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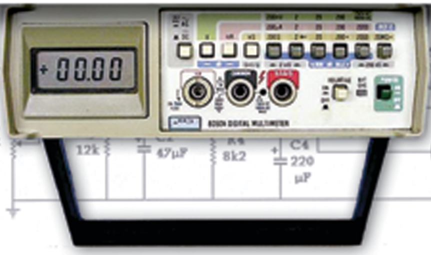

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Tubes

A Single-Ended "E-Linear"™ Power Amplifier

Back

By Pete Millett

This simple single-ended amp achieves good power output and low distortion by employing "E-linear" feedback in the output stage.

PHOTO 1: THE COMPLETED SE AMP.

I've been on a quest for the ideal tube power amplifier. Aren't we all?

To me, the ideal amplifier is simple, powerful, and sounds good—simple goals. Unfortunately, these goals are all at odds in the real world. But I've come up with a design that I think hits the mark pretty well (*Photo 1*).

POWER AMP TOPOLOGIES

There are several different basic topologies used to build tube power amplifiers. You can group them into two basic divisions: push-pull and single-ended; with numerous variations within each group, such as ultralinear connection, parallel feed, and so on, and you can use different devices (triodes, pentodes, and so on). There are devoted followers of all types who believe theirs is the best, often with great conviction!

Push-pull amplifiers, by their nature, cancel a good amount of the distortion that's inherent in a tube amplifier. They can also deliver more power than single-ended amplifiers, if they are biased such that one of the tubes is cut off on part of the signal swing (class-AB operation). But push-pull amps are more complex than single-ended amps, requiring a phase splitter, and twice as many output tubes. And the distortion cancellation is only of even-order harmonics, as I'll discuss shortly.

Single-ended amps are simpler, requiring fewer active stages. They do not benefit from the distortion cancellation of opposing tubes as a push-pull amplifier does, so they typically have high distortion. For example, 1% THD at 1W is not uncommon for a typical SE triode design.

FFTS AND THE SOUND OF HARMONIC DISTORTION

Without getting too technical, you can transform a representation of an electrical signal from the time domain into the frequency domain by doing a "Fourier transform." Think of a waveform on an oscilloscope, with time on the horizontal axis and amplitude on the vertical, as opposed to the display on a spectrum analyzer, with frequency on the horizontal axis. A pure sine wave would be represented by a single peak on a spectrum analyzer display, at the frequency of the sine wave.

If you distort that sine wave, peaks start to appear at other frequencies—the sine wave is no longer "pure." Nonlinearities in an amplifier will cause energy to appear at the harmonics of an input sine wave's frequency. This is called harmonic distortion.

As an example, if you apply a 1kHz sine-wave signal to the input of an amplifier, you will get at the output mostly an amplified 1kHz sine-wave signal, a reproduction of the input. But you will also get small amounts of energy at 2kHz, 3kHz, 4kHz, and so on. These are harmonic distortion products, the 2nd, 3rd, 4th, and so on, harmonics of the 1kHz inputs signal. The amount of each of the harmonics, or "harmonic profile," is key to our perception of how an amplifier "sounds."

Different types of distortion result in different mixtures of harmonics. The spectrum of a square wave, for instance, contains the fundamental frequency and all odd harmonics, but no even harmonics. You can see how an amplifier heavily into symmetric clipping, then, will generate many odd harmonics. You can see the other extreme

PHOTO 2: WOODEN BASE.

PHOTO 3: DRILLING THE CHASSIS PLATE.

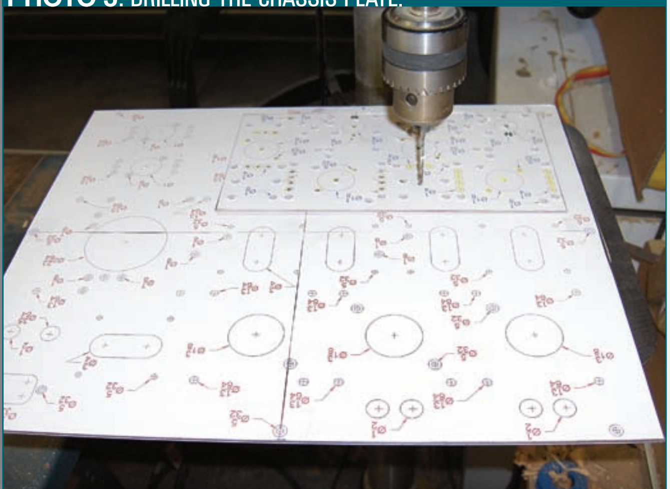


PHOTO 4: FINISHED CHASSIS PLATE.

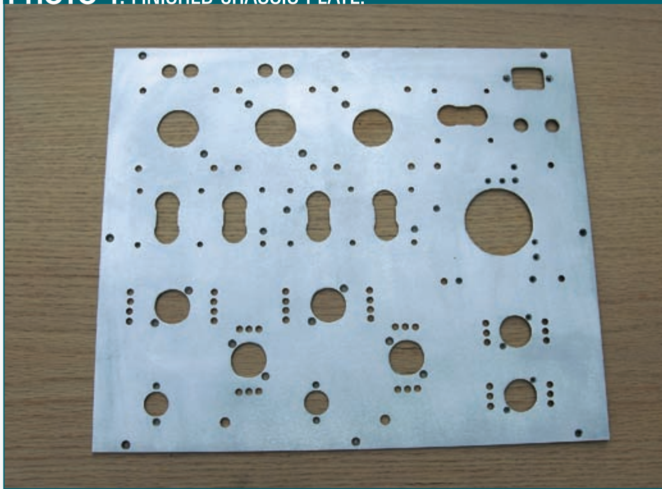
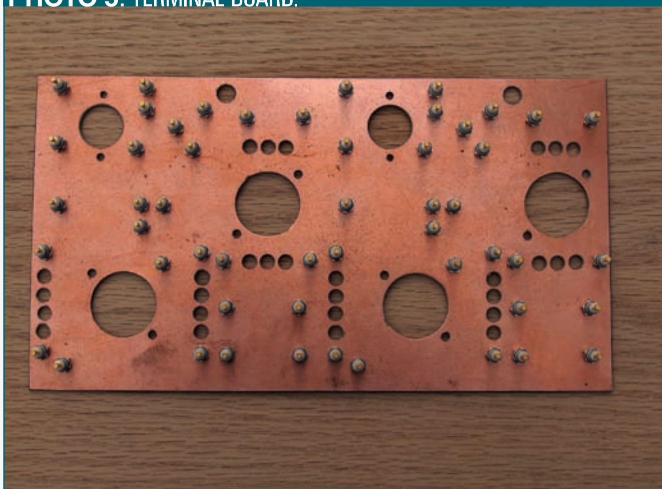


PHOTO 5: TERMINAL BOARD.



on a single-ended tube amplifier stage that is biased very close to plate current cutoff; it generates a lot of even—but not much odd—harmonic distortion.

Though it's a bit of an oversimplification, you can think of odd-order distortion as symmetrical; i.e., acting on both the upper and lower parts of a waveform. Even-order distortion, on the other hand, is asymmetrical, acting only on the lower or upper part of the waveform.

There have been many articles on the psychoacoustics of harmonic distortion, both old and new, so I won't dwell on it too much here. Some may disagree, but I believe the following to be true:

- less distortion is better than more distortion
- even-order distortion (2nd, 4th, and so on) is less audible than odd-order distortion (3rd, 5th, and so on)
- low-order distortion (2nd, 3rd, and so on) is less audible than high-order distortion (5th, 6th, and so on)
- low-order distortion tends to mask higher-order distortion

I think most people would agree that the ideal amplifier would have no distortion. You can't achieve that, so the next best thing is to minimize distortion, but keep the distortion profile such that the low, even-order harmonics dominate, and don't allow large amounts of high-order odd harmonics.

SINGLE-ENDED VS. PUSH-PULL

As I mentioned previously, push-pull (P-P) amplifiers, by their nature, cancel a large percentage of even-order harmonics. They do this because of their symmetry. The opposing tubes, each having similar nonlinearities but operating in opposite polarity, tend to behave together as a more linear amplifier.

If you look at the distortion spectra of push-pull amps compared to single-ended amps, they are typically very different. SE amps have lots of 2nd harmonic, and P-P amps have very little, since much of it gets cancelled. The harmonic profile of many P-P amps is dominated by the 3rd harmonic. This is the primary sonic difference between P-P amps and SE amps, and somewhat explains the debate about which sounds better. P-P amps usually have less overall distortion, but the distortion generated tends to be more odd-order.

To me, neither high-distortion SE nor P-P amps sound great. They are different, to be sure. I can listen to a high-distortion SE amp with some types of music and it sounds quite nice, but with complex music, the sound becomes muddy. In fact, I had convinced myself that 1–2% of 2nd harmonic actually sounded better than a lower distortion amplifier, until I listened to a wider variety of music. High-distortion P-P amps generally sound harsh and annoying to me, especially with simple music.

On the other hand, low-distortion SE or P-P amps can both sound quite good to me. So I'm a firm believer that the topology doesn't matter. It's the quantity, and nature of, the distortion generated that's important.

From a design standpoint, I favor an SE amp, mostly because of its simplicity, and the fact that by their nature SE amps tend to generate more even than odd-order distortion. The design problem, then, is how to achieve acceptably low distortion with a simple SE amplifier design.

THE E-LINEAR AMP DESIGN

You can approach the problem of keeping distortion low and power output high from a number of directions. One of the main approaches is to add feedback into the amplifier to correct nonlinearities.

You can apply feedback from output back to the input. This is called global, overall, or loop feedback. I'm not a big fan of this type. It can be difficult to stabilize such circuits, which tend to introduce some higher-order distortion products. I haven't listened to many amps with loop feedback that sound good.

You can also apply feedback locally to a single amplifying stage. An unbypassed cathode resistor is a simple implementation of local feedback. It lowers the gain and distortion by feeding back part of the output signal to the input and moving the cathode voltage in response to the plate current. The ultralinear connection, popularized by the hi-fi push-pull amps of the '60s, is also a form of local feedback, in which part of the output signal (typically around 40%) is taken from a tap on the output transformer and fed to the screen grid.

The E-linear amplifier starts with an ultralinear-connected output stage and adds another local feedback path—from the output transformer (and screen grid) to the control grid of the output tube. This combination of multiple local feedback paths dramatically lowers the distortion and output impedance of the output stage and gives you performance that cannot be obtained with pentode, triode, or ultralinear connection alone. It sounds good, too!

Feedback from plate to grid is certainly nothing new. An article published in 1938 by an RCA engineer describes in detail the use of plate-grid feedback to tailor the characteristics of a beam power tube for audio amplifier use. The ultralinear connection has been around since the early 1950s—an invention usually credited to David Hafler. The E-linear configuration can be viewed somewhat as a combination of the two.

The term "E-linear" was coined recently by Emotive Audio (www.emotiveaudio.com) as a description for the combination of ultralinear feedback to the screen grid of the output tube and to the driver stage, or to the control grid of the output tube. I'd be surprised if nobody had tried this earlier, but I haven't found any references to this type of output stage connection. It seems that several people started playing with this configuration at about the same time, including myself.

DESIGN DETAILS

I wanted this amplifier to have only two stages and to generate at least 10W of output power. To accomplish this, as well as have enough gain to be able to apply the E-linear feedback scheme, high transconductance (Gm) pentodes (or beam tetrodes) are almost a must. I tried many different tubes, both driver and output types. To get the output power and low distortion that I wanted, I wound up using a KT88 beam power tube as the output tube and a D3a pentode as the driver.

The D3a (also known as the 7721) is a German tube used in telephone equipment, made by Telefunken, Valvo, and Siemens. The D3a is uncommon in North America, so I know many of you will ask, "Isn't there another tube that's easier to find that will work?" My answer is that although there are other tubes that will work, I don't know of any that will work nearly as well. I tested many pentodes (for details, see my web page at www.pmillett.addr.com/pentodes.htm). The

FIGURE 1: AMPLIFIER SCHEMATIC.

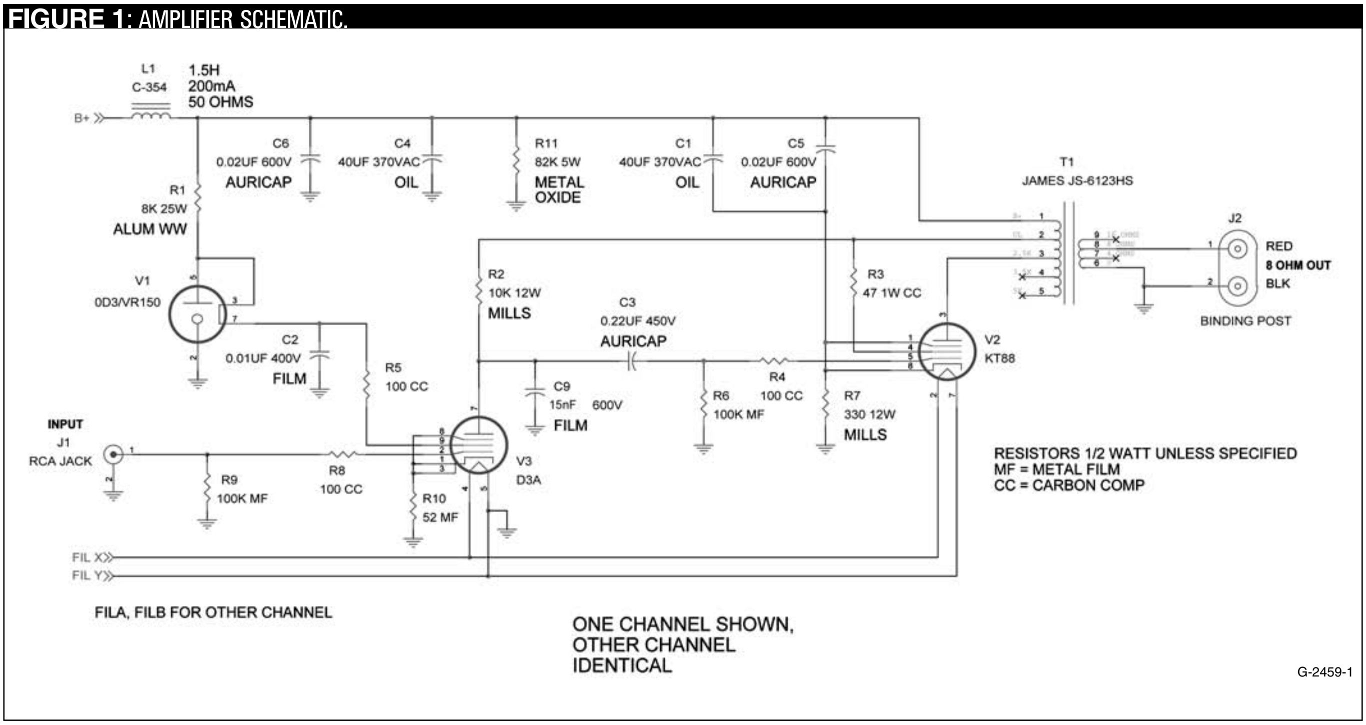


PHOTO 6: CAPACITOR MOUNTING CLAMPS.

