing. I was not disappointed this time—starting from the first track I got that goofy smile I get when I know I put together something good.

The amp was quiet, with no trace of hum or buzz audible even with my ear next to the 92dB efficient speaker. There was some broadband white noise though, a bit of a hiss. I couldn't hear it at my normal listening position a couple of meters from the speakers.

The sound of this amp is not like any single-ended triode amp I've heard. It has a certain edginess, not the "liquid" sound everybody talks about triodes having. It sounds *live*. I suppose that this is partly because this amp does have a fair amount of odd harmonic distortion, compared to most SET

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amps. But I suspect there's more to it than that.

I never enjoyed listening to blues on the FE164 speakers, and I was fairly sure I couldn't reproduce listenable blues with 3W. But I was very surprised that this combination was fabulous with blues—there was bass coming out of the Fostexes that I'd never heard before, and it felt like I was getting a live audience with the King (B.B. King, that is) in my listening room.

With such good results from the FE164s, just for fun I thought I'd see what would happen with the amp connected to some *really* inexpensive speakers; bass reflex again, using Radio Shack's old 40-1197s (more or less a version of the Fostex FE103). They lacked low bass (what can you expect from a 3″paper driver in a tiny box?), but they really sounded nice—better than they had with any other amp I'd tried with them. Vocals, in particular, were lovely.

As you've probably heard before, building a good-sounding audio system is all about synergy. The best amp with one set of speakers might not sound good with another. With this amp, it looks as though I hit upon a magic recipe to make inexpensive full-rangers sound good! aX

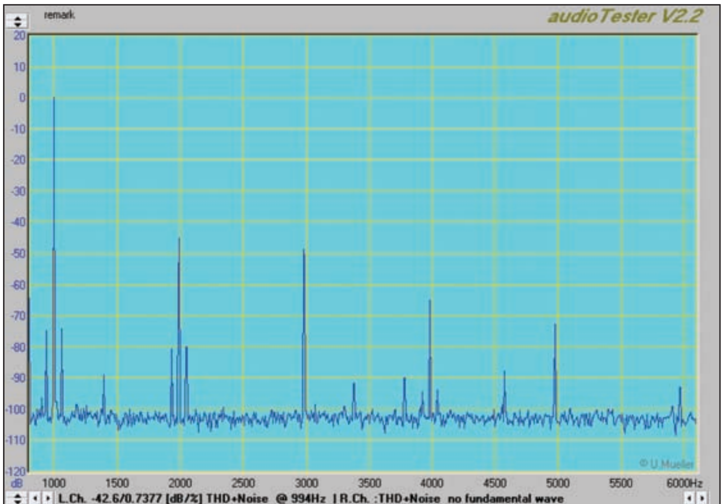

FIGURE 7: Output FFT.

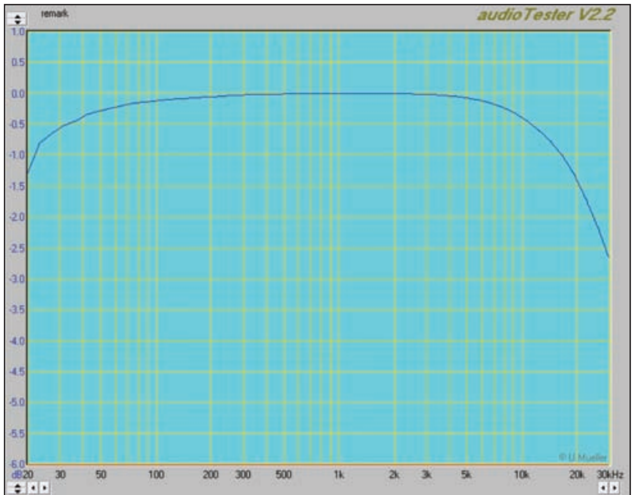

FIGURE 8: Frequency response.

SOME SUGGESTIONS FOR BUILDERS

This isn't too difficult a project to build, but I have a few suggestions that might save potential builders some pain and anguish.

Don't try to squeeze this amplifier into as small a chassis as I did, unless you have a fair amount of experience building point-to-point projects and a lot of patience. It really is difficult to work in tight quarters like this—I bear a scar from a very nasty soldering iron burn to prove it. I'd suggest at least an 8″ × 12″ × 2″ chassis. Even in a larger chassis, you can still use the same terminal board design.

A larger chassis will also give you a little more flexibility with choosing transformers and chokes. Finding ones that match what I used could be a little difficult. I'm guessing most folks won't want to fly to Japan to pick up a couple of $20 output transformers.

Luckily, there are domestic options. Both Hammond and Edcor Electronics have small single-ended output transformers that should work well in this amp. In particular, the Edcor XSE-10-8-8K is almost exactly the same size as the Japanese transformers I used, and is *very* reasonably priced. You want a transformer with between 5kΩ and 8kΩ primary impedance, good for at least 40mA of DC current and 5W of power.

The power transformer I used, the Hammond 261G6, seems to work fine, but I'm pushing its ratings a bit. It does run hot—so it may be better to use something a little bigger. You need around 220-250V AC, and 100mA DC (a rating of at least 180mA AC RMS would be good). Filaments are 6.3V at just under 2A.

An inexpensive option that should work well would be to use a 120V to 240V isolation transformer (one rated for 40-50W should do, such as the Triad N-68X), plus a 6.3V 2A filament transformer. Note that you can compensate (to some extent) for a different transformer voltage by changing the resistor in the filter (R15 in **Fig. 4**). You want to shoot for around 300V B+.

The TL7700 IC used in the delay circuit can be hard to find in a DIP package. I found some at Allied Electronics (www.alliedelec.com), which also stocks triad chokes and transformers. If you can't find a DIP IC, you can use an SOIC and mount it using an SOIC-to-DIP adapter board (available from the usual sources such as Digi-Key and Mouser).

You can download schematics and full-size mechanical drawings, including the layout of the terminal board, from my website www.pmillett.com. Happy building! ■

Valve (Tube) Regulated **Power Supplies**

Choose one of these tried 'n' true circuits to meet your power supply needs.

By Terry Bicknell

How many times have you looked for a special power supply that was better than the ordinary transformer-rectifier-capacitor types and found that they just didn't cut it, or were too expensive on smoothing capacitors? Often you take second best and get on with it. Well, here are some tested systems that don't cost an arm and leg and are easily "tailored" to your needs.

WHAT IS A REGULATED POWER SUPPLY?

A regulated power supply (psu) is one that has its output controlled by a system of feedback involving a controlled element, a reference, and a DC amplifier. It may have either fixed or variable output.

For this article, I consider only the more common uses, that of providing an accurate and stable source of DC supply with a very low impedance, low noise, and low ripple. This is one of the most common uses in audio and instrumentation.

There are three basic types: series regulator, parallel regulator, and a combination of these two. There is also a switch-mode or switcher type, but due to the noise and rf output, this is not used much in valve-related audio, as far as I can tell.

WHY USE A REGULATOR?

A good question! A carefully designed standard psu with pi input filtering can give a very good level of ripple rejection—in fact, an output for light loads, in millivolts! However, a regulator needs an output voltage that is constant over all the current range, and a good filter cannot do that. It also has a very low output impedance, which makes it ideal for critical jobs such as instrumentation and dual audio amplifiers. It also should have a very low output noise.

Tables A and **B** show typical ripple and noise with a pi input filter psu and capacitor input filter. These are basic type circuits in valve equipment and are used for comparison here. AC mains input is set at 235V AC throughout both tests. In **Table A** the ripple output is excellent at less than 2mV; however, the regulation is less than desirable for a varying load. Note that an ordinary capacitor input filter will produce still far less desirable results, although its output voltage will hold up longer. In **Table B** the basic output voltage regulation of the capacitor input filter is better than the pi type, but the pi has far better ripple rejection.

HINTS AND TIPS

I have built and tested the following series of circuits for this article. Each circuit has a typical set of performance data and some application hints. By the end of the article you should be able to choose a type suited to your immediate need and have some fun building it, too.

The first thing to remember about a regulated psu, or simply "regulator," is that it is very wasteful of energy—not at all green! Because of the high voltages across the series tube (or valve), it requires a separate well-insulated heater supply. Also, the amplifier valves will probably need a separate heater supply. The power transformer is becoming complicated with all those LT or LV windings.

You'll also need to look at how you supply the high voltage DC. Will you use the efficient silicon rectifiers, or yet another valve—a large power-gobbling rectifier with yet another separate heater winding? I examine these choices in more detail later.

The next thing to consider is the fact that the series valve requires about 90 to 120V across it to provide you with the regulation range. This means the HV supply will need to meet this, and as a rule of thumb for the transformer HV winding, take the maximum desired output voltage and add 125V—more on that later, too.

The next item of importance is the reference circuit. Do you use a zener diode, which is probably the easiest, or a reference tube such as an OA5 or 6561? I normally use zeners, but their

Power and Safety Warning!
Please note that there are lethal voltages in these circuits. Take every precaution to protect yourself when building and testing. In particular, discharge all electrolytics. Neither the author nor the magazine can accept liability for accidents.

TABLE A - RIPPLE FROM A PI INPUT FILTER

Va (RMS)	V(DC)	I(DC)	Ripple (RMS)
333	454	0	less than 2mV
330	436	10	Lightly loaded
329	422	20	>2mV
327	405	30	
324	385	40	
323	372	50	> 5mV

Pi filter components: 500µF + 30H choke + 500µF
% Regulation = 22%

TABLE B RIPPLE FROM A CAPACITOR INPUT FILTER POWER SUPPLY

Va(RMS)	V(DC)	I(DC)	Ripple (RMS)	
331	450	0	5mV	Greater than for a pi type even
331	445	10	80	when lightly loaded.
329	436	20	150	
327	426	30	200	
325	416	40	240	
323	412	50	300	single 500µF electrolytic

% Regulation = 9.2%

use requires some thought. Most HV zeners above 50V have quite a low power dissipation and require careful consideration from the operating current point of view. Depending from where you take the reference supply, you will need to look at the chosen diode's operating current limits and work well within.

In general, placing the reference circuit in the output of the regulator means that it is being fed from a "regulated" output—included in the feedback loop—and should therefore be more stable and provide a more accurate reference. However, you need to allow for the current drawn, especially if you make a small low power regulator such as the single tube one here. You can also feed the reference from the unregulated DC supply feeding the anode (plate) of the series tube. This has the advantage that the reference current is not part of the output current, and it may allow a wider range of control for simpler circuits.

No work on regulated power supplies is complete without some comments on wiring and current paths. Most of you will not be building a double insulated system, so you'll need to provide a solid frame earth, which needs to be separate from the common ground return and fixed on a separate chassis bolt. You can also solder the electrostatic screen of the power transformer to that point.

Use heavy gauge wire for the power wiring from the rectifiers to the main electrolytic(s) because there is a high circulating current charging the electrolytic in this region. Common ground return paths need to be of a similar gauge, and the output ground needs to be wired directly to the smoothing electrolytic, with no other connections.

It is unwise to have long leads from rectifiers to the smoothing electrolytic, especially if they go via a PCB, because charging current, circulating currents, and common impedance can affect performance. Keep power circulating currents as far away from the main wiring as is practical to avoid stability problems.