D3a performed the best by far in this particular circuit.

I can't say with certainty exactly why the D3a worked the best in this circuit. There are certainly other driver pentodes that have higher Gm and lower distortion than the D3a when measured alone. But in combination with the KT88 output stage, the D3a was the clear winner. This may be partly due to distortion cancellation between the D3a and KT88—two SE amp stages can partially cancel even-order harmonics in much the same way that a P-P amp does. Perhaps the D3a is just a very good match for the KT88.

D3a tubes are easy to find and are fairly inexpensive in Europe. These days, it's as easy as getting on the Internet or the telephone (don't forget the time difference though!) and placing an order with a credit card. In a few days your tubes will arrive in the mail, direct from Europe. Some suppliers who sell D3a tubes are listed at the end of this article.

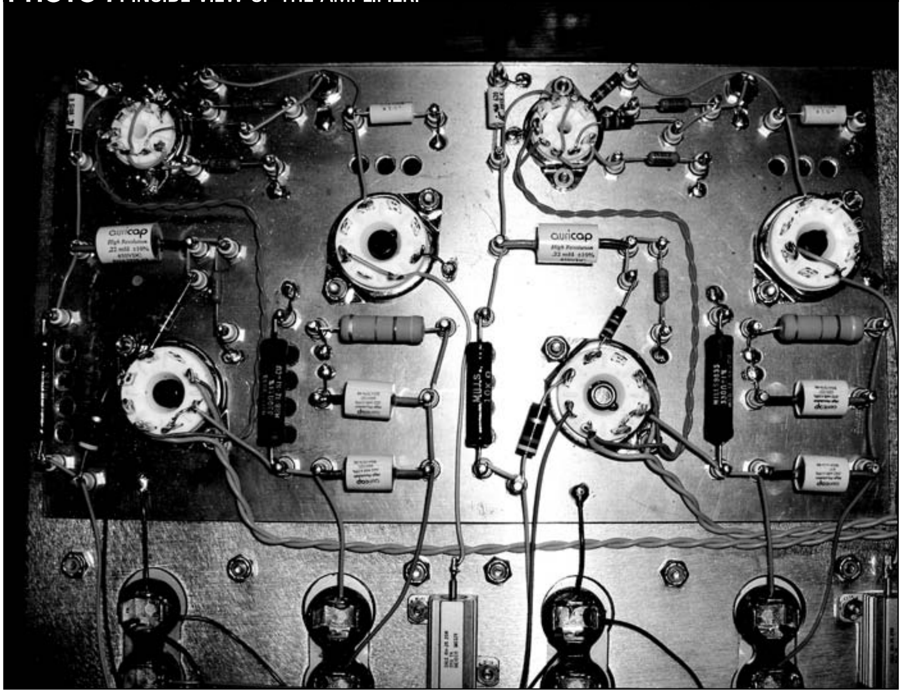
The screen grid of the D3a driver stage is supplied with a shunt-regulated 150V supply. Regulation is provided by a 0D3/VR150 gas discharge VR tube, one for each channel.

The plate of the driver is capacitively coupled to the KT88 control grid using a good-quality coupling capacitor. I've never been really excited about using designer audiophile capacitors, but this time I decided to see what I was missing. I used Auricap film capacitors, which have received very good reviews, are reasonably priced, and appear to be very well built.

The resistive plate load for the D3a is connected to the screen grid of the KT88, and both are connected to a ~40% ultralinear tap on the output transformer, which provides the E-linear feedback. Refer to *Fig. 1*, the amplifier schematic, for details.

Since a pentode has very high plate resistance, the driver's output impedance is dominated by the plate load resistance, which needs to be low enough to provide plenty of drive to the KT88 control grid. It also allows the feedback from the ultralinear tap to be coupled directly to the KT88 control grid. A pentode almost acts like a constant-current source, so any AC voltage you apply to the top of the load resistor will appear

PHOTO 7: INSIDE VIEW OF THE AMPLIFIER.



on the plate virtually unchanged.

If you used a triode driver, the triode's relatively low plate resistance would form a voltage divider with the plate load resistor and decrease the amount of feedback. The E-linear scheme still works in this situation, but the amount of feedback effectively getting coupled to the output tube grid is reduced by approximately a factor of Rp/(Rp+Rl), where Rl is the plate load resistor and Rp is the driver tube's plate resistance.

You will notice a small (1500pF, or 0.0015µF) capacitor from the D3a plate to ground. This capacitor provides a drop in gain at very high frequencies and is advisable because any inductance in the driver's plate circuit—such as a wirewound plate load resistor, or even the inductance of the wiring—results in increasing gain at high frequencies. This gain, in combination with phase shifts in the output transformer, can result in RF oscillation. I selected this capacitor, which mitigates the gain increase to prevent oscillation, based on tailoring the frequency response and by minimizing any ringing on squarewave signals.

Along those same lines, I used gridstopper resistors on all the grids. These resistors, which should be carbon composition for minimum inductance, help eliminate RF tuned circuits from being parasitically formed with the wiring of the grid circuits, which can also lead to high-frequency oscillation in high-Gm tubes.

To achieve the lowest distortion and output impedance, run both the D3a driver and KT88 output tube close to their dissipation limits. I can't tell you how this will affect tube life, but since the D3a is a long-life telecommunications tube, I expect it will still last quite a while. I don't have much experience with the different KT88 tubes, but I know many amplifier designs using KT88s run at or over the limits successfully.

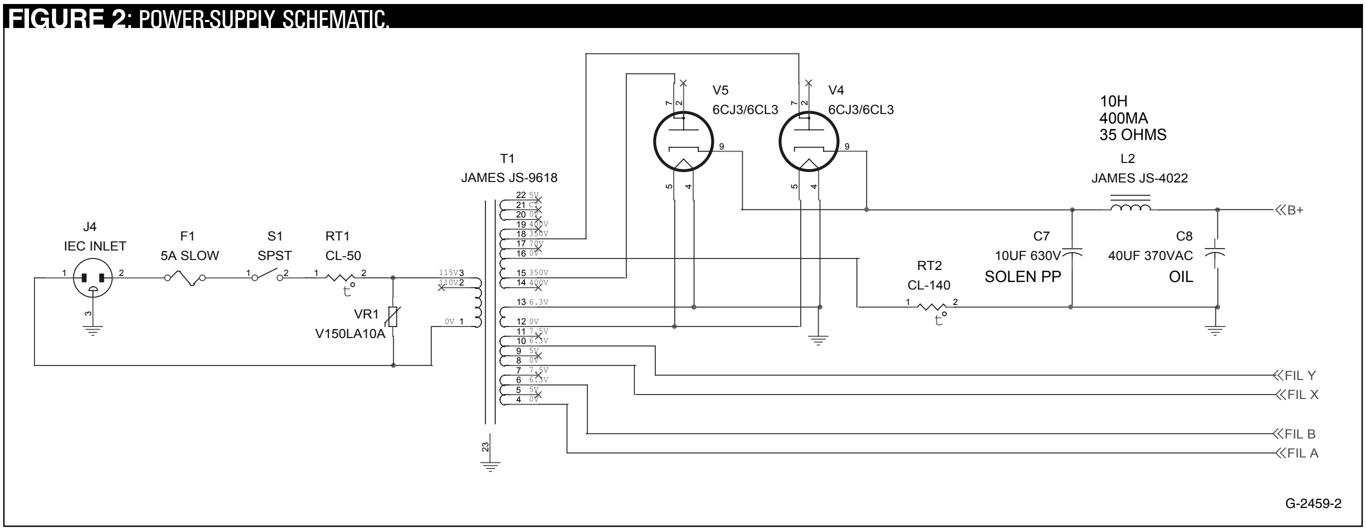
The KT88 is biased with a cathode resistor. You'll notice that there is no cathode bypass capacitor—or actually, that the cathode is bypassed to the B+ side of the output transformer. This "ultrapath" capacitor provides the AC current return for the output stage, so most of the current flows in a loop through this capacitor, the OPT, and the KT88. The theory here is that you need to keep as much of the signal current out of the power supply as you can.

If you can live with lower output power, you can also use this same circuit with any number of pentodes or beam tetrodes. I tried the 12E1, 6L6, EL34, 807, and a few others, singly and in parallel, and all worked well. The 12E1 performed almost as well as the KT88. Just change the cathode bias resistor to get the operating current low enough so the plate dissipation is just under the rated limit and select the output impedance tap appropriate for the tube.

OUTPUT TRANSFORMERS

Probably the single most critical component in a SE amplifier is the output transformer. I've tried quite a few, including some expensive Japanese transformers I picked up while traveling.

FIGURE 2: POWER-SUPPLY SCHEMATIC.



I heard through the Internet of an inexpensive brand of transformers made in Taiwan, called James. James iron was (at the time) unavailable in the US, but was quite popular among the East Asian (Hong Kong and Taiwan) DIY crowd. Well, hard-to-get parts hadn't stopped me before!

I bought a pair of SE output transformers, James JS-6123HS, on eBay to see whether they were as good as what I'd heard. They were indeed quite good, measuring just as the specs indicated, and looking every bit as good as the Tango XE-20S, a much more expensive transformer available in Japan.

Like the XE-20S, the JS-6123HS is a "universal" SE output transformer, with taps for 2.5k, 3.5k, and 5kΩ primary connections, as well as an ultralinear (screen grid) tap. It also has 4, 8, and 16Ω taps on the secondary, which makes for a versatile transformer, great for experimenting. This transformer became the basis for many breadboard experiments, one of which resulted in the amplifier described here.

The fact that James transformers weren't available in the US actually delayed my writing this article for a time. I figured most readers wouldn't be very happy having to pay to ship 50 lb of iron from Taiwan (air shipping costs almost as much as the transformers). I also wasn't sure how well other transformers would work in this design, since it is a little unconventional. It's possible that some transformers would have enough phase shift at high frequencies that they may not be stable with the E-linear feedback configuration.

Luckily, the availability problem has recently been fixed. Euphonia Audio (www.euphoniaaudio.com) in New Jersey is now importing the full line of James transformers, so North American buyers have access to their products, which I think have the best cost/performance of all the audio transformers I've used.

POWER SUPPLY

I built a stereo amplifier, and for a simple two-stage amp, this design's power requirements are substantial. I used a B+ voltage of 425V, with a total current requirement of about 360mA. This breaks down to 100mA for the output stage, 40mA for the driver, and 30mA for the shunt regulator for the driver screen, per channel. The D3a driver draws significant screen current as it nears saturation, so lots of current needs to flow through the shunt regulator tubes to keep the screen voltage from dropping out on peaks.

A big power transformer is required for this amplifier, with a B+ winding of around 760V CT at 400mA DC. James has a wide range of power transformers, and luckily for North American builders, Taiwan has the same power supply (115V 60Hz) as the US. I used a JS-9618 mains transformer, which I lugged back from Taiwan in a suitcase. It's a beast at almost 20 lb!

I used tube rectification and CLCLC filtering in the power supply. Refer to *Fig. 2*, the power supply schematic, for details.

You need large rectifier tubes to deliver this much power supply current. I tried paralleled 5U4GB rectifiers, but the voltage drop across the rectifiers was quite large, making a "soft" HV supply. I also tried mercury vapor rectifiers, which worked well but required extra circuitry to provide a warm-up time, and caused a big start-up surge once I applied plate voltage.

I settled on color TV damper diode tubes

(6CJ3, 6CL3, or equivalent) as rectifiers. They provide a low voltage drop (actually lower than the Mercury-vapor tubes), a nice, slow, controlled start-up, and simplicity. They're also very inexpensive.

Filtering is provided by a CLC filter shared by both channels, followed by a second separate LC filter for each channel to provide some isolation. Sticking with the matching James iron, the first filter inductor is a 10H, 400mA JS-4022, which is the same size and shape as the output transformers. The second choke is a 1.5H, 200mA C-354 Dynaco replacement choke, a nice part available from Triode Electronics.

I used all polypropylene filter capacitors—a small axial-leaded Solen cap in the first stage, and oil-filled motor run capacitors for the others. Polypropylene-in-oil motor run caps are quite a bargain. They are available new on eBay for only a few dollars each, or you can order them from distributors such as Allied Electronics for a bit more. A 370V AC rated capacitor is good for 600V DC (I confirmed this with the manufacturer). I also bypassed the motor run caps in the signal path with small Auricaps.

When designing power supplies with tube rectifiers, you must be careful not to exceed the peak-current capability of the rectifier tubes. Many designers use a capacitor that's too large in the first stage of the filter, which causes the peak current through the rectifier to be very large, which in turn drastically shortens the rectifier's life. Large current spikes also tend to couple noise everywhere. Just to be certain, I mea-

FIGURE 3: RECTIFIER CURRENT AND OUTPUT VOLTAGE.

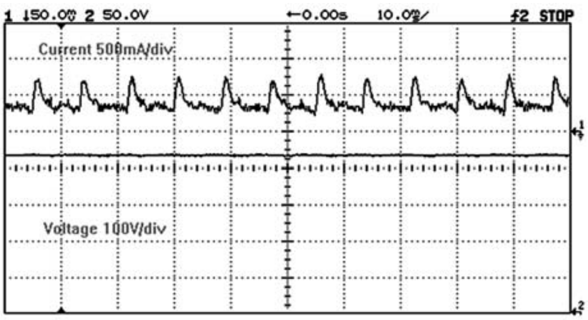


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FIGURE 4: CHASSIS PLATE.

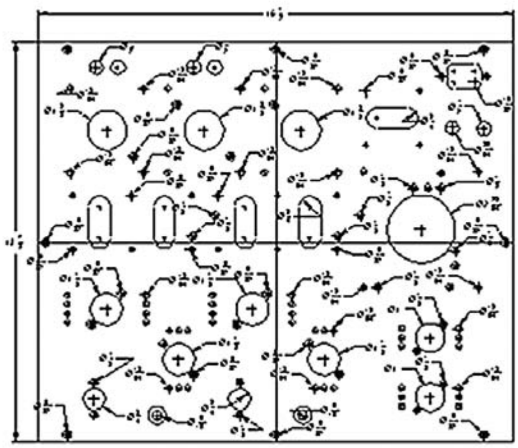

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FIGURE 5: TERMINAL BOARD DIMENSIONS.

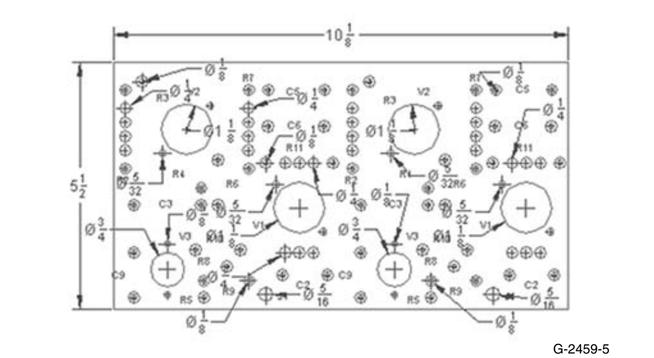
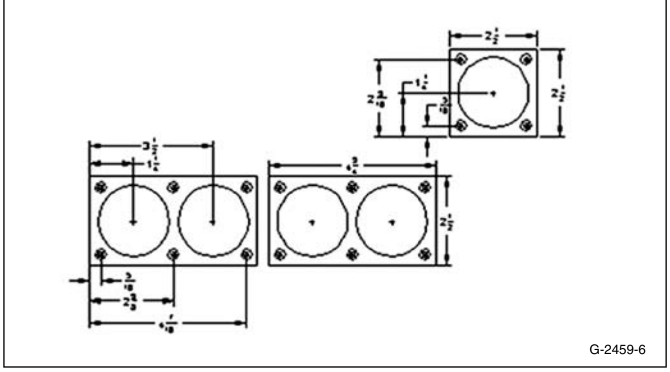


FIGURE 6: CAPACITOR CLAMPS.



sured the repetitive peak current as shown in *Fig. 3*, which shows the peaks to be about 750mA. This is well within the ratings of the 6CJ3.

I used an inrush current limiter between the transformer center tap and ground to limit the current surge at start-up. I suspect that this is unnecessary with the slow-starting damper diode rectifiers, but I left it in after experimenting with mercury-vapor tubes, where it was definitely needed.

IMPLEMENTATION

I built the amplifier on a metal chassis plate, mounted to a wooden base (*Photo 2*). I used a terminal board to mount most of the components inside. The amplifier measures about 17″ wide (which is about as large as I could fit into my stereo rack) and is just over 14″ deep.

I fabricated the metal chassis plate, starting with a cut-to-size piece of 0.075″ thick 6061-T6 aluminum sheet I bought from Online Metals. I printed 1× scale drawings of the chassis plate (*Fig. 4*) on full-sheet sticky label stock, and applied them to the sheet. I find that this is an easy, accurate way to transfer a design to a piece of metal. Just make sure you measure the prints so that your printer is really printing at the correct scale!

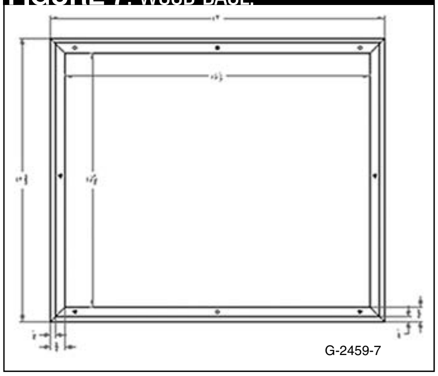
I then cut the holes, using a drill press, step drill (Unibit), chassis punches, and a reciprocating saw. *Photo 3* shows the panel on the drill press. I finished the aluminum simply by texturing the surface with a belt sander. The finished panel is shown in *Photo 4*.

Note that there are ventilation holes drilled opposite the power resistors, near the tubes. This provides a bit of airflow through the chassis. The amplifier still becomes quite warm during operation, because there is quite a bit of power being dissipated.

I built the amp using a terminal board to hold many of the components. I talked about this construction method extensively in an article that was published previously ("A Low-Mu Triode Preamp," *aX* 2/04), so I won't dwell on it here.

I used a piece of blank PCB material to mount single-point solder turret terminals, which are positioned exactly where you want to place components. I used the copper foil as a ground plane. *Figure 5* shows the dimensions of terminal board, with parts placement in *Fig. 5A*; *Photo 5* shows the complete (unwired) terminal board.

FIGURE 7: WOOD BASE.



Mounting the oil-filled motor run capacitors presented a bit of a challenge. You can buy metal clamps to attach them to the chassis, but they're large and not very nice looking. I made some clamps (*Photo 6*, *Fig. 6*) to hold the caps onto the chassis, using some plastic sheet that I had sitting around (you could also use wood or other material). Drill a hole through the sheet just big enough for the capacitor body to slip through, but not big enough for the rolled lip on the can to pass through (2″ worked just right for the caps I used).

FIGURE 8: INSIDE ASSEMBLY.

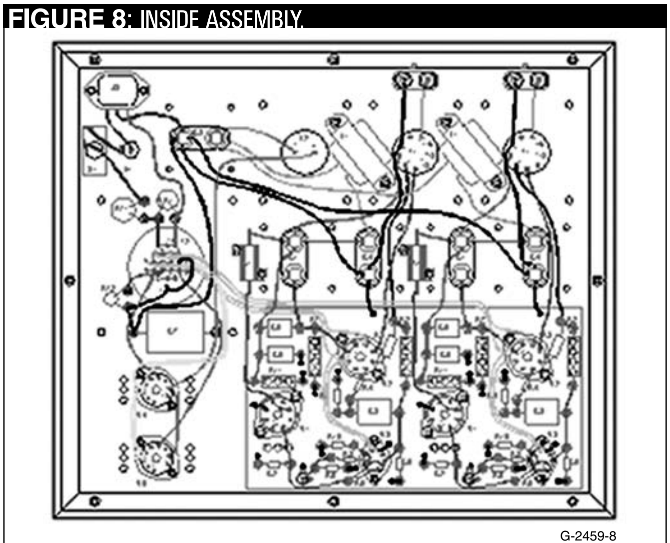
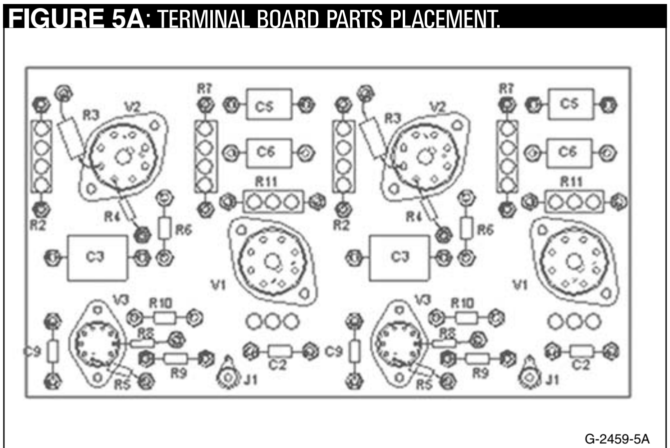


FIGURE 5A: TERMINAL BOARD PARTS PLACEMENT.



Using a router, I cut a small rabbet into the plastic for the lip, so that the completed clamp sits flush on the chassis, capturing the lip on the can and holding it to the chassis. I slipped a thin rubber o-ring around the cap before passing it through the clamp to provide a firm grip on the cap, then secured the clamp to the chassis plate with screws.

I made the wooden base in the same way as the one I made for the low-mu preamp using a perforated aluminum sheet from a local surplus store for the bottom to allow some ventilation. You can see the finished wooden base in *Photo 7*. *Figure 7* shows the dimensions of the base.

To help you duplicate this design, you can download full-scale drawings as well as CAD source files (should you wish to make some changes) from my web site at www.pmillett.addr.com. You can then use a printer to generate accurate templates to fabricate the parts. Parts are listed in *Table 1*.

WIRING AND COMPONENT INSTALLATION

I used the space underneath the large transformers and inductor to hide the screw heads for some of the components mounted on the underside of the chassis, such as the second stage filter chokes and some power components. I mounted these to the chassis plate first, using flush-head screws. I mounted the metal-cased wirewound resistors feeding the 0D3/VR150 shunt regulator tubes for the drivers to the chassis in this way as well, so that the chassis plate can act as a heatsink for them. A little thermal grease under the resistors is a good idea to improve the heat transfer.

I secured the terminal board to the chassis plate with the screws that mount the tube sockets. Then you can mount the remaining chassis parts (transformers, chokes, and so on) to the top of the chassis. I mounted the motor run caps to the chassis using the fabricated clamps.

With all the parts mounted to the chassis, install the wiring. I used Teflon hook-up wire, both solid and stranded. Solid wire, which is easiest to wrap around the terminals, is best to make a tightly twisted pair, such as I used for AC filament wiring. Refer to *Fig. 8* for details of the wiring. Note that not all tube sockets are oriented with the same rotation as the ones I used, so you may need to make some adjustment.

The sockets I used were very nice Chinese ceramic ones from Angela Electronics. Hint: select the tube sockets you want to use before you start punching holes in the chassis, since they vary quite a bit in actual diameter. The sockets I bought turned out to be a little small for the holes I punched—not enough to cause a problem, but they were a bit loose fitting.

After completing the wiring, I installed the leaded components on the terminal board and single-point terminals, wrapping their leads around the terminals and soldering. Adding the parts after the wiring makes it easier to change them after the fact, if you want to try different component values. A view of the inside of the completed chassis is shown in *Photo 7*.

MEASUREMENTS

My goals included an output power of at least 10W, low output impedance, and low distortion. The measurements provide an indication of how well I met these goals.

At 5% THD (which is roughly the onset of clipping), the amplifier delivers 15W (per channel) into an 8Ω load. The output impedance on the 8Ω output tap measures about 2Ω, which gives a damping factor of 4, respectable enough for a SE amplifier. At 1W output, THD + noise is about 0.2%, lower than most typical SE triode amplifiers. THD rises gradually to 1.8% at 10W out.

Noise at the amplifier output measured 0.5mV in the left channel and 1mV in the right, consisting of mostly 120Hz hum. I suspect the right-channel noise is higher than the left due to coupling between the filter choke and the right channel output transformer, because I didn't see this noise when the amp was in the breadboard stage and spread out a bit more. This level of noise is not audible with my ear next to my 92dB efficient speakers, but could be an issue for some. If so, I'd recommend moving the power-supply components away from the amplifier circuitry as much as possible.

The amplifier's frequency response at 1W into 8Ω is flat within ±1dB between 10Hz and 25kHz. The −3dB points are 8.5Hz and 73kHz. Low-frequency performance is limited by the finite size and inductance of the output transformers; distortion increases below about 60Hz at full power. This is typical of SE amplifiers, and really doesn't present much of a problem with normal music, because there is very little energy at the lowest frequencies.

One indication of stability and frequency response is the amplifier's response to an audio-frequency square wave. Instability often shows up as ringing on the edges of the square wave, especially on amps that employ feedback through transformers. Ringing can also be the result of the inherent resonances between the inductance and interwinding capacitance in the output transformer, and can give an indication of the quality of the output transformer.

Figure 9 shows an oscilloscope trace of a 1kHz square wave (with very fast edge rates on the input). You can see that although there is a slight amount of ringing, it is well damped, and the response looks very good compared to many tube amplifiers. This holds true over all output powers, with virtually no change in the waveform up to clipping.

My other goal was to achieve an ideal harmonic profile, one in which the 2nd harmonic is the highest, followed by the 3rd, 4th, and so on. To measure this, I applied a 1kHz sine wave to the input and did an FFT analysis of the output, using my digital oscilloscope. The scope doesn't have very high resolution as a spectrum analyzer, but it's good enough for a relative indication of whether I met this goal.

Figure 10 gives the FFT of the output at 1W into 8Ω, showing the fundamental through 4th harmonics (the 5th and higher harmonics were in the noise, so I didn't proceed further). You can see that the 2nd harmonic is the highest, with the 3rd and 4th present in lesser proportions. This is the ideal harmonic profile that I was aiming for.

Figure 11 shows the distortion residual. This is what's left after you remove the fundamental signal (1kHz, in this case). You can see that there is mostly 2nd harmonic (2kHz) signal, as well as some 3rd harmonic (3kHz).

Figure 12 shows the spectrum at 10W. At this power level, you lose the ideal harmonic profile, as the 3rd harmonic has become larger than the 2nd. This is typical of most amplifiers—as they approach clipping, distortion becomes more symmetrical, and the odd harmonics start to increase. This is still not so bad, as there is plenty of 2nd and 4th harmonic to help mask the 3rd. Keep in mind also that the average power delivered when listening is usually less than a watt, unless you're fond of listening at hearing-damage levels, or have very inefficient speakers.

FIGURE 9: SQUARE-WAVE RESPONSE (1KHZ, 1W INTO 8Ω).