In practice, I usually place the output divider (sensing) across the output terminals. This avoids any common impedance from the PCB or direct wiring. The reason is that the DC reference is not the electrolytic nor DC power input, but the output loaded terminals.

I next look at the basic raw DC part of the regulator, its general requirements, and its rectifiers. You'll need a power transformer with at least three low voltage windings for each of series valve, control valve, and perhaps your external circuitry or a separate low voltage transformer. If you choose thermionic rectifiers, you'll need to have another isolated winding.

RECTIFIER CHOICE

You can choose between a valve or semiconductor diode. Diodes are far more efficient, easier to wire, and neater. However, the input electrolytic needs to sustain the maximum peak voltage immediately after switching on, until the regulator draws current, whereas the valve warms up with the rest of the circuit, which can be less stressful on the smoothing or input electrolytic. The downside to a valve rectifier is the voltage drop across it, which can easily be more than 25V compared to the semiconductor diode's. You may need to calculate this into the

DC supply to the regulator. It effectively increases the impedance of the source and may limit action.

As with both types of rectifier, you need some form of current limiting resistor to help protect the rectifiers and electrolytic from peak charging current. This is usually 33 to 150Ω per side; however, it depends upon other factors such as the transformer DC winding resistance and value of the smoothing electrolytic, which always needs to be specified generously and have a low ESR for best performance.

You'd be correct in thinking that perhaps 50Ω of series resistance will degrade the performance of raw DC input. It does, but as with all compromises, the trade-off is in the protection and life of the diodes and the electrolytic as well as limiting that high charging current. The following approximation relates ripple and charging current.

The specification for the transformer needs to be generous, rather than tight, with 5% regulation being better than 10%. This means that the DC input will be more constant with load; therefore, the regulator has less work to do effectively, and thus performance improves.

Calculation of DC regulation factor

$Vr = (V_{nl} - V_{fl}) / V_{fl} \times 100$ (%) ... Ideally 5% or better

where (nl) = no load voltage and (fl) = full load voltage.

A simplified calculation for the series protection resistor

$Rt = Ra + Rsec + \{(Np/Ns)^2\} \times Rp$ ... HT supply resistance

where Ra = resistor in each anode leg

Rsec = resistance of half secondary

$\{(Np/Ns)^2\}$ turns ratio squared

Rp = primary resistance.

Note that the term Ra implies that there is a physical resistor in the anode of each diode leg, the value of which needs to be between 33% and 50% of the total value of the HT supply resistance above, to ensure adequate protection. Generally, I use 33% for tubes and 50% for semiconductor diodes.

Note on ESR: ESR is "effective series resistance." It is a measure of how good the electrolytic is. Low practical ESR values of milliohms are common and needed for a good psu. See maker's spec-sheets. It's not good to use "old type" electrolytics in power-supply units.

To relate the high charging current of the electrolytic to the DC current drawn by the load, check out this approximation between DC current drawn, % ripple voltage, and diode peak current. This peak current is the charging current that flows for less than 1ms at 50Hz, and it can cause some problems with noise.

1% ripple relates to approximately 12 times Idc, charging current, and
5% ripple relates to approximately 5 times Idc, charging current, and
10% ripple relates to approximately 4 times Idc, charging current (where Idc is load I).

There's a critical trade-off in the specification of the power transformer: the quality of the electrolytic (low ESR) and charging current and, hence, regulator performance. The following will give some idea of the trade-offs.

Calculation of supply ripple factor

%Rip = V(RMS)/V(DC out) × 100 (%)
where
V(RMS) = AC ripple output and
V(DC) = DC voltage

DC Voltage Regulation (Vr)

$\%Vr = (V_{nl} - V_{fl})/v_{fl}$ where
V(nl) = no load voltage and V(fl) = full load voltage

The DC regulation at the input is im-

TUBE (VALVE) CHOICES

There are many tubes to choose from in the power category, ranging from the specialized 6080 and 6AR7 to the basic EL84 and 6AQ5. The 6080 and 6AR7 are specially designed triodes with good heater cathode insulation and large cathode areas. Workable alternatives to these are the KT66 and KT88 or EL34 (6CA7) as used here. All operate in triode mode.

The triode gives better performance than a tetrode or pentode in the series pass position, because it has a lower anode resistance; however, it requires more drive. For small or specialized regulators, 6AQ5 or ECL86 as used here is quite effective.

Control valves are pentodes for the best performance in simple circuits because of their anode (plate) characteristics. However, they need a lot of care around the selection of Vg2. I have found that a good start for this is at 70% of Vout maximum. I used the EF86 in this article; however, you can also use the 6AU6 and others such as the EF91 and EF94 and even an EF80 (6BX6). In the most specialized circuits ECC83 (12AX7) is used for control DC amplifiers or reference isolation and sensing.

OUTPUT/Series Pass	Controllers/DC Amps	Reference Tubes
EL34 (6CA7)	6AU6	OA2 150V
KT66 (6L6)	6AK5/5654	OA3 75V
KT88 (6550A)	EF80 (6BX6)	OC3 105V
ECL85 (6GV8)	EF86 (6267)	OD3 150V
ECL86 (6GW8)	EF94 (6AU6)	5651 92V
6V6	12AX7/5751	Not used here
6L6	ECC82 (12AT7)	
807	ECC83 (12AX7)	
6080 (6AS7)		
6098 (6AR6)		

Zeners - Philips series	Zeners - 1N series
3W 5% BZT03 C75/A52R	5W IN5378 B 75V "B" series are 5%
75V for circuit 2 and	types and recommended for use here.
3W 5% BTZ03 C100	5W 1N5378 B 100V
100V for circuit 4	5W 1N 5383 B 150V
3W 5% BZT series	
75V and 100V for circuit 3	

portant to maintain the performance of the regulator. Although a slightly larger transformer may be the outcome, it's worthwhile.

THE TESTED CIRCUITS

1. Simple two valve system (series pass and control valve)
2. Three valve circuit—DC amplifier plus isolated zener reference
3. Three valve circuit employing a triode pentode for control
4. Simple low-cost ECL86 regulator
5. A high performance, general-purpose bench unit

For many "one-off" jobs, a simple regulator that can deliver 50mA is all you need. The collection of circuits here offers at least that and, in the case of the last circuit, greatly improves on all desired parameters. These provide enough current to test most small amplifiers and signal line equipment, but not enough to run a power amplifier. Also in the case of the simple one valve circuit, you can build a cost-effective and very compact regulator that gives good performance for its simplicity.

The design of the power transformer will not be shown, because normally this is a specialist job. However, your junk box might provide a solution. You'll need an HT winding capable of approximately twice the desired output current to get the best regulation.

GENERAL OPERATION

For a given output voltage, the output is maintained "constant," or regulated, by means of a basic comparison between the output terminals as set by RV1 and the reference voltage, as set by the zener. Changes due to loading are "sensed" by the output divider networks, of which RV1 is a part, and that change is passed on to the DC amplifier V2. V2 then produces a corresponding but inverted change as drive to the series pass or regulator tube. This change in drive then corrects the effects of the load on the output.

For the circuit in **Fig. 2**, the only real difference is the fact that the zener reference is "isolated" by a cathode-follower circuit. This reduces the loading on the zener, and, because it is in the output circuit, it will be very accurate.

Figure 5 is a special case involving coupled triodes in a very high gain DC amplifier and reference system. V3 (right-most) is similar to other circuits in that there is a DC amplifier connected to the sensing control and a reference connected to the opposite grid of the pair. The difference voltage between the two anodes is then fed to the two middle triodes to produce the drive to the series pass tube and hence the correction needed.

Figure 1 and **Table 1** show a design example for a 50mA regulator with an adjustable output and a range of 100V.

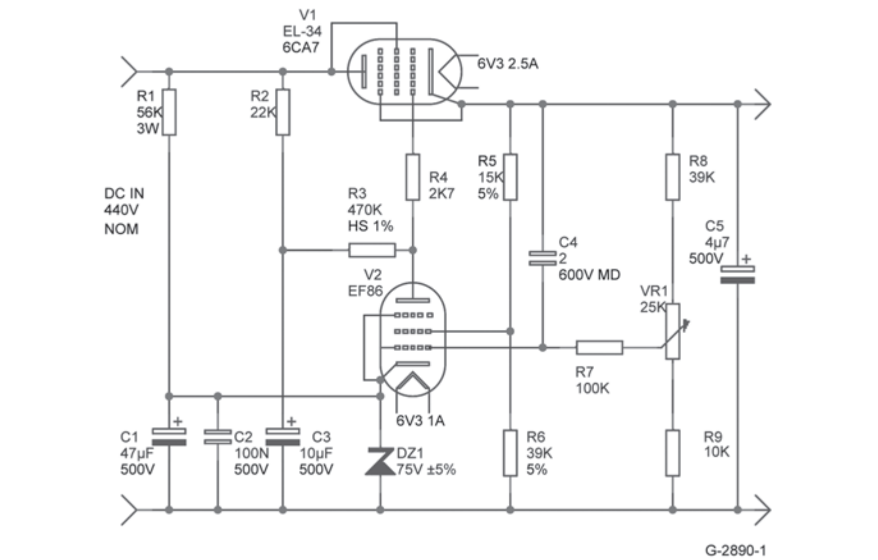

FIGURE 1: Circuit 1. Basic regulator. EL34 (6CA7) and EF86 (6267) plus zener diode.

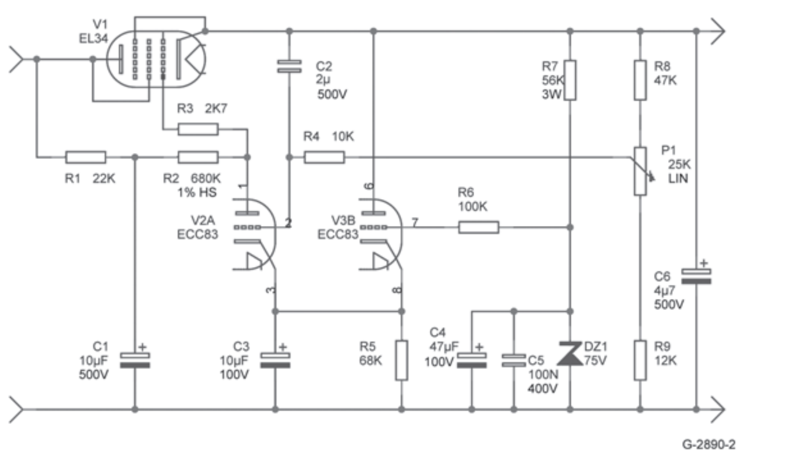

FIGURE 2: Circuit 2. Isolated zener. EL34 and 12AX7 (ECC83).

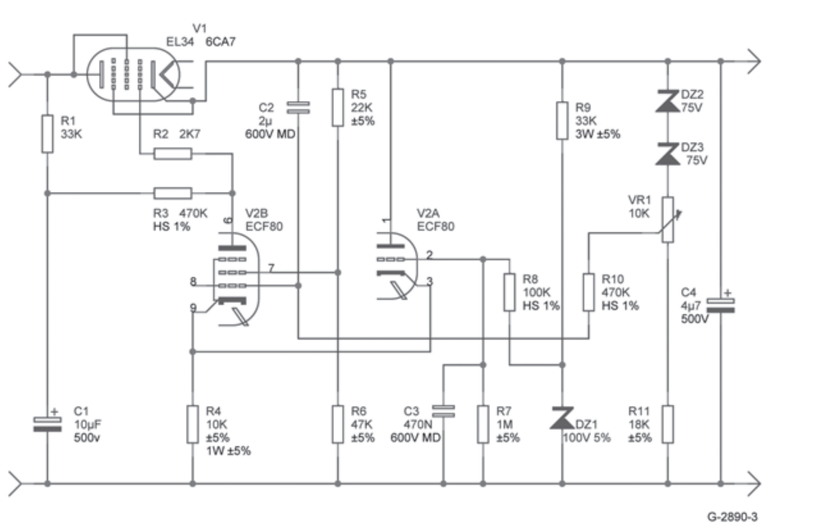

FIGURE 3: Circuit 3. EL34 (6CA7) and ECF80 (6BL8) general-purpose type.

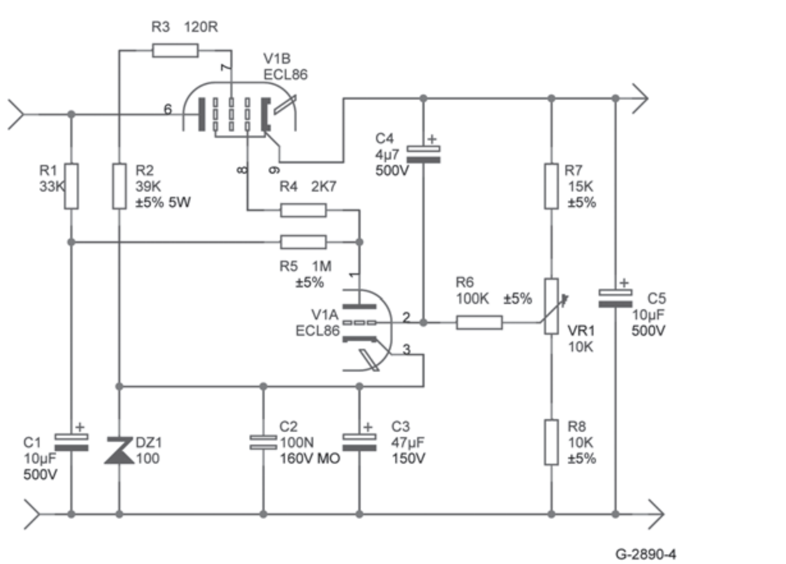

FIGURE 4: Circuit 4. Simple one tube type. ECL86 (6GW8).

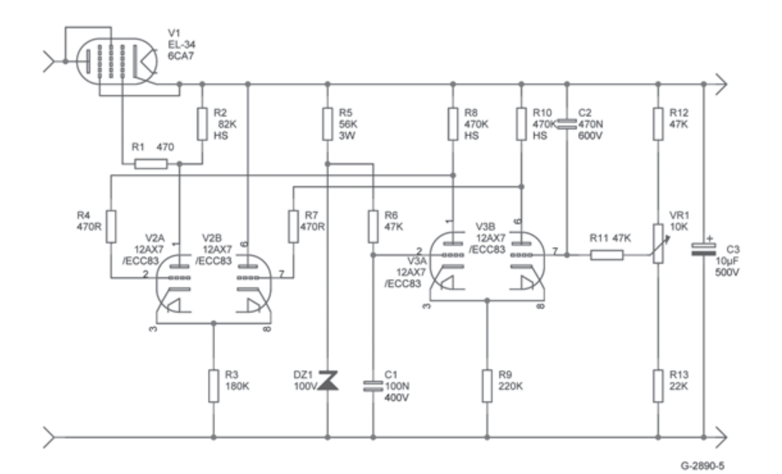

FIGURE 5: Circuit 5. High-performance unit.

TABLE 1
EL34(6CA7) AND EF86 BASIC REGULATOR

I = 0	180V	230V	280V	Hum and noise
25mA	180V	230V	279.8V	3mV RMS,
30	179.8	229.8	279.7	unless
35	179.9	229.8	279.7	otherwise
40	179.8	229.8	279.7	stated
45	179.8	229.7	279.5 (1)	in
50	179.8	229.7	279.5 (1)	note (1).
55	179.8	229.7	279.4 (1,2)	
60	179.7	229.7	279.4 (1,2)	
65	179.7	229.6	279.3 (1,2)	
70	179.7	229.6	N/A	
75	179.6 (1)	229.6 (1)	N/A	
80	179.5 (1,2)	229.6 (1.2)	N/A	

Note 1: Output hum and noise approached 5mV.
Note 2: Output hum and noise approached 5mV and regulation decreased to more than 0.5V.
Line regulation for ±5% and 10% input based on 235V AC and 280V DC at 40mA.
-10% = 279.8V, -5% = 279.8, 0% = 279.7, +5% = 280.1, and +10% = 280.3V.
Drift: This is tolerable for such a simple system—approximately +0.5V with settling of a similar order during first hour.
Output impedance approximately 5Ω midrange.

These might include a general-purpose basic regulator for which no provision for line regulation is required. Line regulation for the simple circuits is minimal as indicated, so, too, is the drift figure. The useful attributes are simplicity, ease of construction, reliability, with moderate performance and easy-to-obtain parts.

The transformer (in all cases here) is required to have 3 × 6.3V windings plus the HT winding, plus an electrostatic screen between primary and secondaries. The basic specifications of these windings would be 6.3 at 1A for the amplifier, 6.3 at 2.5A for the series valve, and a suitable 6.3 for the externals. The HV winding can be a single one or a tapped type depending on your choice of rectifier. This one is center-tapped for use with silicon diodes.

The circuit includes an old 80mA center-tapped transformer I found in my junk box that met the winding spec but was a little light on current at 80mA. It just made 5% Vr at 50mA and was degrading after that. I chose solid-state diodes with a 50Ω protection resistor. You can also use a single winding solution and a bridge rectifier, which, due to its two diode voltage drop, is not as efficient as the center-tapped version chosen, but is easier on transformer specs.

The sample circuits assume the output is centered on 250V.

NOTE: For **Figs**. **1**, **2**, **3**, and **5**, the EL34 is used in triode mode. **Figure 4** uses only an ECL86. The EL34 is not the best choice, but it is freely available and robust enough for this article.

Example:

1. Input smoothed DC measured as 440V.
2. If regulation is 5% then ideal lower limit at 50mA will be approximately 420V.
3. Zener current for a 3W type 30mA maximum diode will operate safely at 5-7mA plus valve current.
4. Maximum valve current (Ia + Ivg2) —800µA.
5. Set series zener R from maximum Vin. 440V and 7mA = 62K, and at 420V I = 6.7mA.

5a. Power rating of 62K is

2.8W—use 4 or 5W type.

6. Set value of potential divider chain at output for 3.75mA = 74K.
7. Set value of screen supply to 70% of Vout. Preferred values of 15K and 39K. 5mA bleed.
8. Determine a value for variable R from 74K total of potential divider (sensing) chain.

Valve is near to zener voltage, V2 saturation at approximately 98V and cut off at about 93V on g1. This is valve data.

8a. Value of resistance (R lower plus RV1) for 280Vout 26.67K, and for

TABLE 2 Three valve circuit EL34 plus ECC83

I = 0	200V (200.5)	250V (250.6)	300V (300.1)	
10	200.4	250.3	300	Noise and hum
20	200.4	250.3	299.9	initially
25	200.2	250.2	299.5	less than 1mV.
30	200.2	250.2	299.3 (1)	
35	200.2	250.2	299.2 (1)	
40	200.1	250.2	299.2 (1)	
45	200.1	250.1	299.1 (1,2)	Noise and
50	200.0	250.1	299.0 (1,2)	hum 5mv.

Note 1: Hum and noise less than 5mV, > 1mV.
Note 2: Regulation considered worse case when > 0.5V at maximum load.
Output impedance: 10Ω midrange.
Drift: Not determined. Settling < 0.5V during first hour.
Line regulation: Not accurately determined, < ±0.8V approximately.

USES

This is another general-use circuit having low-running current with an isolated zener reference. Its range can be changed easily to meet other needs. It has fairly good noise performance. It is adaptable, easy to construct, and reliable.

TABLE 3 ECF80 (6AB8) and EL34(6CA7)

I = 0mA	255	280	300 (299.9)	Hum and noise
25	255.0	280	299.9	less than
30	255.0	280	299.9	1mV
35	255.0	279.9	299.9	up to
40	255.0	279.9	299.9	50mA load,
45	254.9	279.8	299.9	increasing to
50	254.9	279.8	299.9	5mV at maximum
55	254.9	279.8	299.9	output.
60	254.8	279.8	299.8	
65	254.8	279.7	299.7	
70	254.8	279.6	299.7 (1)	
75	254.7 (1)	279.6 (1)	299.6 (1)	
80mA	254.6 (1)	279.5 (1,2)		

Note 1: Hum and Noise initially less than 1mV; at high loads approximately 5mV RMS.
Note 2: Regulation was very good until the top end, where it approached 0.5V.

Line regulation: at 300V and 40mA =	-10%	-5%	0%	+5%	+10%
	299.6	299.7	300	300	300.1

Output impedance: Approximately 4Ω midrange (2 to 8Ω).
Drift: Not determined but settling was > 0.3 and < 1V.

USES

Circuit 3 could be used to supply a permanent load at relatively high current. There was a minimum of drift during the first hour and it has relatively good line regulation. Hum and noise output was very good over the whole range. This is a reliable above average simple design that can be modified easily to give a greater current output with different tube and transformer.

top resistor 39K, RV1 = 25K and lower resistor = 10K.

9. For 280V output V1 will need a grid voltage of the order of 275. This sets the anode voltage of V2, which will be close to Va maximum - cutoff.
10. Minimum Va will be close to 210V, which correlates to minimum Vout of 180.

Does it work? Yes, as the results in **Tables 1-5** attest. The accompanying numbered **Tables** (**1-5**) correspond to the circuits. *aX*

TABLE 4 Single valve ECL86 (6BG8)

I = 0	200 (199.8)V	250.0 V	300 V	
10	199.8	249.9	299.9	Hum and Noise
20	199.7	249.8	299.8	increased
25	199.7	249.8	299.7	above
30	199.6	249.8	299.6	40mA.
35	199.6	249.7	299.6	
40	199.3 (1)	249.7 (1)	299.5 (1)	Regulation
45	199.2 (1)	249.6 (1)	299.2 (1,2)	>0.5V(2)
50	199.1 (1)	249.6 (1)	298.9 (1,2)	>0.5V (2)
H & N	< 5 mV	< 5mV	> 5 mV (2)	

Note 1: Hum and noise less than 5mV but increasing with load.
Note 2: Regulation decreasing, 0.5V or > 0.5V at maximum load.
Note 3: Max cathode current of output section is 65mA.
Output impedance midrange: Approximately 5Ω at 40mA.
Line regulation: Not determined for this circuit.
Drift: Settling during first hour, approximately 0.5V thereafter, < 0.3V.

USES

The uses for this very basic system are generally fixed voltage regulators—a bit like a PCB card regulator—a sub system regulator, or an exceptionally compact regulator for small spaces. The circuit is adaptable and economical in transformer design. This is very reliable and easy to set up and has surprisingly good performance for its simplicity. Many other tubes can be used as long as heater cathode insulation requirements are met.