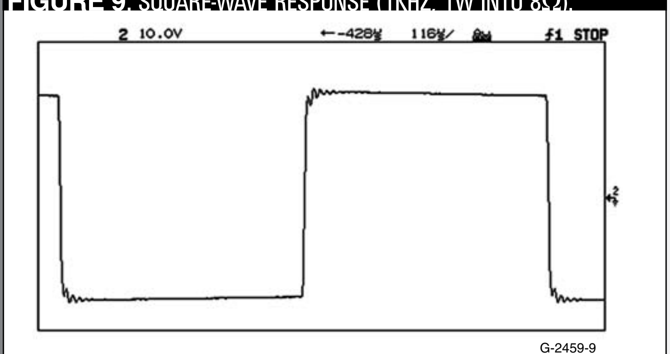


FIGURE 11: DISTORTION RESIDUAL (1KHZ, 1W INTO 8Ω).

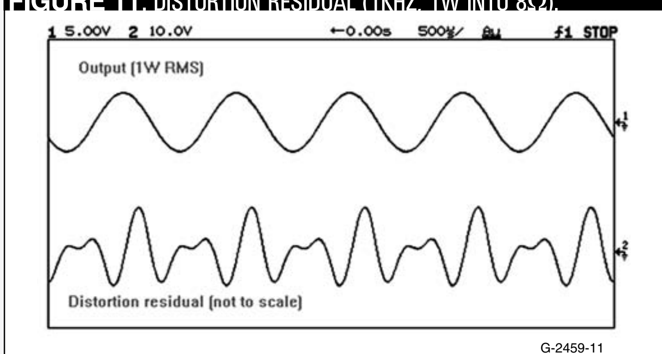


FIGURE 10: FFT SPECTRUM ANALYSIS (1KHZ, 1W INTO 8Ω).

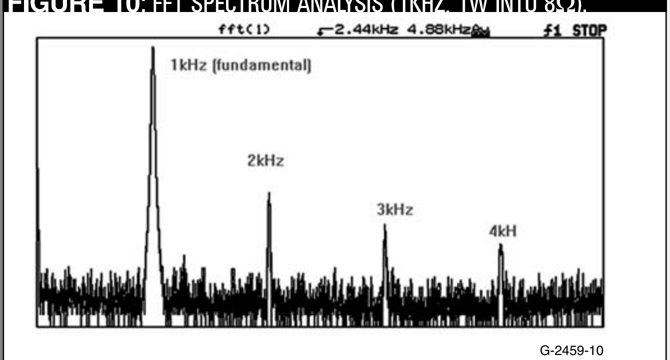
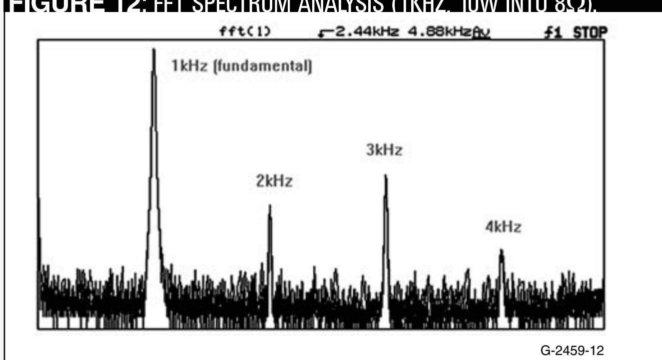


FIGURE 12: FFT SPECTRUM ANALYSIS (1KHZ, 10W INTO 8Ω).



LISTENING IMPRESSIONS

I confess to being an engineer, so I tend to put a lot of faith in the measurements. But the bottom line is how the thing sounds.

This is definitely the best-sounding amplifier I've built! It has detailed sound and doesn't muddy up complex music the way I've found a higher distortion SE amp does. It also lacks the harsh sound that I associate with a high-distortion P-P amp. Bass response is solid, not loose or boomy. I find myself drawn into the music more than with my other amplifiers.

Overall, I deem it a success! *aX*

RESOURCES

www.audioasylum.com - Great Internet bulletin board for all things audio. Check out the "Tube DIY" section for lots of discussion about tube circuits.

www.pmillett.addr.com - The author's web site, where you can download full-scale drawings of this project.

http://jianshin.myweb.hinet.net/ - James Audio's website. In Chinese, but you can figure most of it out.

SOURCES

Euphonia Audio – US importer of James audio transformers
(917) 861-3471
www.EuphoniaAudio.com

Online Metals – Metal sheet material, cut to size
(800) 704-2157
www.onlinemetals.com

Digi-Key – Full-line parts distributor
(800) 344-4539
www.digikey.com

Mouser Electronics – Full-line parts distributor
(800) 346-6873
www.mouser.com

Handmade Electronics – Mills resistors, tube-related parts
(610) 432-5732
www.hndme.com

Allied Electronics – Motor-run caps
(800) 433-5700
www.alliedelec.com

Triode Electronics – Tubes, C-354 chokes, tube-related parts
(773) 871-7459
www.triodeelectronics.com

Antique Electronic Supply – Tubes, capacitors, tube-related parts
(800) 706-6879
www.tubesandmore.com

Angela Instruments – Tube sockets, tube-related parts
(301) 725-0451
www.angela.com

Ask Jan First (Die Wustens) – European tubes (D3a)
+49-6154-575556 (Germany)
www.askjanfirst.com

Audiotubes.de – European tubes (D3a)
+49-8373-921657 (Germany)
www.audiotubes.de

Halfin – European tubes (D3a)
+32-2-209-2981 (Belgium)
www.halfin.com

TABLE 1

Parts list

REFERENCE	QTY	DESCRIPTION	VENDOR	VENDOR PN
R1 (L+R)	2	8k 25W Aluminum wire-wound resistor	Mouser Electronics	71-RH25-8K
R2 (L+R)	2	10k 12W Mills MRA12 resistor	Handmade Electronics	MRA12 10K0
R3 (L+R)	2	47Ω 1W carbon comp resistor	Digi-Key	OA470K
R4, R5, R8 (L+R)	6	100Ω ½W carbon comp resistor	Digi-Key	OF101J
R6, R9 (L+R)	4	100k ½W RN60 metal film resistor	Mouser Electronics	71-RN60D-F-100K
R7 (L+R)	2	330Ω 12W Mills MRA12 resistor	Handmade Electronics	MRA12 330R
R10 (L+R)	2	52.3Ω ½W RN60 metal film resistor	Mouser Electronics	71-RN60D-F-52.3
R11 (L+R)	2	82k 5W metal oxide resistor	Mouser Electronics	286-82K
C1, C4 (L+R), C8	5	40µF 370V AC polypro-in-oil capacitor	Allied Electronics, eBay (see text)	225-5036
C2 (L+R)	2	0.01µF 400V film capacitor	Antique Electronic Supply	C-TD01-400
C3 (L+R)	2	0.22µF 450V Auricap film capacitor	Antique Electronic Supply	C-AUD22-450
C5, C6 (L+R)	4	0.022µF 600V Auricap film capacitor	Antique Electronic Supply	C-AUD022-600
C7	1	10µF 630V Solen polypropylene capacitor	Handmade Electronics	4-10582
C9 (L+R)	2	0.0015µF 600V film capacitor	Antique Electronic Supply	C-TD0015-630
V1 (L+R)	2	0D3/VR-150 VR tube	Antique Electronic Supply	T-0D3-A_VR-150
V2 (L+R)	2	D3a/ 721 pentode	Ask Jan First, audiotubes.de (see text)	D3a
V3 (L+R)	2	KT88 output pentode	Triode Electronics	KT88ehmp (pair)
V4, V5	2	6CJ3/6CH3 damper diode tube	Antique Electronic Supply	T_6CJ3_6CH3
at V1, V3 (L+R)	4	Octal tube socket 1.125″ hole	Angela Instruments	Octal socket #3
at V2 (L+R)	2	9-pin miniature tube socket 0.75" hole	Angela Instruments	9-pin socket #7
at V4, V5	2	9-pin novar (compactron) socket 1.063" hole	Antique Electronic Supply	P-ST9-103
T1 (L+R)	2	James JS-6123HS SE output transformer	Eupohonia Audio	James JS-6123HS
T2	1	James JS-9618 power transformer	Eupohonia Audio	James JS-9618
L1 (L+R)	2	C-354 choke (1.5H 200mA 50Ω)	Triode Electronics	C354
L2	1	James JS-4022 choke (10H 400mA 35Ω)	Eupohonia Audio	James JS-4022
RT1	1	CL-50 inrush limiter (7Ω 5A)	Mouser Electronics	527-CL50
RT2	1	CL-140 inrush limiter (50Ω 1.1A)	Mouser Electronics	527-CL140
RV1	1	V150LA10Ametal-oxide varistor (MOV)	Mouser Electronics	570-V150LA10A
F1	1	5A slow-blow fuse	Mouser Electronics	5760-13005
at F1	1	Fuse holder	Mouser Electronics	5767-345311
S1	1	SPST power switch 120V 5A minimum rating	Mouser Electronics	691-2FA53-73/TABS
J1 (L)	1	RCA input jack (black)	Mouser Electronics	164-4321
J1 (R)	1	RCA input jack (red)	Mouser Electronics	164-4325
J2 (L+R)	1	Dual binding post speaker terminal	Antique Electronic Supply	S-H263
J3	1	IEC power inlet/filter 120V 5A minimum rating	Mouser Electronics	592-6EEAP

A Hybrid Tube/MOSFET Headphone Amplifier

Back

By Erno Borbely

This noted audio expert makes his popular hybrid amp design available for headphone use.

In the 1/98 issue of *Glass Audio*, I wrote about a hybrid tube/MOSFET line amp, which, because of its musical sound, became a very popular amplifier ("Low-Voltage Tube/MOSFET Line Amp," which also appears on www.borbelyaudio.com under Special Articles). DIY amateurs wished to use it in many different applications, such as CD buffer, I/V converter, power amplifier, and headphone amplifier. It worked very well in all line-level applications, but the second stage was not laid out for high-current operation, so driving headphones was not possible. I have therefore redesigned the circuit to allow high-current operation.

The result is the EB-804/421, a single-ended (SE) pure Class-A amplifier, capable of driving headphones between 32 and 600Ω. The amplifiers need ±15 to ±24V regulated supplies at 160/100mA and 6.3V DC at 300mA for the tube heater. I recommend feeding the amps from separate supplies. The PCB for one amp is 90 × 80mm.

FIGURE 1: EB-804/421 SE CLASS-A HEADPHONE AMPLIFIER.

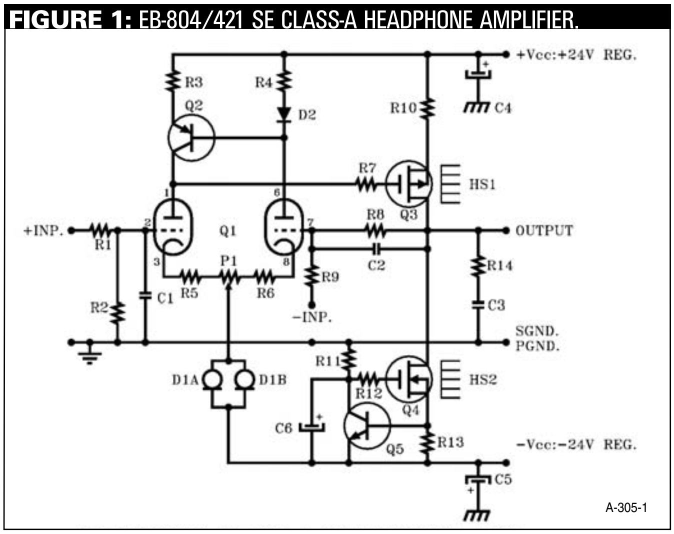
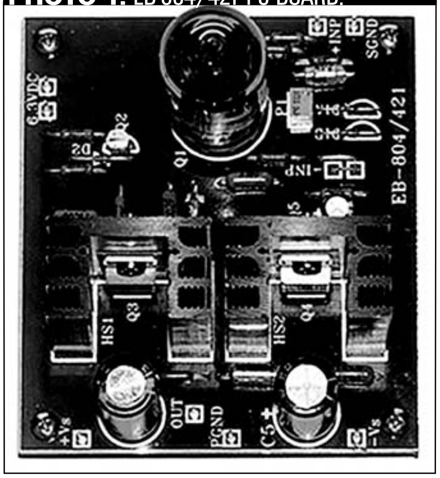


PHOTO 1: EB-804/421 PC BOARD.



CIRCUIT DESCRIPTION

The schematic is shown in *Fig. 1*. The topology is the same as the hybrid tube/MOSFET line amp. Q1 is a double triode that operates as a differential amplifier, with approximately 2mA in each of the triodes. A constant-current diode D1, which supplies the source current to the differential amp, includes two J508 or E-202 diodes in parallel. You can also use a single J511, which delivers 4.7mA.

The two anodes, which produce out-of-phase signals, are converted to a single-ended signal using a current mirror composed of Q2, D2, and resistors R3/R4. Q3, a P-channel MOSFET in TO-220 package, is used in common-source mode as a Class-A single-ended second stage. I replaced its drain resistor with a second constant-current source, supplying the Class-A current of 100 or 160mA.

The constant-current source, which increases the gain and improves the linearity of the second stage, is made up of Q4, an N-channel MOSFET in TO-220 package, and its associated components. I used the Hitachi 2SJ79 and 2SK216 for Q3 and Q4, respectively. You can also use the Toshiba 2SJ313 and 2SK2013, but note that the pinout is different from the Hitachi (GDS versus GSD).

The amplifier can work with a ±15V to ±24V supply. The maximum dissipation allowed for Q3 and Q4 is 2.4W each, so the supply voltage determines the maximum current. At ±24V the current is 100mA and at ±15V it is 160mA. Resistor R13 sets the current: it is 6R8 for 100mA and 3R9 for 160mA.

You must heatsink Q3 and Q4. I am using the SK76–37.5 with 8K/W thermal resistance. The temperature on the heatsinks is about 55° C, so proper ventilation is absolutely necessary! The PS/regulator I recommend for the hybrid tube/MOSFET headphone amplifier is the EB-802/243.