TABLE 5 EL34(6CA7) + 2 12AX7 (ECC83)

I = 0	250 (250.1) V	300.0V	350 (350.1)V	Hum and noise
20	250.1 <0.1V	300 <0.1V	350.1 <0.1V	< 2mV,
25	250.1			except for
30	250.1			350V range
35	250.1			above 50mA.
40	250.1			Circuit became
45	250.1			critical
50	250.1 <0.1V	300 <0.1V	350.1 <0.1V	due to
55	250.1		350 (1,2)	falling DC input
60	250.1		(1,2,3)	and
65	250.1			voltage
70	250.1		Out	across
75	250.1		of	regulator.
80	250.1 <0.1V	300 <0.1V	Range	
85	250.1			
90	250.1			
95	250.1			
100mA	250.1 <0.1V	300 <0.1V		
	<2mV	<2mv		

Note 1: Hum and noise < 2mV except on high end, see note 2, 3 > 5mV.
Note 2: Regulation decreases rapidly after 350V at 50mA due to DC input.
Note 3: DC input to regulator falling beyond lower limit for this circuit (approximately 415V).
Output impedance: 1Ω or less up to 350V.
Line regulation: +10% +5% 0% -5% -10%
I = 50mA 300 300 300 300 300
Drift: < 0.5V settling in first hour. Thereafter less than 0.2V overall.

USES

This high-performance circuit could be used for a variable supply (with better DC input) or a fixed regulator of high quality and low noise and hum output.

Build a Hybrid **SE OTL Amp**

Here's a different OTL amp design that delivers plenty o' power to the speaker.

By AJ van Doorn

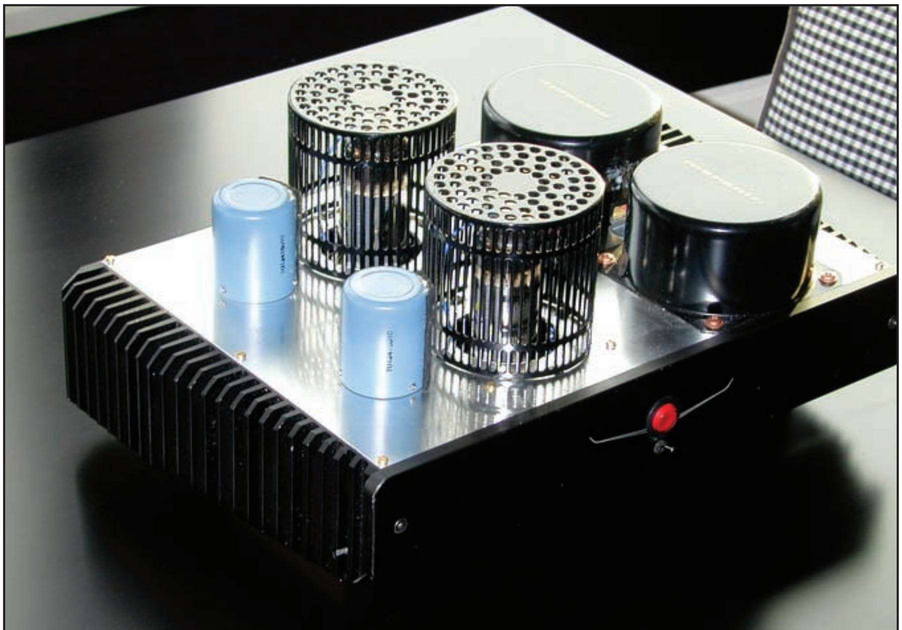

PHOTO 1: The OTL PL509-based amp.

To have a proper load impedance for output power tubes, you need an output transformer, regardless of what configuration you use—SE or PP, triode, ultralinear, or pentode mode. At the end of the golden valve age, new low voltage valves have been developed to get rid of output transformers, such as the EL86(6CW5) connected in SRPP mode, delivering 8W to a 500Ω speaker.

POWER TO THE SPEAKER

As you see in *aX* articles, there are several suppliers of good OTL machines. The main problem is the low load seen by the output stage of the amplifier. Even a 16Ω loudspeaker needs an output current of 1A to deliver 16W, which is rather hard to get from vacuum tubes.

It is, however, quite feasible to increase this load impedance in such a way that the tube also delivers a moderate current, with sufficient power going to the loudspeaker. You must monitor the current of the tube and feed this signal to an amplifier that delivers all the extra power to the speaker. This amplifier can be a transistor or tube power amplifier, which should act as a multiplier and make an exact copy of the input signal over a wide frequency range; no phase shift can be allowed, otherwise, information will be lost, especially in the bass and treble.

The first time I used this principle was in the Audio Aero Transtrac amplifier—a stereo PP triode amp using EL34s. The tubes deliver only 2×10W, and, after multiplying the current with a power op amp, 40W was delivered to the speakers. The basic schematic is shown in **Fig. 1**. The OPT has a primary impedance of 3200Ω, the secondary is 8Ω. The secondary current of the OPT is monitored by a 470mΩ non-inductive resistor (Rs) and activates the inverting input of a power op amp (TDA7294), delivering the extra power to the loudspeaker.

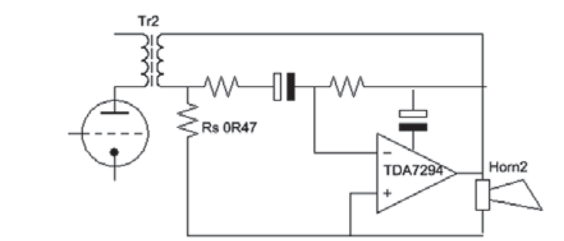

FIGURE 1: Transtrac amp circuit.

I started by monitoring the output voltage of the OPT. There was no chance to get it going, due to LO and HI phase errors, but measuring the secondary current instead did the trick. As the op amp increases the secondary output voltage of the OPT, the plate load increases as well—to about 10kΩ (instead of 3200Ω)—and the tubes are in class A mode all the time, hence the outgoing current has very low distortion, too. The rather small OPT has a large bandwidth and the amp is one of the best of its kind.

At the CES show many people congratulated me for the sound this amp produced. However, this Transtrac isn't a real OTL. I tried several ways to make an OTL, even with the room-heating 6C33 tubes. I decided to look for a system similar to the Transtrac (**Fig. 2**).

CIRCUIT OPERATION

On the right is the loudspeaker connected to the TDA7294 power op amp from Thompson. I've used this chip in many different applications and consider it one of the best of its kind and very musical indeed. The non-inverting input is grounded by R8, and as long as there is no signal on the inverting input the voltage to the speaker terminals is zero. On the left is the tube amp.

With the cathode connected to -100V DC, I used a small input transformer to isolate the input circuit from it. I had on hand a rather good Marantz model with a step-up ratio of 1:2. I loaded the secondary with a 10k resistor. I don't fancy open load transformers.

The input signal modulates the tube via the stop-resistor R2 connected to grid 1 of the tube, a PL509 that can deliver plenty of current. In this configuration the plate current is 200mA DC, going to the speakers via R6, which function to monitor the outcoming plate current. As long as only DC is flowing, R6 delivers 6V DC to OP1, which is a differential line receiver, model INA137.

It has a gain of 1/2, so OP1 outputs 3V DC, which is blocked by capacitor C4 so at zero signal the output of the power op amp TDA7294 is zero. But where is the 200mA from the tube going?

This current produces an offset current in the output stage transistors of the TDA7294 but its internal auto-bias control system keeps the offset voltage below 1mV DC on the speaker terminals, which not only means that the output impedance of the TDA is very low, but also that the positive output power transistor collects the full 200mA DC, so the chip works in class A as well. Maybe this is one of the reasons that the amp has such a wonderful open sound.

It is important to know that as the INA137 line receiver is fed by ±15V DC it will accept input signals up to ±42 Vp before clipping occurs. With an output

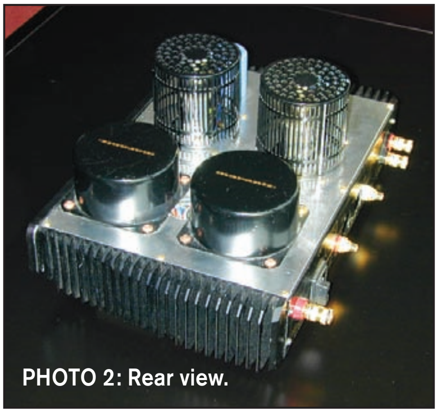
PHOTO 2: Rear view.

of 50W to the 8Ω speaker, the TDA7294 outputs ±28 Vp, which is about the maximum possible with the ±32V DC power supply used for the TDA7294.

The gain of the TDA7294 is 30×, so for full output the signal on its inverting input must be about ±1Vp. However, the INA137 has a gain of ½, so the signal on R6 must be ±2Vp corresponding to a variaton in current of almost ±70mA peak, hence at maximum output, the plate current of the PL509 varies between 130 and 270mA. If you prefer a larger plate current modulation, just lower the value of R6 to 22 or 18Ω. This will, of course, reduce the sensitivity of the amplifier. The tube, with a plate current of 200mA DC, dissipates only 20W. I measured a small loss in the treble—attributable to neither the transformer Tr1 nor the TDA7294. That's why capacitor C2 is in parallel with the bias resistor R1; it slightly increases the gain of the tube for

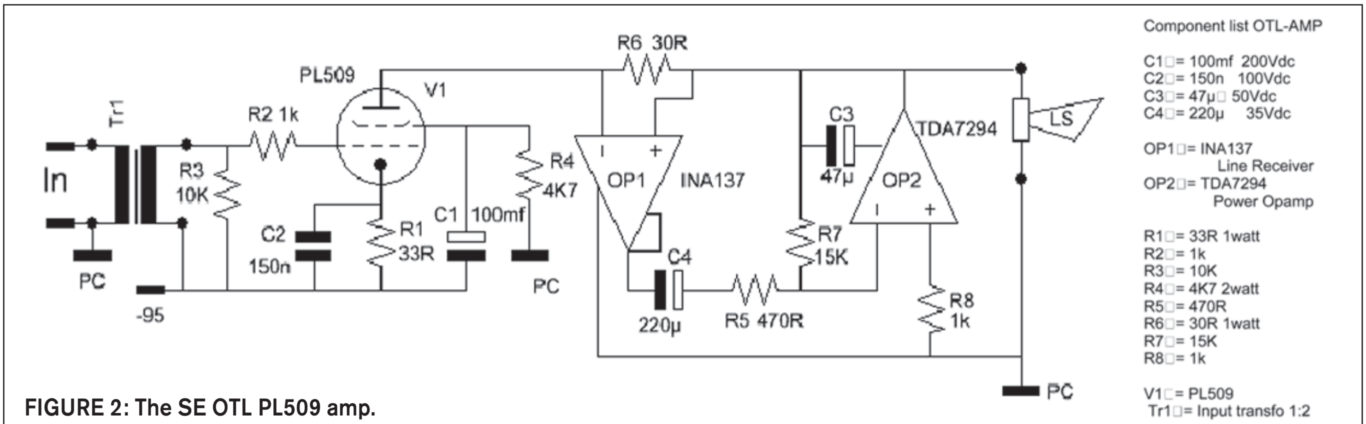

FIGURE 2: The SE OTL PL509 amp.

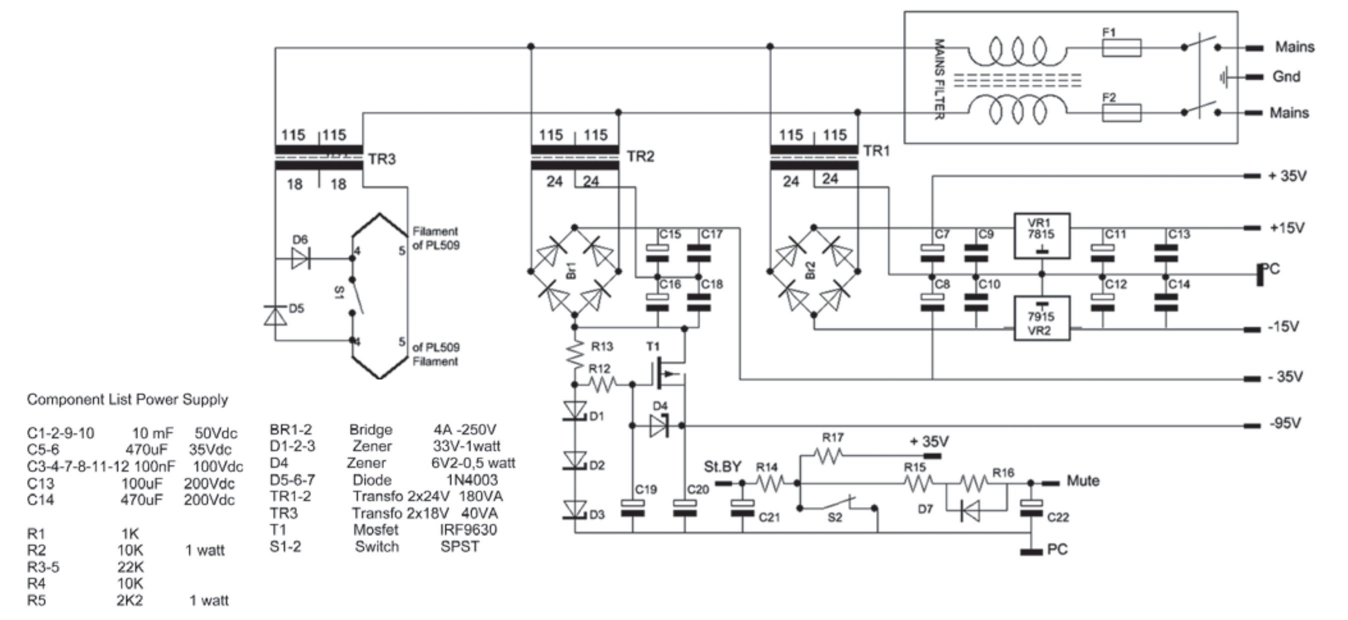

FIGURE 3: PL509 power supply.

the treble region.

POWER SUPPLY

SE amplifiers, especially, need a proper power supply (**Fig. 3**).

Luckily, I had most of the parts on hand from old Marantz monoblocks.

One transformer provides the ±32V DC for the TDAs and ±15V DC by adding two simple regulators. The ±15V DC feeds the INA137, but you must be sure that its pin1 (ref) is connected straight to the input circuit of the corresponding TDA, otherwise hum and noise may appear.

The PL509 should operate at about -100V DC at 200mA. Please note that this PL500 power supply is the same as for the TDA but is connected to the negative rail of the TDA power supply so both the positive as well as the negative rail of supply deliver the 400mA for both tubes.

You could use a separate -100V DC supply for the tubes. But then only the positive rail of the TDA supply would deliver the 400mA DC for the tube plates and the unbalance would produce a higher ripple on the TDA. Also, the ripple was rather low, so I added the MOSFET regulator for an absolute quiet -100V DC supply.

A third transformer is used for the filaments of the PL519s. When starting, each filament gets only the halfway-rectified voltage, and when closing switch1, the full AC voltage will be supplied; the loss in the diodes is negligible in this case. The TDA7294 has a standby and muting circuit activated by switch 2. For details just look at the TDA specs. If you like, you can combine S1 and S2 into one single DTDP model.

Figure 4 is just the same amplifier but using PL500s instead. I didn't make this amplifier myself, but it may be handy to know that PL500s are easy and cheap to get. For further information refer to the following websites.

www.st.com (type TDA7294)—TDA7292
www.ub-audio-shop.de/downloads/a80kurzen.pdf—PC boards for TDA circuit
www.ub-audio-shop.de/downloads/nt-kekurzen.pdf—PC boards for PSY circuit
www.tubedata.org—PL509 or PL500
http://ti.com (type ina137)—INA137 or INA2137
amplimo.nl—Transformers

RESUME

I use this amplifier at home with a pair of 98dB LYRA speakers from WLM, Austria. You can find out all about these fabulous speakers on the web globeaudiomkt.com. Years ago, I developed all the tube gear for Audio Aero in Toulouse, France, and now I'm making all kinds of very special so-called tracking amplifiers, of which I'll report at a later date.

Several friends (with golden ears) came to listen to this amp and they all agreed that this OTL was one of the best they had ever heard. At the VAD HI-FI show we replaced some P-P and SE tube amps with this little OTL and this one was a clear winner, so make one! aX

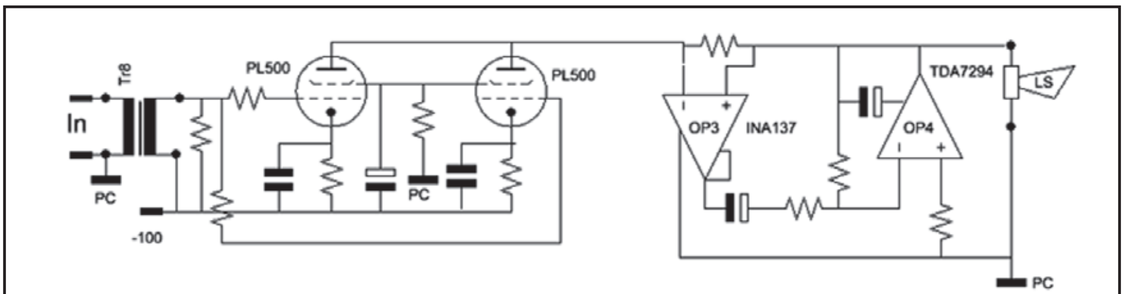

FIGURE 4: The SE OTL PL500 amp.


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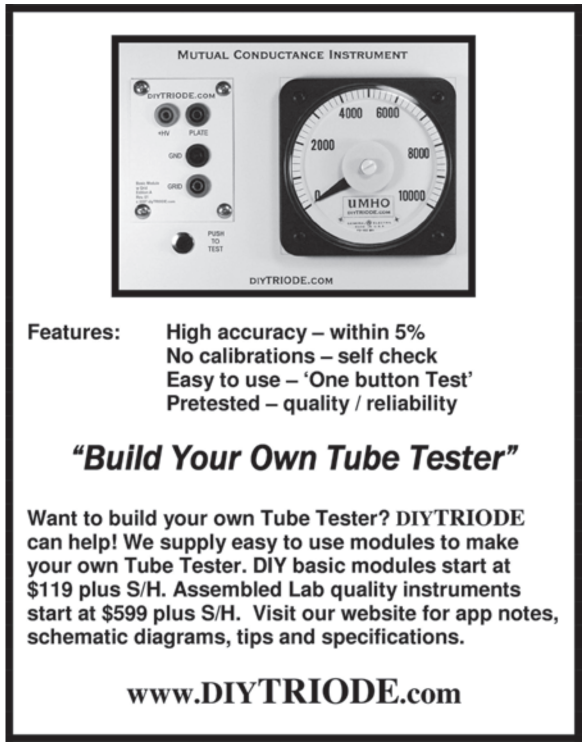
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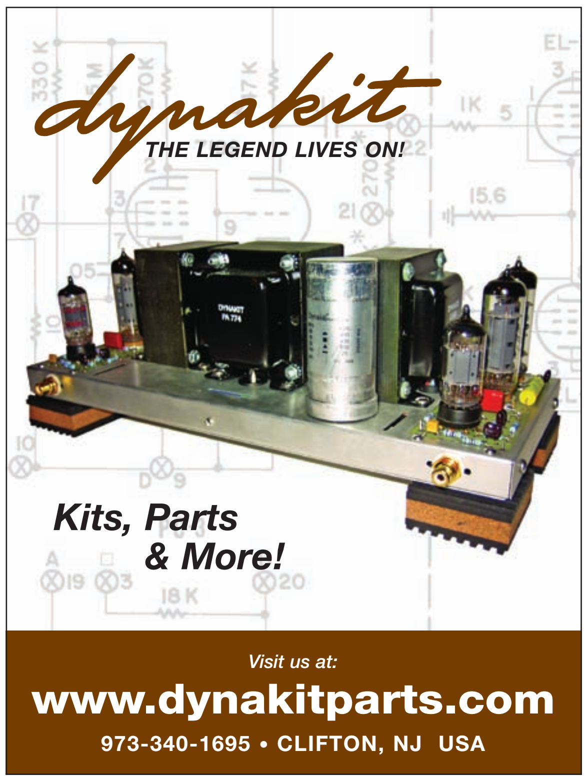
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The **Sidewoofer** Line Array

This small line array includes a woofer in a ported enclosure.

Speaker building has undergone a number of fads since its inception in the 1950s. First, there were the bass reflex and horn-loaded enclosures that caused the saws to ring. Next came the sealed enclosure with its low compliance woofers, and then the reflex re-emerged through the application of better science by Thiele and Small, sparking further amateur construction. The 80s and 90s saw interest in the transmission line, and the rebirth of the horn challenged the weavers of MDF. The new millennium, and a paper by James Griffin, Ph.D., brought line arrays out of the public address arena to the forefront of home audio, and their requirement for a large number of drivers has made the raw speaker dealers happy.

I have built all—or variations—of the above over the years and have settled on the line array as my current favorite. I like its low distortion, excellent imaging, and slim profile. In order to achieve their fine imaging performance, line arrays require you to use small drivers of 5″ or less. Consequently, unless you perform special equalization of the input signal, they lack those last few octaves of bass that all speaker builders seek. You can acquire those missing octaves by the addition of a subwoofer, or the application of aggressive equalization, but in my case, I did not want a bulky box intruding into my limited space listening area, nor the complication of a dedicated equalizer.

My solution is the "Sidewoofer Line" (**Photo 1**), which combines some of the attributes of the line array along with a vented woofer in a narrow slab enclosure, 22.5″ deep, 37.5″ high, and 8.75″ wide. My bride said they resembled tombstones, while another friend suggested they could serve as large bookends. Their observations aside, I will stick with the Sidewoofer Line name for now.