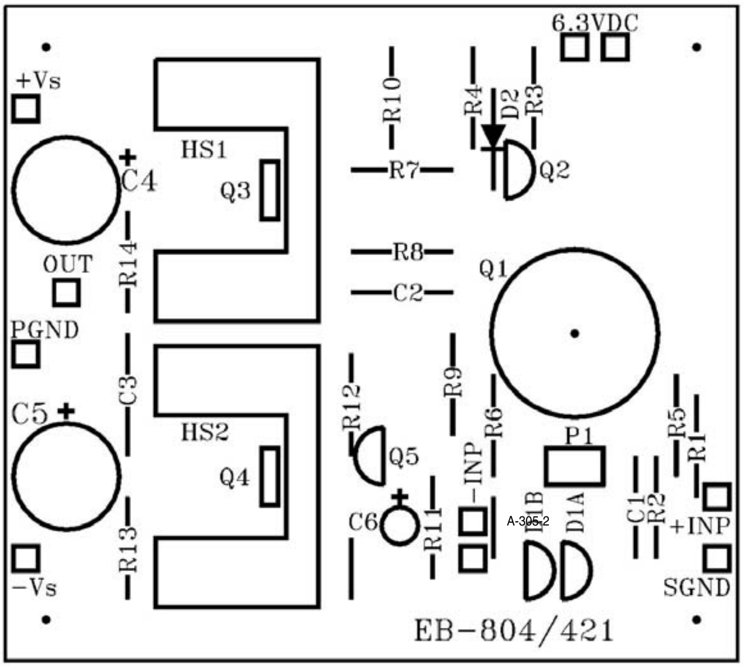
The input tube requires a 6.3V/350mA heater supply. Use a well-regulated/low-ripple supply for this (EB-793/204 is recommended). I recommend that you ground the negative side of the heater supply to the PGND on the PCB.

LINEARITY NOTES

The input tube dominates the overall distortion characteristics of the amplifier. Tubes of different manufacturoutput stabilize to 0V. Alternatively, you can leave the heater on all the time or in a stand-by mode with—say—4V, in which case you can apply the full heater voltage and the supply voltage simultaneously.

The feedback resistors R8 and R9 set the closed loop (CL) gain of the amp. Normal gain is 10×, or 20dB. Changing R9 can change this gain. CL output impedance is 15Ω. Equivalent input noise depends on the tube used and is 1.2–1.5μV!

FIGURE 2: STUFFING GUIDE FOR THE HYBRID TUBE/MOSFET HEADPHONE AMP (90 × 80MM).



The maximum output power into different loads depends on the supply voltage and the available current from Q4. With ±24V and 100mA in the second stage, the amp delivers >100mW into 32Ω and >250mW into 600Ω at 1% THD. With ±15V and 160mA the power into 32Ω increases to 300mW at 1% THD.

The maximum power is limited by the available current at low load impedances and by the available voltage swing at high impedances. If your headphones are low impedance, you should operate the amplifier at ±15V with 160mA in the second stage, and if they are high impedance, use a ±24V supply with 100mA. Since high impedance headphones require less power than the low impedance ones, the ±15V operation will probably give more than enough power for ear-shattering SPL over the whole impedance range.

ASSEMBLY

Figure 2 shows the stuffing guide for the hybrid tube/MOSFET headphone amplifier. Start the assembly by installing the solder pins, jumpers, and then

taking into consideration the rest of the electronics.

And this is not the end of the story. The Walkman-type devices are usually operating with two 1.5V batteries, for a supply of 3V total. Assuming that the audio amp would be able to work "rail-to-rail," the equivalent audio signal would be 3/2.82 = 1.06V RMS, and you could generate a maximum current of:

$$I = 1.06/40\Omega = 26.5\text{mA}$$

This would give a maximum power of:

$$P = U \times I = 1.06\text{V} \times 26.5\text{mA} = 28.1\text{mW}$$

And SPL difference would be:

$$\text{SPL.diff.} = 10 \log 28.1 = 14.5\text{dB}$$

And the maximum SPL: 100dB +14.5dB = 114.5dB, which is actually "only" 10dB less than the maximum. However, remember that in most cases the audio signal would be less than the one calculated, or the amp would already be clipping at a lower value. In a 32Ω headphone with 100dB SPL for 1mW input, the maximum power would be 35mW and the maximum SPL would be 115dB!

Real-life ICs, made specifically for low-voltage operation, will usually deliver less than this. Look up the National LM4911, which is a stereo headphone amp—it delivers 25mW into 32Ω at 1% THD from a 3V battery (12mW from 2.4V). This means that just a bit over 80% of the battery voltage is "converted" into audio! I bet most of the Walkman-type devices don't deliver much more than 10–15mW of "clean" audio!

What would happen if you connected a 600Ω headphone to this amp? The maximum current would be: 1.06/600 = 1.77mA, the maximum power: 1.06V × 1.77mA = 1.88mW. The SPL difference is:

10 log 1.88 = 2.74dB and the maximum SPL is 98dB + 2.74dB = 100.74dB.

Obviously, 600Ω headphones are less suited for this kind of application. For amps with low supply voltage, you need to use low-impedance headphones, assuming, of course, that the amp can deliver the necessary current.

Headphone amps operating from ±9V batteries fare much better in terms of maximum power. Assuming an 80% ratio between battery voltage and audio signal, such an amp could deliver over 600mW into a 40Ω headphone. Of course, the battery would also need to deliver the necessary current (over 120mA!), and the question is how long it would be able to do that? The same amp would manage only about 40mW into a 600Ω headphone, so even an amp working with ± 9V power supply cannot cover the whole impedance range.

In addition to the problem of available power, most of the low-voltage, battery-operated headphone amps are working with very low bias current to save battery life. This means in most cases Class-B operation. Now it's well known that Class-B is far from ideal in terms of sound quality due to crossover distortion, but there is really not much you can do when the amp must be portable and operate from low-voltage batteries. Still, there are many people listening to portable devices, so it cannot be all that bad!—**EB** output stabilize to 0V. Alternatively, you can leave the heater on all the time or in a stand-by mode with—say—4V, in which case you can apply the full heater voltage and the supply voltage simultaneously.

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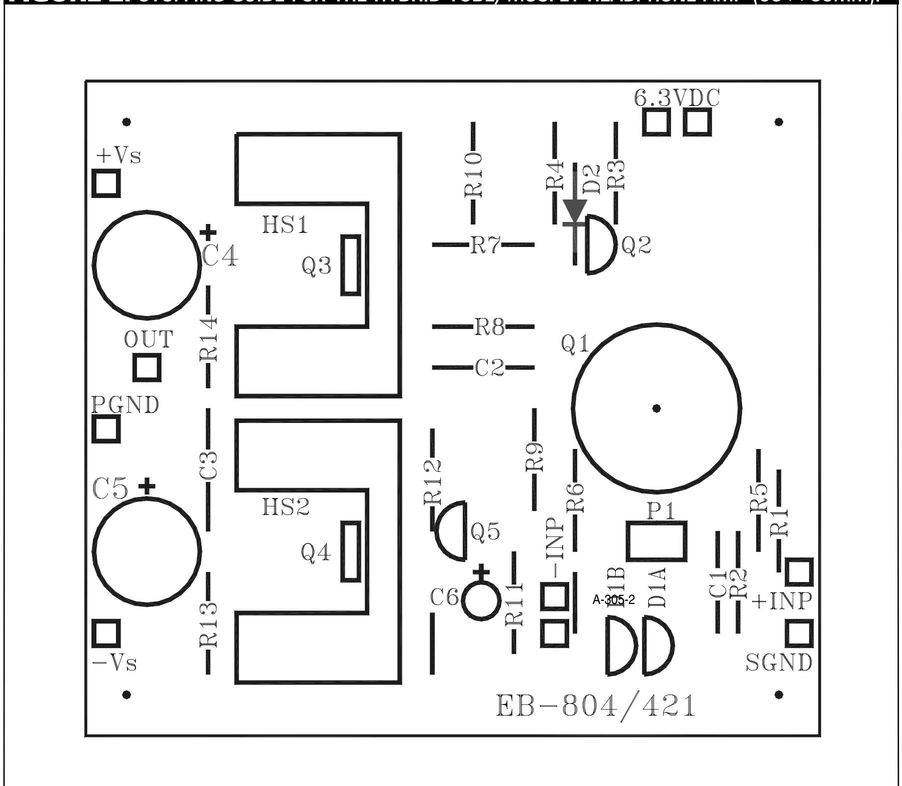

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Next install Q2, Q5 and diodes D1 (A/B). Mount Q3 and Q4 on the heatsinks with insulator and install them on the board. Make sure the MOSFETs are properly tightened to the heatsink. Then install the tube socket and all the capacitors, with C4 and C5 being the last ones. Finally, plug the tube into the socket.

SETUP PROCEDURE

If possible, test each amplifier separately before installing it in the chassis. This simplifies measurements, adjustments, and, if necessary, component changes. If you have access to a scope, connect it to the output of the amp and check whether radio frequency (RF) oscillations are present. If you have a complete audio instrumentation in your workshop, perform the usual gain, frequency response, noise, total harmonic distortion (THD), and intermodulation distortion (IM) measurements.

Connect the +INP and the −INP to SGND. Apply the appropriate supply voltage (±15V or ±24V) and the 6.3V DC heater voltage to the amplifier. Connect a digital voltmeter (DVM) across R13 and check the voltage drop. It should be 0.62–0.65V. This sets the current to approximately 100mA or 160mA in the second stage, depending on the value of R13.

Let the amp run for about 20 minutes before you adjust the offset. Connect the DVM to the output of the amplifier and set the offset voltage to 0V with P1. This completes the DC adjustments.

The EB-804/421 kit is available from North American representative Larry Black; for US/Canadian customers, see www.audiokits.com. Other countries can order the kit directly from Borbely Audio, www.borbelyaudio.com. ***aX***

A note on intellectual property rights. DIY amateurs may build this circuit for their own use; however, commercial use or duplication in any form is not authorized. Many of my published circuits have been copied without my permission—not just by Far-East, but also by reputable European and US companies, and used in commercial products. This is against the international intellectual property rights.—**EB**

PARTS LIST
EB-804/421

Resistors, Trimpot

R1,R7,R12	100R
R2	1MEG
R3,R4	499R
R5,R6	47R5
R8	10k, RN60
R9	1k1, RN60
R10	33R, RN60 (±24V)
	7R5, RN60 (±15V)
R11	10k
R13	6R8, RN60 (±24V)
	3R9, RN60 (±15V)
R14	47R5
P1	100R Copal

All resistors non-magnetic Dale CMF 135 and RN 60.

Capacitors

C1 (Optional)	100pF, 160V PS
C2	10pF, 160V PS
C3	1000pF, 160V PS
C4,C5	220µF, 35V
C6	100µF, 25V ROE EKO

Tube, Semiconductors

Q1	ECC86/6GM8, ECC88/6DJ8, E88CC/6922, 6H23Π-EB
Q2	2SA872
Q3	2SJ79 INS.
Q4	2SK216 INS.
Q5	2SC1775
D1	J508 or E-202 (two in parallel)
D2	1N4148

Miscellaneous

PCB	EB-804/421
HS1,HS2	SK76-37.5

12 × 1mm solder pins
4 × M3 screws, nuts
2 × 9-pin PCB tube socket
Mounting hardware

TLAH: Transmission Line Array Horn

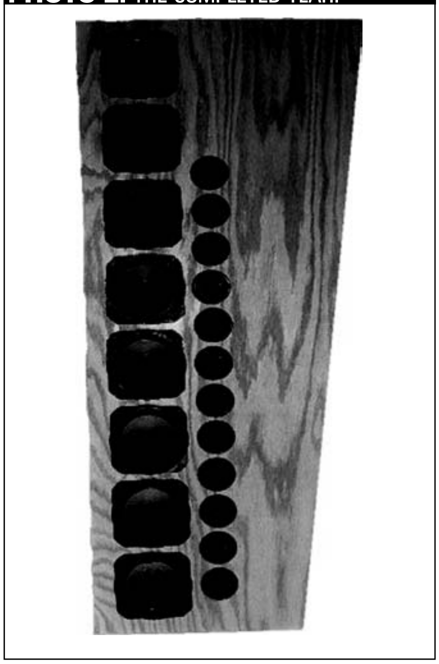
By Bill Fitzmaurice

Back

A dozen tweeters and eight mid-woofers complement the author's horn sub—for a total package that delivers a really big sound.

PHOTO 1: THE TLAH PAIR IN MY LIVING ROOM. IT APPEARS QUITE NARROW, BUT 3′ FROM THE REAR WALL THE ROOM EXPANDS TO THE LEFT ANOTHER 7′.

PHOTO 2: THE COMPLETED TLAH.



For quite some time now, many of you have been asking me to design another high-efficiency speaker for hi-fi/home theater use, and I've promised to do so as soon as my schedule allowed. I don't know when that would have been except for the good folks at Parts Express, who offered a deal that I couldn't pass up. Last winter they sold me wide-range 4″ mid-woofers at the silly price of only 87 cents apiece. I didn't know what I was going to do with them, but that price was too good to believe, so I ordered 16 of them. The shipping was going to cost better than half what the drivers would, so to fatten the order a bit I also ordered two dozen closeout ⅜″ dome tweeters, which at 32 cents apiece cost less than the price of a six-pack.

The two boxes full of drivers sat in my basement through the winter and spring, and even after I opened my workshop for the season they remained uncrated, as I still needed to do the final revisions on my Tuba 18 sub (*aX* Jan. '05). When completed, the Tuba 18 prototype ended up in my living room, and it wasn't there a full day before I realized that the high-sensitivity folded-horn sub was making the rest of my system sound, well, rather small. My MTM satellites just weren't in the same league as the Tuba. I knew that I needed to come up with a high-sensitivity replacement for them, and then my thoughts returned to the 40 drivers sitting idle in the basement. The answer to what to do with them was obvious: line array.

LINE 'EM UP

Line arrays are a natural for high sensitivity, since you achieve an additional 6dB of voltage sensitivity over a single driver with every doubling of the number of drivers. I figured that eight Chinese generic mid-woofers per side—rated perhaps optimistically at 87dB/2.83V—should deliver at least 95dB. That would be at 4Ω, but my amp would not find that a problem. The Japanese tweeters claimed a similar sensitivity, so surely a dozen of them per side (rated at 6Ω) would prove formidable. I could wire them as four parallel sets of three tweeters apiece in series, for a 4.5Ω load.