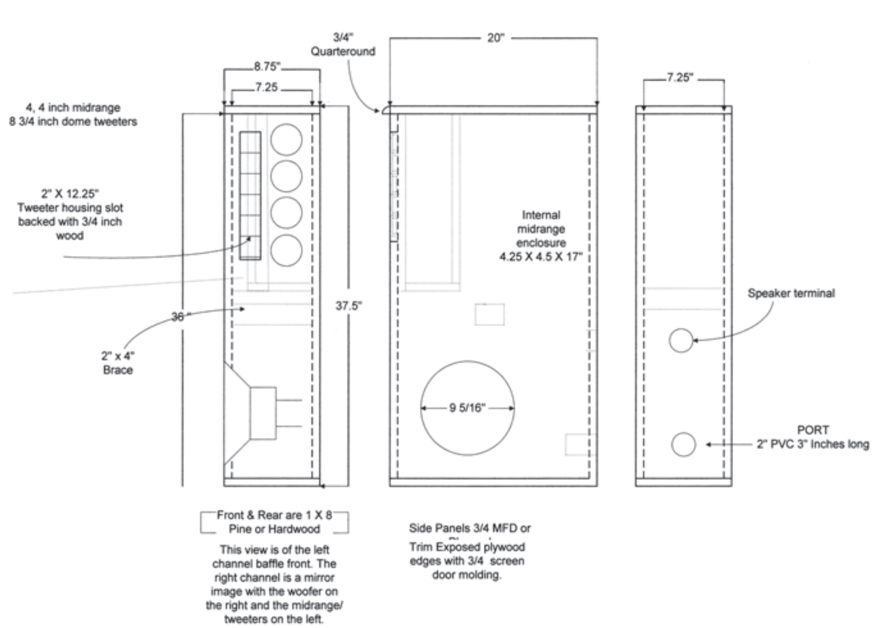

FIGURE 1: Sidewoofer Line schematic.

THE SIDEWOOFER LINE DRIVERS

A number of recent articles in *audioXpress* have described line arrays (*aX* Nov. and Dec. 2005, and Jan. 2008). Two of the published systems built by Bill Fitzmaurice utilized a ¾″ dome tweeter, with a 6k to 20k range and a 4″ "full-range" driver. One of the systems, built by Roger Russell, used 20 drivers with no tweeter and a dedicated equalizer. Russell has since launched his own company, IDS, to sell a commercial version of his speaker, with equalizer, for $18,000. As someone who has never spent over $300 for any pair of speakers, I wish him well.

PHOTO 1: Finished enclosure less grille.

The tweeters and midrange drivers used in my system are surplus items offered in limited quantities by Parts Express. The dome tweeters sold for 25 cents, and the 4″ "full-range" speakers were 49 cents each. Like many *aX* readers, I stocked up on these low-cost jewels. Unfortunately, most are no longer available, but any driver with specifications similar to the following should work as well as or better than the originals.

Tweeter: ¾″ dome power 20W, z = 4 or 8Ω, Fs 4200, Fr 6k to 20k, SPL 87

Midrange: power 5W, z = 8Ω, Fs 105, Fr 75-15,000, SPL 86

The tweeters could also be 2″ cone types. (In the 1950s, *Bozak* featured an array of vertical cone tweeters that provided fine imaging.) Just make sure the SPL closely matches the SPL of the midrange driver.

You can find suitable surplus speakers at All Electronics, Digi-Key, and Parts Express. I have found that the Parts Express customer service people go out of their way to help. Many of their technical staff read *aX*, and they know what is in stock and what would meet your project requirements.

For the woofer, I chose the fine performing Dayton Model DC 250-8 10″, which has a specified F3 of 33Hz in a 1.9ft^3 vented enclosure. The Sidewoofer Line is about 2.5ft^3 when the internal brace and midrange enclosure are subtracted, giving a F3 of 29Hz with a 3″ long 2″ PVC pipe port. The crossover, in keeping with my preference for electrical simplicity, is a three-way first-order design using components I already had in my collection. I used the LDP computer programs developed by David Weems, for both the enclosure alignment and the crossover design.

THE ENCLOSURE

The enclosure is constructed of a combination of ¾″ plywood and dimensional lumber. The sides are 20″ × 36″ ¾″ plywood, but you can also use MDF. My local lumberyard cut a 4′ × 8′ sheet of ¾″ plywood into four 20″ wide slabs, and a neighbor made the finished cuts with a tablesaw. This resulted in a complete "kit" of pre-sawn lumber ready to assemble.

The front and rear panels are constructed of ordinary clear pine 1 × 8 boards. The top and bottom are ¾″ pieces of plywood with edge treatments. OK, I know, conventional wisdom says you never use such low-density lumber in a speaker cabinet, yet I have been doing it for years. It reduces the need for wood cutting and is cost-effective.

You can also use dimensional hardwoods for the top, bottom, front, and rear panels. The caveat is that you must use them only in narrow panel configurations. In this instance, the front panel is braced by a combination of a backboard for the tweeter assembly and the enclosure for the midrange drivers. If the pine scares you, make the entire cabinet of plywood or MDF. During assembly, I used clamps and liberal amounts of Elmer's carpenter glue with 2″ finished nails spaced 4″ apart. I predrilled for the nails, which I countersunk, and filled the depression with stainable wood putty.

CONSTRUCTION

Start by cutting out the two front panels, keeping in mind you are making "mirror" images of one another. The left panel has the tweeters on the left of the midrange units, and on the right panel they are to the right of the midranges (**Photo 2**). The tweeter mounting required an unorthodox solution, so you should prepare for their mounting prior to cutting out the midrange driver holes. You cannot mount them in the usual fashion due to the configuration of the 2″ square mounting plate designed to mount into thin metal or plastic automobile panels.

Instead, surface-mount them into a 2.25″ × 12.25″ slot cut in the front panel (**Photo 3**). The slot is backed up with a piece of ¾″ pine or plywood into an "L" shape because it will also serve as the platform for the internal midrange enclosure. The tweeters are 5/8″ deep, and fit nicely in the slot, becoming almost flush with the front panel. They are held in place with 1″ drywall screws with the wiring laid in the ¼″ space next to the drivers. I thought about gluing them in place but their mass, and magnetic opposition when placed side-by-side, did not make gluing an easy option. The tweeter feed wires fit into the channel, and a 3/16″ hole leads the feed wire into the cabinet.

The close positioning puts the driv-

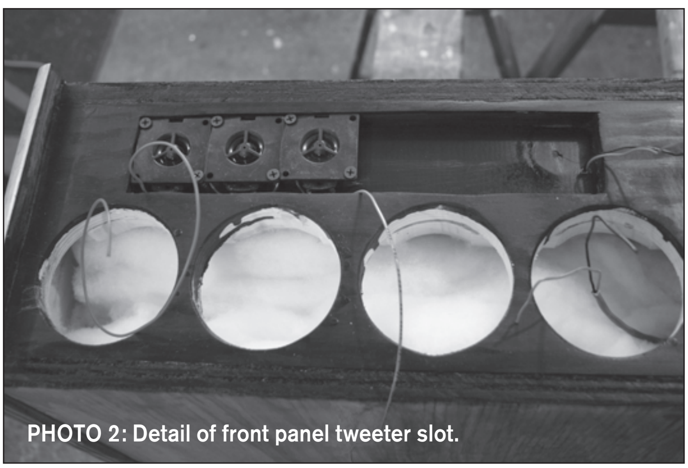
PHOTO 2: Detail of front panel tweeter slot.

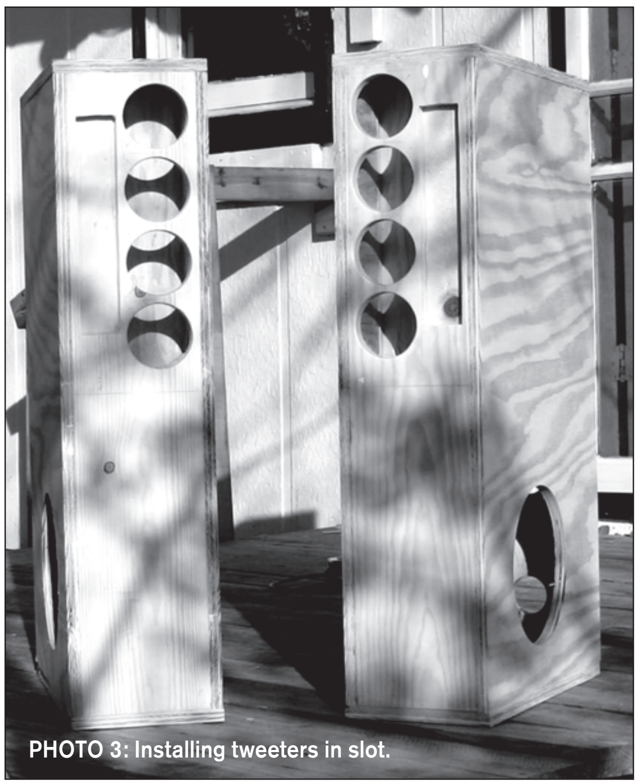
PHOTO 3: Installing tweeters in slot.

er centers as close as possible to foster proper line array coupling. Be careful when cutting out the slot, because you will be only ½″ from the edge of the panel. Use a good-quality jig jaw blade and go slowly. Once the back plate is installed flush with the front panel, the integrity of the panel will be restored.

Using a Jasper Circle Jig, I cut out the midrange holes, after making the tweeter-mounting slot. The speaker I used required a 3 11/16 hole spaced 4″ center-to-center. This spacing places the square driver frames right up against each other, providing proper acoustic coupling of the drivers. The tweeters have four mounting holes, but two 1″ drywall screws are more than sufficient to hold them in the panel slot.

The midrange drivers have their own isolated enclosure to prevent woofer back wave modulation. This internal baffle is made of 1 × 4 pine using the tweeter slot back plate as a platform. Partially construct the bottom, and one side on the rear of the front panel. The top and other side are formed by the walls of the main enclosure.

Install the panel that forms the wall of the midrange enclosure first, caulking all the joints. Then attach the back of the midrange cabinet and drill a 3/16″ hole for the feed wire. Stuff the midrange enclosure with slightly teased polyester fill material (Dacron). The fs of the drivers was raised to an acceptable 150Hz by the baffle. The woofer hole is 9 15/16″ and is mounted slightly off-center toward the front of the enclosure to ensure clearance for the 3″ long reflex port and to lessen standing waves in the enclosure.

The rear panel of the main enclosure is cut out 2 7/8″ for a round, five-way speaker terminal and a hole for the 3″ long 2″ PVC pipe reflex vent. The vented enclosure provides a f3 of 33Hz, and blocking the port raised the f3 to only 45Hz, which still sounded fine in listening tests.

The side panels are joined to the front and rear panels using clamps, Elmer's carpenter glue, and 2″ finishing nails countersunk with a punch and filled with wood putty. The cabinet is internally braced by a piece of 2″ × 2″ in the center. When installing the top, put a liberal layer of caulk on the internal baffle pieces that are flush with the top. I nailed the top set nail heads with a punch, filling the holes with wood putty (**Photo 4**).

The bottom is applied last, which allows final caulking of the enclosure joints, and then you can caulk the bottom through the woofer hole. Mount the crossover on the bottom panel behind the woofer. I lined the main enclosure on all sides with 1″ of polyester quilt padding stapled to the walls, and trimmed

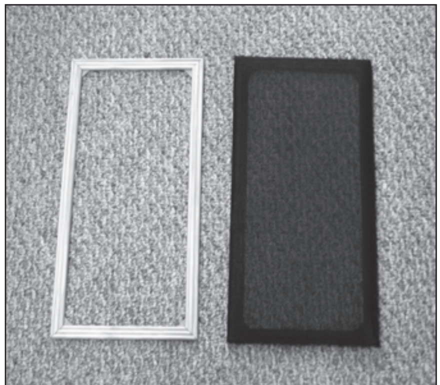

PHOTO 4: Raw enclosures showing mirror image front panels.

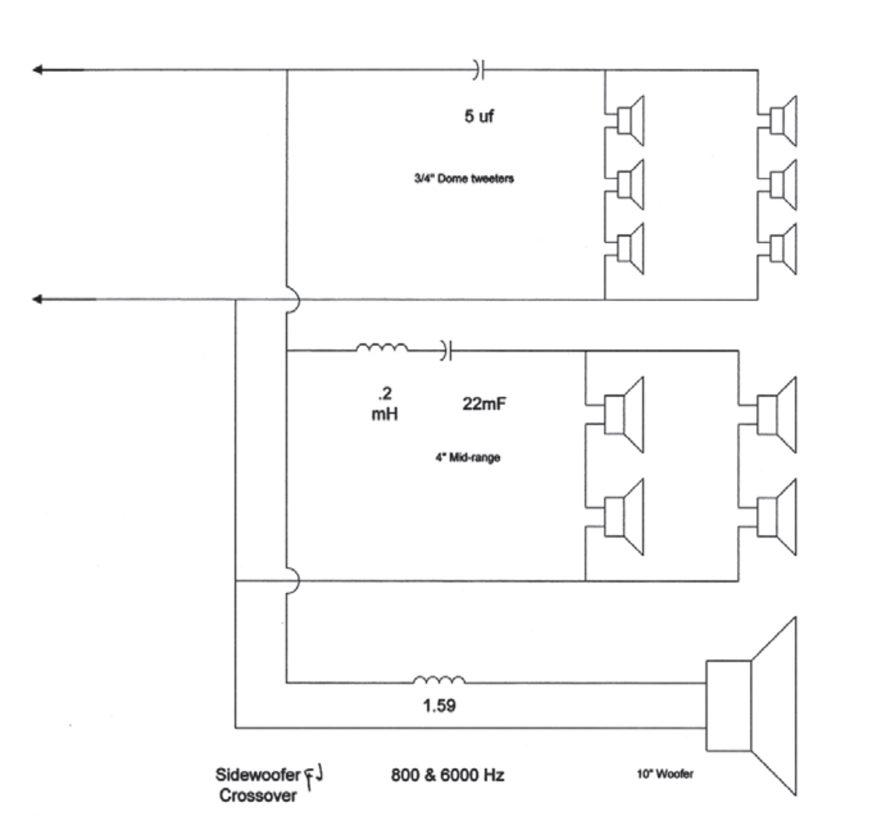

FIGURE 2: Crossover.

the plywood edges of the top and front panel with screen door molding. A piece of 3.4″ quarter round serves as the front panel overhang, under which the grilles are positioned.

FINISHING

You can stain the exposed wood as desired. I painted the back and upper baffle front flat black and fashioned a grille to cover the drivers. I covered the exposed plywood edges with milled ¾″ screen door molding, and the base molding is some scrap door casing I had in the wood box.

The grille frame for the front panel is made of milled ¾″ screen door molding with 45° mitered joints (**Photo 5**). Each joint is reinforced by a corner gusset made from four-ply illustration board. This makes a lightweight frame that, when attached with Velcro, does not resonate with the audio. I first glued together all the frame joints using picture frame clamps, then glued and clamped the corner gussets and I painted the frame flat black and stretched and stapled the grille material over the grille

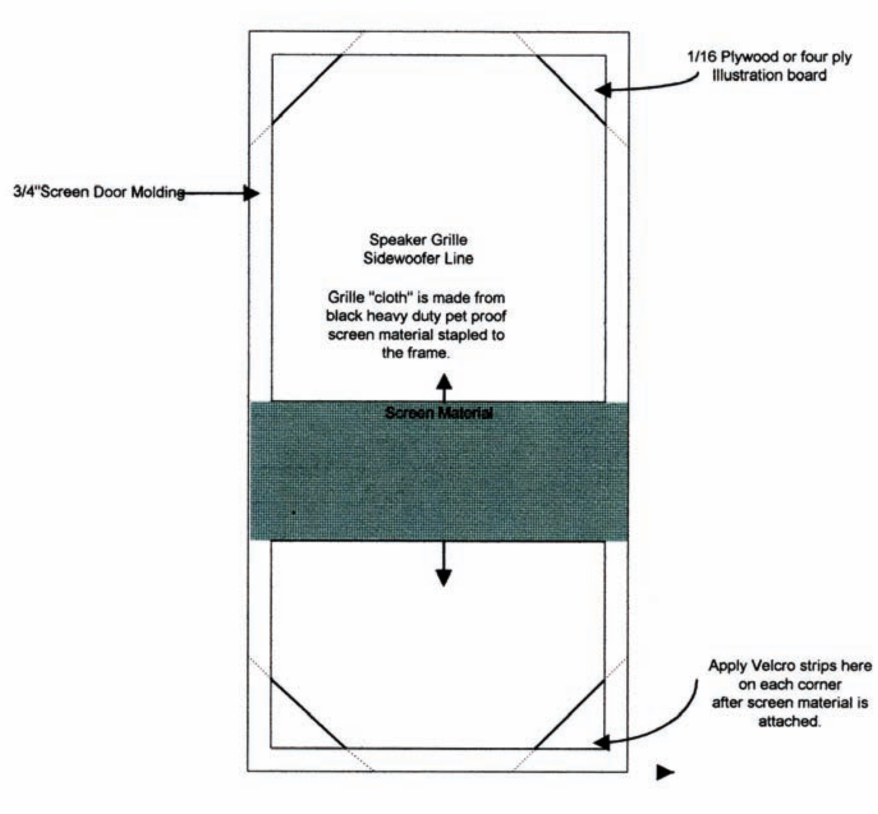

FIGURE 3: Screen material.

frame. I used a reinforced screen door material that is designed to be pet-claw-proof.

This black plastic coated nylon is open to all audio frequencies, while being rugged enough to protect cone drivers from accidents. It also allows just a hint of the drivers to show through for those who want to see as well as to hear their audio creations. The frame is attached to the enclosure with matching Velcro "sticky back" strips stapled on the corners of the frame and mating enclosure surface.

For the woofer, you can purchase ready-made round frames from Parts Express, part number 260-426. Or, as I did, lay the woofer face down on the pet screen material and cut along the edge with a carton knife. Once the woofer is mounted, just glue the grille to the speaker gasket using clear drying "tacky glue" from your local craft store. There is plenty of clearance for the woofers' extreme excursions, and the woofer is protected by the screen material.

WIRING THE SIDEWOOFER LINE

I chose to make the low crossover point

800Hz and crossed the tweeters at 6kHz. The 4″ midranges perform best in the 200 to 7000Hz range, and, along with the tweeters, they complement the side firing woofer. To make wiring easy, mount the crossover in the bottom behind the woofer (**Photo 6**).

Wire two of the midrange speakers in series and then parallel them to achieve 8Ω. The tweeters were 4Ω, so I wired three in series and then paralleled them resulting in 6Ω impedance. I soldered

PHOTO 5: Front panel speaker grille made from screen door molding and heavy-duty pet screen material.

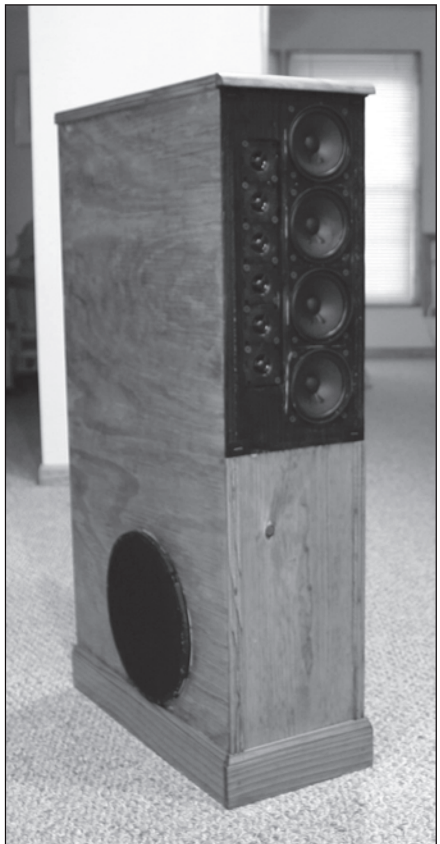

PHOTO 6: Crossover installed in enclosure bottom.

all connections and mounted the drivers with 1″ drywall screws. I applied a layer of silicone caulk to the underside of the frames of the woofer and midrange units. This ensures a good air seal, and the silicone makes the woofer easy to remove if you must change the driver or tweak the crossover.

TESTING

We are fortunate to have a large walk-in closet with a sloping ceiling, which serves as a fairly good anechoic chamber. It is stuffed with my bride's clothes and a hundred pair of her birkenstocks, all of which nicely absorb sound. (Why does any woman need a hundred pairs of shoes? Answer: For the same reason her husband has over 200 raw speakers in his shop.) I used a Radio Shack sound level meter, C weighted, and a B&K Model 3001 portable audio generator fed into a solid-state test amp with no equalization controls. Because of the position of the side-mounted woofers, I made measurements from two different positions and then averaged them into the one response chart (**Fig. 4**).

Subjectively, they reproduce sound with amazing imaging and solid bass. I like classical organ and orchestral music, and, my bride is into heavy metal—both Virgil Fox and Metallica met or exceeded our subjective listening tests. If you want the unique imaging of a line array with some fine low end, this system will meet your requirements. *aX*


PHOTO 7: Finished enclosure with grille.


PHOTO 8: Finished enclosure pair.

REFERENCES

1. Griffin, James, Ph.D., *Design Guidelines For Practical Near Field Line Arrays*. Copies available from Dr. Griffin at jrgriffin@cookeville.com.

2. Russell, Roger, "A Unique Stereo Column System," *audioXpress*, Nov. 2000.

Parts List

(All are Parts Express numbers)
4″ Midrange: 269-568
¾″ dome tweeter: 269-708
10″ Woofer: 295-315999
Speaker terminal: 260-370
10″ woofer grille kit: 260-426 (optional, see text)

Crossover parts

2 each 5µF at 100V
2 each 22µF at 100V
2 each 0.2mH choke
2 each 1.59mH choke

Lumber

4 each 20″ × 36″ ¾″ plywood or MDF side panels
4 each 1″ × 8″ × 36″ clear pine or hardwood front and rear panels
1 each 1″ × 4″ × 4′ pine for midrange baffle
4 each 1″ × 10″ × 18.5″ pine or hardwood top and pieces
2 each 2″ × 4″ × 7¼″ cross brace
2 pieces 2″ ID × 3″ long PVC pipe
16′ ¾″ screen door molding for trip and speaker grilles

Miscellaneous items

24″ of Velcro "stick back" strips, black (craft department at Wal-Mart)
Two yards of 36″ wide pet screen (Home Depot or Lowes)
Box of 1″ drywall screws
10′ #18 wire for internal speaker connections

3. Fitzmaurice, Bill, "The Short Line Array," *audioXpress*, Dec. 2005.
4. Fitzmaurice, Bill, "The CurveArray," *audioXpress*, Jan. 2008.