The mid-woofers would work fine in a sealed or vented box, but because the line array demanded an oblong cabinet shape, another option seemed logical: transmission line. Normally a TL is energized by a driver or two toward one end of the pipe, and I had no way of knowing whether it would work with drivers along the full length of the line. But if it didn't I could always seal the end of the box or toss in a duct. The driver Qts of 0.77 and Fs of 105Hz meshed pretty well with the idea of a 100Hz TL, because the best results in a TL are usually obtained when the driver Fs is close to the line quarter-wavelength frequency and Qts is as close as possible to 1.0 (something to keep in mind when shopping for your own bargain-basement specials).

A quarter-wave tube for 100Hz would measure 34″ long, sufficient to house the eight fours, though I added a couple of extra inches for good measure. I wanted to use a tapered line and to tilt the cabinet downward somewhat, because my living room demanded a high mounting position. These two requirements meshed by placing the larger end of the line at the top. This also meant that the line exit would be nearly a quarter-wavelength from the ceiling at 100Hz, and if I wasn't careful the resulting Allison effect could negate a lot of its bass output.

FIGURE 1: CABINET BOTTOM (TEMPORARY).

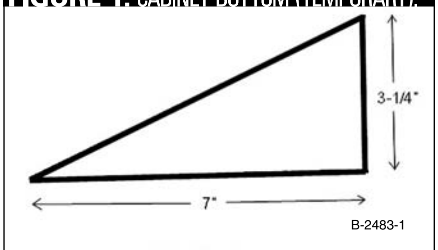
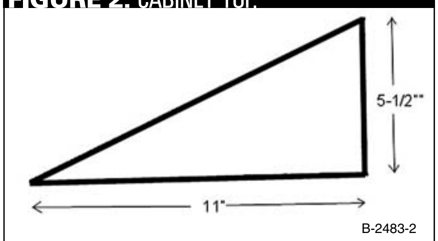


FIGURE 2: CABINET TOP.



PLACEMENT

Log on to almost any Internet audio forum and you'll find an inordinate number of posts dedicated to proper speaker placement. There are even computer programs devoted to curing placement woes, though a basic understanding of quarter-wavelength cancellation and half-wavelength augmentation is all you really require. My intended location put the line exit 40″ from the ceiling, which would invite a cancellation node just below the Fp of the TL. I decided to use corner placement to give a boost in the bottom end to help counter this. To enhance the corner placement aspect, and actually introduce a bit of horn loading to the vent output, I made the cabinets triangular.

This also put the cabinets in a naturally toed-in position, while seamlessly melding the baffle with the room walls to eliminate the baffle diffraction step. The resulting conglomeration is a hybrid of three different speaker types; I call it the TLAH, for Transmission Line Array Horn (*Photo 1*).

There are a few points to remember about line arrays. The first is that you must keep the drivers tightly packed, especially the tweeters. A line array sums the outputs of the point sources within it into a single coherent wave front, but only when the distance between the various radiating planes is less than a

FIGURE 3: TLAH BOTTOM VIEW.

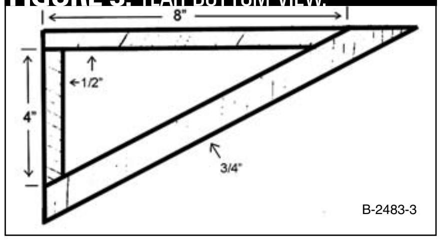
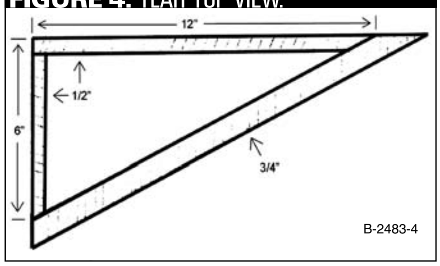


FIGURE 4: TLAH TOP VIEW.



wavelength. Even when the frames are touching, this requirement can be problematic at, say, 13kHz, where a wavelength is only an inch, and typical dome tweeter driver frames are 2″ wide. For this reason, continuous ribbon planars are preferred for line arrays.

The closeout tweeters I used were flangeless, originally intended for coax mounting on an autosound woofer, resulting in less than a 2″ gap between their domes. Closer would have been better, but they proved adequate. When choosing your tweeters, find some that will allow minimal spacing (*Photo 2*). The same caveat applies to the mid-woofers, though you'll find that many in the applicable 3–4″ size have semi-square "pincushion" frames like my closeouts, which makes close placement easy. If you get mid-woofers with round frames, it would be best if they practically touch each other.

As far as the 6dB increase in voltage sensitivity with doubling of drivers is concerned, that aspect is frequency limited; as you add drivers, the increase in sensitivity is nonlinear and there is a rolloff at the upper end. This is somewhat ameliorated by another facet of the line array, which is the near-field/far-field transition. Within the near-field, SPL drops at 3dB per doubling of distance from the source, going to 6dB when the far-field transition frequency has been reached. That frequency, and the distance from the source at which the transition occurs, is dependent on the height of the array according to the formula $r = l^2 f/700$, where l is the array height and r and l are expressed in meters.

In the case of the TLAH, with a tweeter array height of 0.6 meters, the transition at 12kHz takes place about 6 meters out, while the transition at 1kHz for the 0.8 meter high mid-woofer array occurs at less than a meter out. Therefore, at the typical listening position, the high-frequency rolloff is offset to some extent by the near-field condition of those same high frequencies.

With the tweeter and mid-woofer lines side-by-each, there is potential for comb filtering on the horizontal plane near the crossover frequency where the outputs overlap in frequency. Minimize that with a 3rd-order or higher crossover, along with placement of the two lines physically as tight as is practical. Stuffing the line with polyfill is also required to eliminate resonance. While there are formulas on how much stuffing is required, I didn't use them. I found that using enough polyester pillow filling to thoroughly stuff the line without compressing the material worked just fine.

CONSTRUCTION

Construction of the TLAH is relatively simple, though the acute angles at the cabinet edges are a bit tricky. I built the prototype with a baffle made of oak-veneer ¾″ plywood, with the remaining parts made of ½″ plywood. You could

PHOTO 3: ADJUSTING THE SAW BLADE TILT TO THE CABINET ANGLE.

PHOTO 4: BACK ROUTING JIG, PART 1.

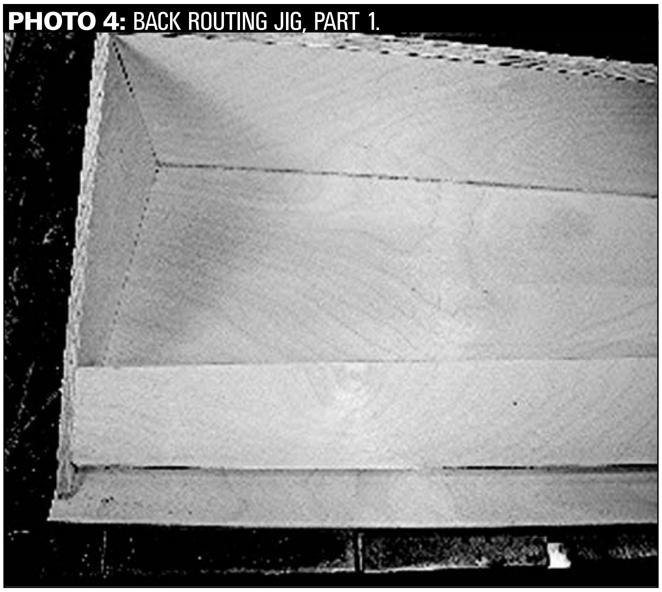
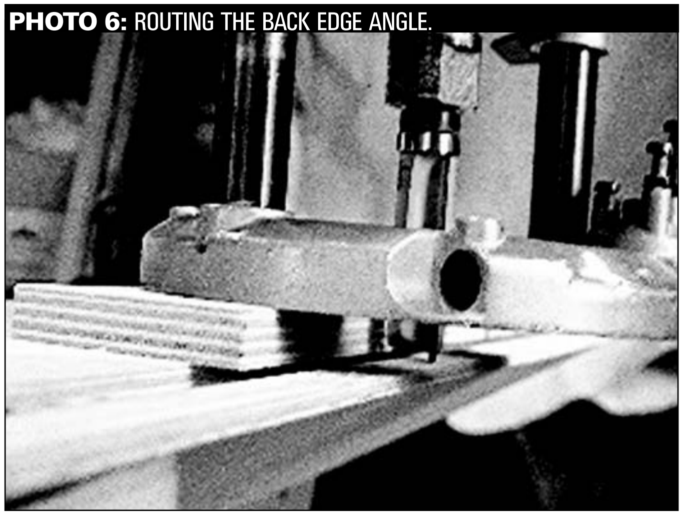


PHOTO 5: BACK ROUTING JIG, PART 2.

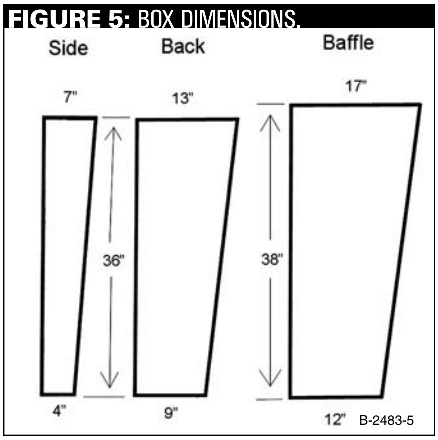
PHOTO 6: ROUTING THE BACK EDGE ANGLE.



also use either particleboard or MDF. As usual, you should glue and screw joints, but there is very little acoustic pressure inside this box, so "built like a tank" construction techniques are not warranted.

These boxes are mirror-imaged twins, so it's best to build them simultaneously to keep track of the left and right halves of the pair. The first step is to cut out the tops and bottoms (*Figs. 1* and *2*). You can see how they interface with the other parts in *Figs. 3* and *4*. The bottoms are temporary construction aids only, so be sure not to use any adhesive on them. Next cut out the

FIGURE 5: BOX DIMENSIONS.

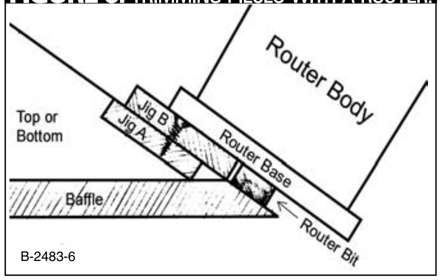


backs and sides (*Fig. 5*), leaving them oversized to allow plenty of selvages for trimming. Each set is jointed to each other first and then to their respective top and bottom (no glue on the bottom!).

The next step is to trim the back and side to the same angle as the top and bottom. It's easy to just draw a line on the inside of the side from the top to the bottom, set a table-saw blade to the correct angle by matching it to the angle of the top and bottom (*Photo 3*), and cut away. However, you can't use the same procedure on the back, because the angle is too severe for the blade to adjust to it. It is possible to duplicate the procedure used on the side by using a very high (6″ or more) rip fence and running the assembly across the saw with the back against the fence. But an easy alternative is to use the router.

Screw a jig of scrap plywood from the top to the bottom, with the plywood just touching the back. Add a second piece of plywood over the first, which will serve as a platform for the router base to rest on (*Photos 4, 5,* and *Fig. 6*). Now run the router across the platform to trim the back edge to the proper angle,

FIGURE 6: TRIMMING PIECES WITH A ROUTER.

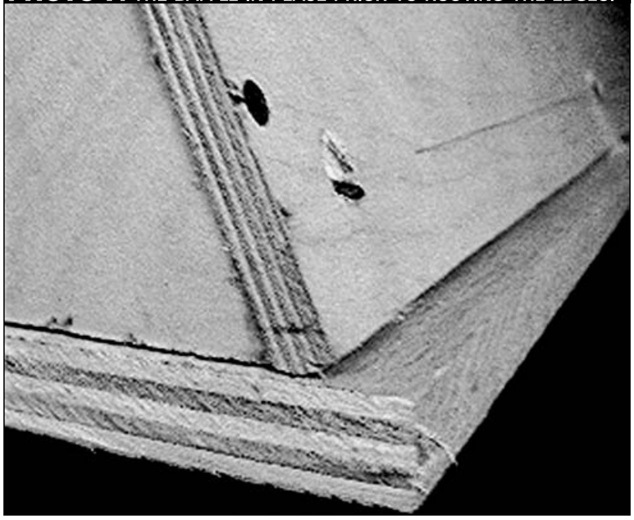


doing so with multiple passes in small increments (*Photo 6*). To be sure that the work piece is stable during the routing procedure, clamp or screw it to the work bench; if you use screws, fill the holes in the cabinet after you're done.

Cut out the baffles according to *Fig. 5*. Note that they are a couple of inches longer than the rest of the box, to allow an overhang of an inch at either end. Temporarily attach them to the assemblies, using countersunk drywall screws driven from the rear (*Photo 7*).

Trim the baffle edges to the correct angles, this time using the platform and router technique on both edges; cutting the baffle with the front side down on the table saw would chip the face. If you don't have a router, you must mark

PHOTO 7: THE BAFFLE IN PLACE PRIOR TO ROUTING THE EDGES.



the baffle where the cut lines go and remove it from the assembly so that you can saw it with the face up.

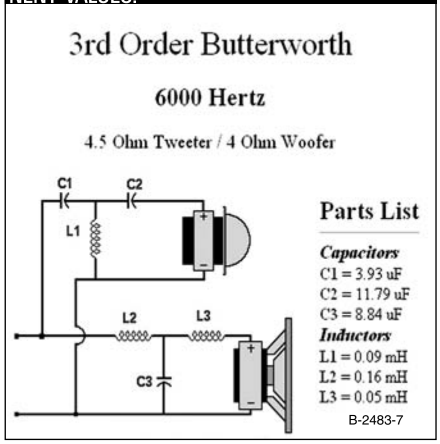
With so many driver holes to cut in the baffle, you probably should invest in a pair of hole-saws. Draw the centerline of the mid-woofers 3½″ from the side edge, and space the drivers as close as practical to each other. Start marking the mid-woofer locations at the top of the box, so that any leftover baffle area is at the bottom. Mark the centerline for the tweeters as close as you can to the woofers, and the tweeters as close to each other as possible, with the lowermost tweeter horizontally on-center with the lowermost mid-woofer. After cutting all the holes, sand the face smooth and apply your finish; add veneer and finish it at this time if you aren't using pre-veneered plywood.