Surplus Speaker Vendors
Parts Express: parts-express.com
All Electronics: all electronics.com
Digi-Key: digikey.com

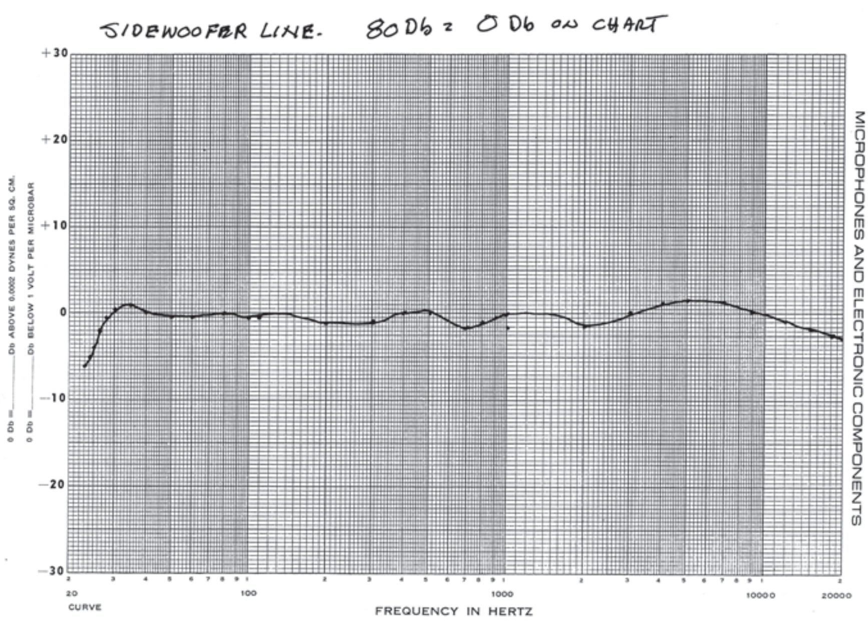

FIGURE 4: Frequency chart.

PHOTO 9: Closeup of tweeter/ midrange array.

CORRECTIONS

I have just noticed a (pretty inconsequential) error in my article ("A Dynamic Microphone Soundcard Amplifier," Feb. '08, p. 22): In the section "Effects of Nonlinearities," item 2 (third paragraph in section):

"A DC offset proportional to the signal level" should read "A DC offset proportional to the square of the signal level."

Paul Loewenstein
paul.loewenstein@gmail.com

There is a small error in eq. 10, p. 10, of my horn theory article ("Horn Theory: An Introduction, Part 1," *aX* 3/08). The square root sign should include the -1 at the end.

Also, the figures that do not have any reference have been drawn or redrawn by me. Bruce Edgar commented that they appeared to be taken from books (Olson, Beranek, etc.) without acknowledgement, but this is not the case.

Bjørn Kolbrek
bjorn.kolbrek@scana.no

MORE ON WALL WARTS

Thanks to Mark Rumreich for mentioning what I forgot—the usefulness of placing transformers outside the box where the audio lives, and for clarifying the voltage doubler question ("Xpress Mail," March '08, p. 49).

Keith Levkoff's suggestion that wall warts have a habit of multiplying is certainly true. I find that the easiest way to pack them efficiently is to use a power strip with the prongs on one side and the round holes on the other, rather than one with the connections centered symmetrically. That way, the wall warts sit at right angles to the strip, and you can fit in many more of them.

If a cathode follower has its cathode sitting at +160V, it's not necessary to float the filament that high. All tube specifications include a maximum heater-to-cathode voltage, often 90V. Floating the filament supply up to +75V (on the ground terminal) would work fine. Most wire is rated to take that.

The efficiency of wall warts, of course, will vary from unit to unit; but the way to keep them from drawing power when the equipment is not in use is to switch off the power strip. Can't get much greener than that.

Finally, thanks to Keith for pointing out the ventilation issue. I don't advocate putting *any* electronic equipment under "carpets or stray clutter," except maybe non-power wires. Wall warts, like everything else electronic, need to stay cool.

Paul Stamler
pstamler@aol.com

RECORDING

I note that "A Matter of Coincidence" is still ongoing, and that it has the benefit of a review by Gary Galo (Jan. '05, p. 44).

I believe the comment on the Opus 1 No. 3 is a little inappropriate though, as the entire trio was done in one day, and neither the mikes nor the musicians were moved. Hmmm! As I recall, the LP of the two trios was record of the month, too, in one of the national magazines of the time.

The recording was done in the Bishopsgate Hall, London, and the piano was the one regularly used by Myra Hess in her day. With the benefit of hindsight, you always think that this or that could be better, but perhaps that is just the evidence of evolution.

Brian Preston
Radstock
County of Somerset
UK

AUDIBILITY OF CABLES

I'm writing in response to Peter M. Moritz's letter, "Cable Tests" (*aX* 12/07, p. 46), which was in response to my letter, "Hearing and Blind Testing" (*aX* 10/07), which made reference to his letter "Cables" (*aX* 07/07). Got that? In my letter, I used his strong words regarding the inaudibility of cables to illustrate a point, perhaps unfairly.

Peter, I sincerely apologize for this. I do appreciate the cable blind testing links you provided in your reply and have read several, so far. The point I was trying to make is that beliefs play a vital and powerful role in how we hear sounds.

Cable blind tests, which change only the given information about which cable is hooked up, just prove that the mind is very open to suggestion. This is true. A person can hear the exact same sound under different circumstances and have an entirely different reaction to it. It's a conditioned response and a function of a person's beliefs, emotional attachment, likes, dislikes, and so on. And, it's all part of our sensory system.

The Jastreboff model for hearing explains this mechanism (www.tinnitus.org). Classical models for hearing are too simple and don't fully explain how we hear sounds. I had to learn the model in order to treat my tinnitus (ringing ears), hyperacusis (hyper-sensitivity to certain sounds), and phonophobia. I am starting work on an article that explains the Jastreboff model and how I think it relates to my audiophile hobby. However, I really don't expect it will end the debate on the audibility of cables—just want to throw some ideas out here.

John Loudenback
johnloudb@earthlink.net

TWEAKS, PERCEPTION, REALITY, AND MARKETING

I was prompted to write this after reading the dialog about Dennis Colin's excellent article, "Intrinsic Fidelity Testing" (Nov. '07, p. 36). I am among those who consider themselves offended by the way pseudo-scientific nonsense has infiltrated the high-end audio marketplace. I would like to try and clarify what I see as the current situation.

First, I am quite convinced that there is such a thing as "intrinsic fidelity." You can measure and compare electronic signals—and can measure whether two signals are similar or different. Assuming that your goal is for a signal to pass through a piece of equipment unaltered (or only altered in a specific way—possibly amplitude), you can measure how well this goal is achieved. The only place for opinion here is that you may personally find that one specific difference is more audible to you than another; you may be more "sensitive" to a non-flat frequency response, while someone else may be more annoyed by THD.

It can be stated, however, that the definition of "high fidelity" is the removal, or at least the reduction, of these differences from the ideal. The dictionary definition of "high fidelity" is along the lines of "accurate to the original." You may even prefer more of a certain type of distortion, but that is *not* "high fidelity." With mechanical devices such as speakers the differences become more pronounced, the approaches to perfect less close, and personal sensitivities more widely variable.

Second, human beings are very much influenced by perceptions.

—We tend to hear what we expect to hear.

—We tend to "become attached to" certain equipment—and choices. Most of us are quite sure that the speakers *we* built sound awfully good. Most of us would also prefer not to admit, even to ourselves, that the expensive equipment we liked in the shop—and cannot return—sounds a bit less impressive at home. Many of these reactions are unconscious—they are built into the human brain and cannot be entirely removed, even with conscious effort—which is why blind testing is so important.

As for tweaks, some actually work (in the factual realm), but many only "work" in the mind of the tweaker. You may be quite sure that your system sounds better with African Wacca-Wacca wood blocks under the speakers, but it would be useful to know whether you're imagining it before you mortgage the house to buy several sets. Reason suggests that you should not be too proud to try the test with your eyes shut and a friend switching blocks.

Personally, I find that music sounds more engaging on bright sunny days, and certain artwork and decor definitely enhance certain moods, which, in turn, make certain music seem more pleasant. This is human nature and is not something of which to be ashamed. I just prefer to know whether I am buying those wood blocks because they have some technical effect, or because I simply find them attractive and they make me happy (especially if they're $500 apiece).

Currently, in the high-end world, there is a tendency to deliberately attempt to blur the lines—you can probably guess that the marketing folks are to blame. If they make a technical argument about how those wood blocks actually do make a difference, and if that argument makes sense to you, odds are that you will hear a difference—whether one factually exists or not. It's simply that your mind hears what it expects to hear. This is why they *tell* you that you'll feel cool driving their sports car—because it actually does influence your perception.

This is also why you will see very few advertisments for tweaks that say simply, "go and listen, then buy them if you like them." The technical arguments serve to set your expectations, which then influence what you hear. This is why you owe it to yourself to do the occasional double-blind test before buying.

Another issue is the "different versus better" question. Many things—interconnects, for example—can make a very tiny but perceptible difference. The problem is our human tendency to assume that every difference must be an improvement—it sounds novel and interesting and it's human nature to consider that it is probably better.

If you try those new interconnects, which make everything sound "clearer and crisper" than it did before, are they really better? Could it be that they are slightly lower in capacitance, so all they are doing is raising the treble a fraction of a dB? If so, not only might they actually not be an improvement, but you might not even notice the change in a few hours after your ears "break in" to the new status-quo.

Again, I might still decide to buy the interconnects if I really prefer the change—but I would want to know whether they are really some wonderful new technological breakthrough or the world's most expensive tone control. Of course, their marketing department and the sales guy at your local high-end shop, who is hoping for a commission on the $5k, will do everything they can to convince you that the change is real and useful.

"Tweaking" as a hobby (as someone described it) has produced a reality where people buy a preamp that's a bit bright, then some wires that are a bit dull to cancel that out, then a new cartridge to "liven up" the midrange. It goes on and on, when it would be much more efficient (and cheaper) to find some equipment that is more or less correct than to persist in seeking tweaks and matches between equipment that cover the flaws of the individual pieces. Back when "tweaking" generally meant buying a cheap piece of equipment, then making economical mods to improve it, this made sense—but now some of us buy expensive equipment, then make expensive mods to try and improve it, which isn't nearly so sensible.

What I find most offensive personally, however, is when the "scientific" explanations for various tweak products are pure and complete nonsense. The current tendency is to offer a scholarly discussion, with a liberal sprinkling of current buzzwords such as "quantum," which sound impressive but often are quite meaningless—if not downright wrong. In some cases, for all I know, the tweak may actually do something audible—but the explanation as provided in the marketing literature is nonsensical to the point of being quite comical.

Colored rocks, especially if sealed in a glass jar, are *not* going to absorb "bad vibrations" from out of the air, and nothing is going to influence the laser in your CD player so that it modifies the plastic in your CDs to reduce the error rate. Yet people, right now, are paying hundreds of dollars for these nonsensical products. (I assume that, because they expect to hear a difference, they actually do—otherwise, they would be very mad.)

I find it offensive that companies resort to these tactics, sad that people fall for them, and truly unfortunate that the magazines who should be debunking them take ads from these vendors instead (and are then unwilling to do anything to offend them).

Keith Levkoff
kLevkoff@panix.com

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