DRIVER INSTALLATION

When the finish is dry, mount the drivers. With small pincushion-style mid-woofers there could be issues with gasketing the flange; I used silicone caulk to seal them from the rear. Likewise, mounting frameless tweeters calls for a special technique. I cut the mounting holes ⅛″ oversize, wrapped foam weather-stripping around the tweeters to hold them in place, and then again glued them from the rear with silicone caulk. You could also use masking tape to hold the tweeters in position while gluing. The crossover components are also glued to the inside of the baffle, with either hot-melt glue or silicone caulk (*Photo 8*).

As previously noted, you should use at least a 3rd-order crossover. I use an interactive website to find basic crossover component values; the address is http://www.lalena.com/audio/calculator/xover/. For my drivers the site gave the values shown in *Fig. 7*. After initial testing, the frequency response was quite adequate, so I didn't alter the circuit. You must consider the driver wiring scheme before buying your drivers.

FIGURE 7: OBTAINING CROSSOVER COMPONENT VALUES.



With eight 8Ω mid-woofers, I wired them as series pairs, with the four pairs then paralleled for a 4Ω load. If your amp demands an 8Ω load, you may use either 4Ω or 16Ω drivers, or you may use nine 8Ω drivers wired as series/parallel "triplets," making the box a few inches longer to accommodate the extra driver. Keep in mind that with the high system sensitivity even 4Ω load won't pull a lot of current at normal listening levels. My 6Ω tweeters allowed me to use 12 drivers wired as four paralleled sets of three, with each set of three series wired, for a 4.5Ω load. You'll need to figure in advance how to wire your tweeters and how many to use to arrive at a load very close to that of the mid-woofers. If that results in a projected SPL higher than that of the mid-woofers, all the better, for reasons to be seen later.

After you have installed and wired the drivers, drill a hole through the cabinet top for the speaker wire, or alternately install a set of binding posts; in either case make sure the top is sealed airtight. Remove the temporary bottom from the box assembly and weather-strip the box edges where the baffle will mate; you don't want to glue the baffle in place in case you ever need to replace any drivers. Fill the box with sufficient polyester pillow stuffing to thoroughly stuff the box, but not so much that it requires a lot of force to compress it when the baffle is placed over the box. Reattach the baffle with screws and attach wall hanging hardware to the back as required. *C'est fini*.

As far as the mounting scheme goes, you aren't restricted to the suggested corner placement. For the reason why, look at *Fig. 8*, which shows a very respectable response from 100Hz. I measured this with the cabinet freestanding outdoors, so you could use the box in a freestanding mode indoors with some sort of stand to raise it to proper height. That height is determined by the tweeters, keeping in mind that the array has a very narrow vertical dispersion. I recommend that you have the listening position on-axis with at least the topmost or bottommost tweeter in the array—somewhere in the middle is even better. If you must mount the boxes high as I did, the downward tilt of the box helps.

FIGURE 8: SPL CHART.

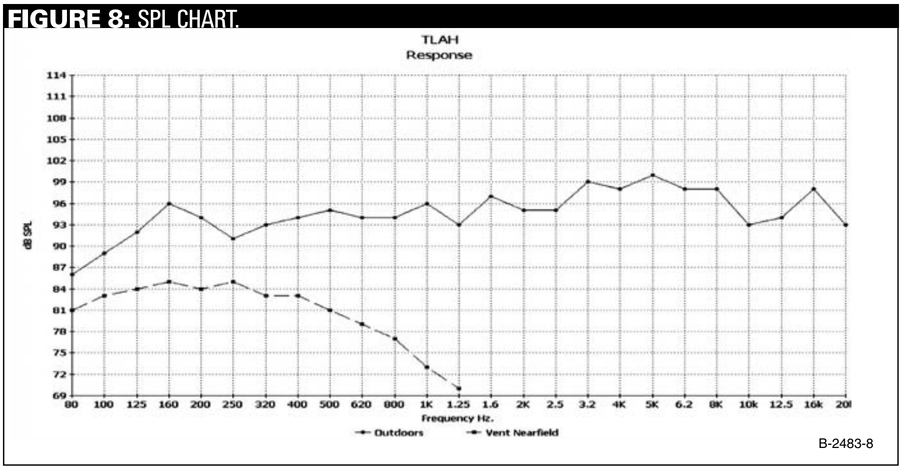
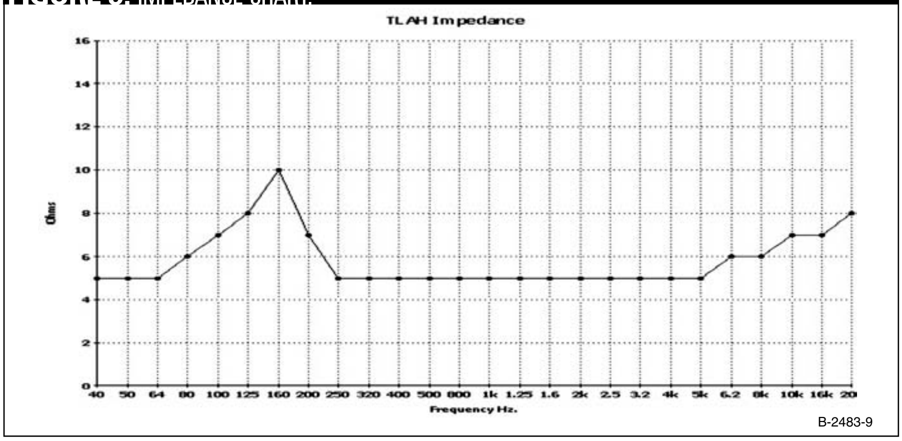


FIGURE 9: IMPEDANCE CHART.



MEASUREMENTS

Figure 8 reflects a freestanding outdoor measurement, and does not consider the effects of corner placement, the horn loading of the vent output, and the elimination of the baffle step, all of which contribute considerably to sensitivity. These enhancements also are mostly in the low end, so with corner placement you can expect an average SPL of about 99dB, with a decided tilt in favor of the bass. For that reason, having the base tweeter sensitivity a few dB higher than that of the mid-woofers is not a bad idea.

The final caveat applies to the vent output, which gives a goodly percentage of the bass. You need to avoid Allison effect cancellation from both the ceiling and floor reflections, which at 100Hz would occur at a distance of about 2.8′. If you cross the box over at the minimum recommended 80Hz, you must be sure that both the ceiling and floor are more than 3.5′ from the vent, which means your ceiling will need to be over 7′ high.

Speaking of the vent, the lower trace on the SPL chart shows the vent output, scaled downward for clarity. The vent response goes quite a bit higher in frequency than would be seen from a normally loaded TL, which is to be expected with the lower drivers so close to the vent. This begs the question about whether the box qualifies as a true TL, and that is answered by *Fig. 9*.

STUFFING

One of the hallmarks of a TL is an impedance sweep with a single low-frequency bump, in the same fashion as a sealed box. You can see that there is indeed a single bump at 160Hz, with exceedingly flat impedance across the rest of the bandwidth that, along with the very flat response, also shows that the stuffing density is adequate. To be sure that your TL is adequately stuffed, run an impedance sweep looking for additional peaks. With adequate stuffing, high-frequency resonant nodes are eliminated and only the single peak remains.

While *Fig. 9* confirms that this box is a real TL, the frequency of the response bump is about a half-octave higher than the line quarter-wavelength frequency and driver Fs, where ideally it would be much closer. I attribute this to the fact that the lower drivers are so close to the line exit that they really aren't being loaded by it, causing an upward shift in the system resonant frequency. This is confirmed by the falling response of the line exit below 160Hz, but as that shortcoming of the design is countered in use by boundary loading, it isn't problematic. I found that my TLAHs integrate just fine with my sub with an 80Hz crossover.

SPL and impedance charts aside, the main question remains: How do they sound? The best term I can think of is "Big." Really big. Yes, their sensitivity allows them to keep up with my Tuba 18 sub, and they will play cleanly to levels well beyond what the WAF (wife acceptance factor) will allow, but moreover they just sound different.

LISTENING

To my ear they sound like a very good set of horns. I believe that the physically large wave front produced by the line is akin to the large wave front produced by the mouth of a horn. I wouldn't hesitate to recommend the design to anyone who's wanted the sound that until now you really needed a horn to get.

The other night my spouse and I watched "The Lord of The Rings: Return of the King" at peak levels over 115dB, and it sounded so good that even she didn't complain. Yet, they take up only a quarter of the space that a 100Hz horn would, and there's also the factor of cost. Mine cost a total of about 60 bucks complete for the drivers, cabinets, and crossovers. That's $60 for the pair. Your cost will vary, of course, depending on what is available on close-out, but even at ten times the price these speakers would still be a bargain. ***aX***

PHOTO 8: REAR VIEW OF THE BAFFLE WITH DRIVERS AND CROSSOVER IN PLACE.

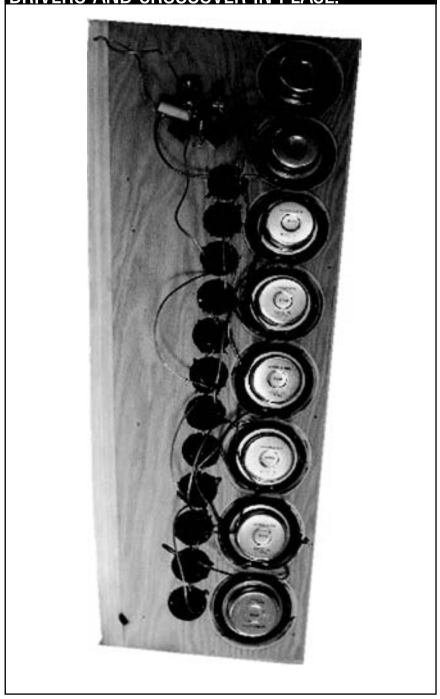
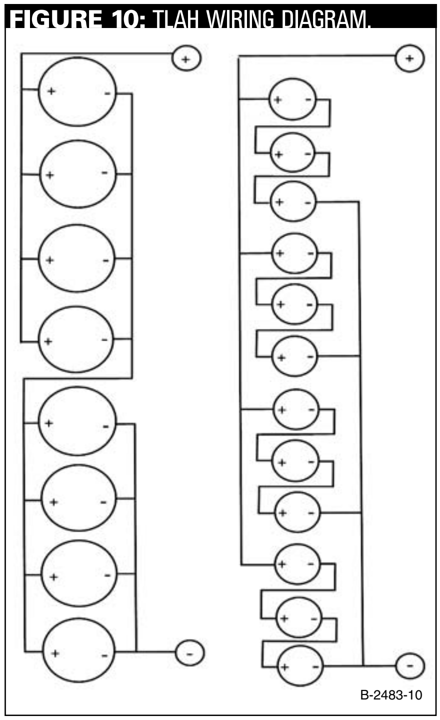


FIGURE 10: TLAH WIRING DIAGRAM.



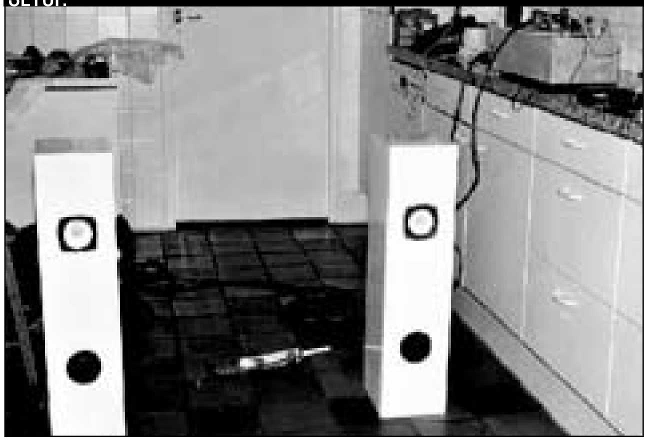
A Simple Audio System, Part 3: Class A Transistor Amp

By Aren van Waarde

Back

In this series final, the author presents a solid-state alternative to the single-ended tube amp and matching pair of speakers presented previously.

PHOTO 1: BUILDING A SOLID-STATE VERSION OF THIS TUBE-BASED SETUP.



The idea for this project came about when I discovered an interesting website on the Internet: the "RED Free Circuits Page[1]." My curiosity was aroused by that name. In what sense can circuits be red?

It turned out that "RED" is an acronym for "Ricerche Elettroniche Dellepiane," or, in English "Dellepiane Electronics Research." Flavio Dellepiane, the maker of the site, is a friendly Italian from Campomorone, 20km north of Genoa. He is an electronics hobbyist who has been building audio equipment for at least 30 years. His well-documented site includes almost 100 projects. Some of his designs have been published in the British magazine *Electronics World* (formerly *Wireless World*) and have been recognized with honors.

A RED CIRCUIT

RED Circuit 80 grabbed my attention. The header of the web page read: "3–5W Class A Amplifier. Behaves like a valve amplifier. Simple circuitry. No crossover distortion." The text continues: "In the old valve days, most commercial audio amplifiers in compact record play-

FIGURE 1: AMPLIFIER SCHEMATIC (BOTH CHANNELS ARE IDENTICAL).

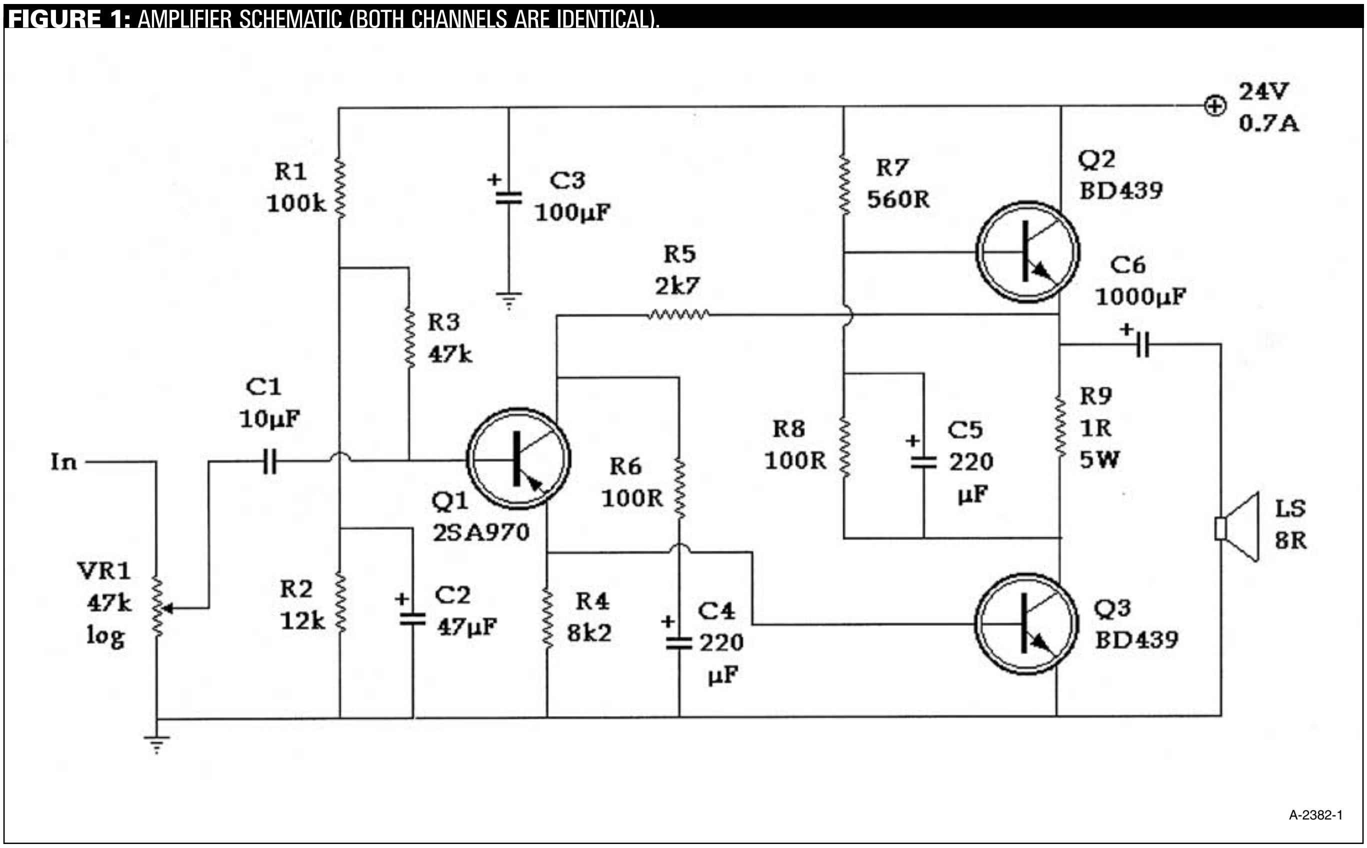
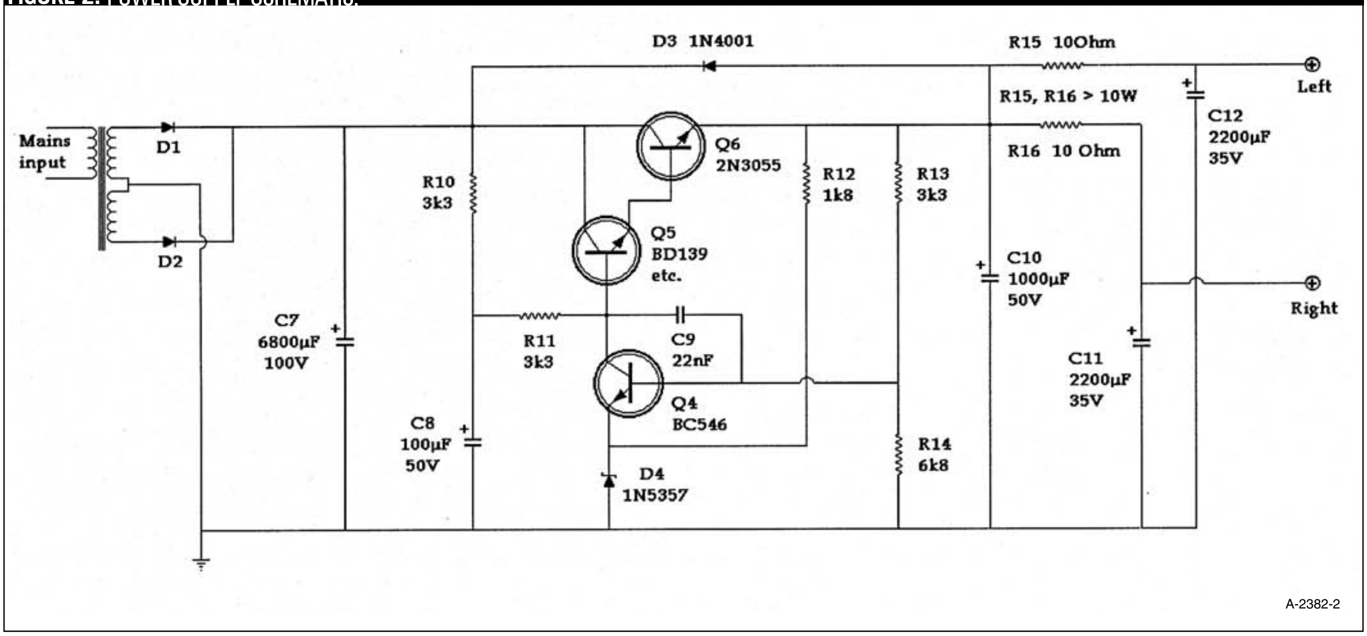


FIGURE 2: POWER-SUPPLY SCHEMATIC.



ers used a single-valve topology. The circuit usually employed a compound valve, e.g., a triode-pentode ECL86. This solid-state Class A circuit is capable of providing a sound comparable to those valve amplifiers, delivering more output power, less THD, higher input sensitivity, and better linearity."

This text was sufficient to get me hooked! Flavio claims that a solid-state amp (with just three bipolar transistors per channel, *Fig. 1*) can produce a sound similar to the valve amp that was the subject of Part 1. Unlikely though this may be, it is not impossible. The Zen series of amplifiers, designed by Nelson Pass, can, in fact, produce tube-like sound. I am the proud owner of a Zen Revisited amplifier (Zen with a pi filter in the power supply and a small amount of negative feedback) that is about five years old and sounds excellent.

Thus, I decided to heat my soldering iron. Initially, I made a single-channel prototype using Japanese transistors from the junkbox: one 2SA970 and two 2SC2581s. Although Flavio used different semiconductors (one BC560C and two BD439s), my first prototype worked exactly as the amplifier on the RED Circuit page: current draw 0.67A (should be 0.7) and voltage at C6 positive 13.78V (should be 13V).

For a second prototype, I used 2SC3263s instead of 2SC2581s, but that turned out to be a poor choice. Because of the low h_{FE} of the 2SC3263s, I could not obtain the right circuit voltages and currents, even after tweaking the values of resistors R2 and R7 (more later). I made my third prototype from a single 2SA970 and two BD439s, which worked out well (current draw 0.7A, voltage at C6 positive 13.7V).

Although BD439 transistors are smaller and have a lower f_T than 2SC2581s, I preferred the sound of prototype 3 over that of prototype 1. So I used another 2SA970 and two BD439s for prototype 4. Now I had two identical channels that I could use as a stereo amplifier. I used 2SA970s instead of BC560Cs because I had them in my junkbox. Also, rumor has it that the 2SA970 is good for audio.

POWER SUPPLY

A stereo pair of RED Circuit 80s draws a continuous current of 1.4A from a +24V DC power supply. Flavio's site does not give any suggestions for a PSU. I thought it might be worthwhile to use a voltage regulator, not because any change in current draw is to be expected, but because of the hum suppression which a voltage regulator may provide.

A simple but effective three-transistor voltage regulator is presented on the Class A amplifier site[2]. This power supply (*Fig. 2*) is a classic circuit, which may have its origin in a Telefunken (or ITT) application note from the 1960s. Foldback current-limiting or short-circuit protection is not included, but also not required for this application.

I built my own PSU with semiconductors from the junkbox: a Philips BC546, a TFK325 (Eastern European medium power transistor), and an ancient 2N3055 from RCA (mounted on a fairly big heatsink). It showed good regulation. The output voltage dropped only 0.07V (from 24.12V to 24.05V) when the current draw was increased from zero to 1.4A (during this test, the channel-decoupling RC networks had not yet been added, and a 15V instead of a 20V zener diode was built into the power supply). The power transformer I used was a nice potted item, cannibalized from an old TEAC (2 × 40W) amplifier.

TABLE 1

PARTS LIST (AMPLIFIER)

PART NO.	VALUE
Q1	2SA970 or BC560C
Q2, Q3	BD439
R1	100k
R2	12k (see text)
R3	47k
R4	8k2
R5	2k7
R6, R8	100Ω
R7	560Ω
R9	1Ω 5W
VR1	Potentiometer 47k, audio taper
C1	10µF 100V foil cap (MKP or MKT)
C2	47µF 35/40V electrolytic
C3	100µF 35/40V electrolytic
C4, C5	220µF 35/40V electrolytic
C6	1000µF 35/40V electrolytic

Parts for a single channel; both channels are identical. All resistors ½W 5% unless otherwise noted.

When fed by this simple PSU, the stereo amp is completely quiet. Even with my ears directly in front of the speakers, I heard no hum or noise.

THE SOUND

Since I am a tube enthusiast, I was biased against this fleapower "sand amp," expecting it to sound grainy and (at best) two-dimensional. But I changed my view. Strange though it may seem, this minimalist design is one of the few transistor amplifiers that tube lovers will immediately accept. It is a great performer for a few bucks, with a clear family relationship to the Zen series of amplifiers—or the Linsley-Hood class A from the late 1960s—but with less power. Yet, there is more than adequate power for my *Petite* speakers (in fact, enough to upset the neighbors).

Strings and percussion instruments sound sweet, with beautiful tonal colors. However, this sweetness appears not to be acquired by the masking of small details. Right now, I am listening to twentieth-century classical music by the Dutch composer Alphons Diepenbrock (recorded on Chandos 8821 and 8878), who wrote complex works in a late romantic style, with a very dense orchestration. On many audio systems, his music is amalgamated into an amorphous mass, making its appreciation difficult.

Not so with Flavio's amp. Individual instruments and voices stand out from a wide and deep soundstage. Yet, there is a sense of wholeness, a harmonious unity to the sound.

Direct comparisons are difficult to make (I cannot do immediate A/B switching), but I imagine hearing the following differences between the transistor and the valve amp.

1. Bass of the transistor amp is superior to that of the valve amp; it's more tightly controlled and more extended.
2. The intelligibility of voices and of vocal lines seems to be slightly better on the valve amp (although voices sound natural on both amplifiers, without artificial sibilants, crossover hardness, or electronic glare).
3. Both amps produce a lively and three-dimensional sound, but the soundstage of the transistor amp seems to be more "forward" than that of the valve amp. Soloists seem to be in front of, or between, the speakers, whereas with the valve amp their apparent positions are behind and between the speakers. I don't have an explanation for this phenomenon.
4. Instrument sizes are not reproduced in the same way. On some recordings of a solo harpsichord, the instrument seems bigger on the transistor amp than on the tube amp. Once more, I cannot explain the reason for this.

Considering the overall performance, I seem to prefer (slightly) the transistor amp over the PCL82 SE amplifier of Part 1. Don't set too high a value to this preference. The tube amp might have won if I had used more expensive output transformers to build it. Yet, this preference indicates the excellent subjective qualities of Flavio's cheapskate amp. For its modest cost, it is a real winner.

CONSTRUCTION NOTES

Since both the amplifier and the power-supply circuits are very simple, I used the "dead bug" style to construct them. Using an unetched piece of printed-circuit board as a groundplane, I glued all components that were not mounted on heatsinks to this groundplane with

TABLE 2

PARTS LIST (POWER SUPPLY)

PART NO.	VALUE
D1, D2	Rectifier diode 100V 2A or greater#
D3	1N4001 (50V 1A)
D4	1N5357 (20V zener)
Q4	BC546
Q5	BD139 or equivalent*
Q6	2N3055
R10,R11,R13	3k3
R12	1k8
R14	6k8
R15, R16	10Ω wire-wound (10W or greater)
C7	6800µF 100V electrolytic
C8	100µF 50V electrolytic
C9	22nF 63V foil
C10	1000µF 50V electrolytic
C11, C12	2200µF 35V electrolytic
Tr	Power transformer, secondary 60V CT$

I cannibalized my diodes from a damaged NAD amplifier. They are marked S6K20 (MUR860 look-alikes).
* I used a medium-power transistor from Eastern Europe, marked TFK325.
$ I employed a nice, potted transformer from an old TEAC amplifier. Unloaded voltage of the secondary was 66V AC with center tap.

their wires pointing upward (like dead bugs). I made all circuit connections with the wires of the components themselves, and I used star earthing, with the groundplane of the power supply as the central earth node.

Consider Flavio's notes: "If necessary, R2 (the 12k resistor) can be adjusted to obtain 13V at the C6 positive lead." In three prototypes I have built, such adjustment was not necessary. "Total current draw of the circuit, best measured by inserting a meter between the positive output jack of the power supply and the positive input of the amplifier board, should be 700mA. Adjust R7 (560Ω resistor) to obtain this value if necessary." In my three prototypes, this adjustment was also not required.

"Q2 and Q3 must be mounted on a finned heatsink of 120 × 50 × 25mm (minimum) dimensions." At this point, Flavio is correct, but I would like to add that you should mount the heatsink in such a way that the fins are free and air can flow freely around them. I had used much bigger heatsinks (160 × 80 × 50mm, cannibalized from a vintage Sansui amplifier). Initially I made a beginner's mistake by mounting the PC boards together with their heatsinks inside a cabinet. Although the top of this cabinet had many holes for ventilation, the heatsinks became blazingly hot after just 30 minutes of operation.

Because the cabinet sides were made of plywood, I opted for a very simple solution. Using a variable-speed orbital jig saw, I cut square holes in the sides of the cabinet and moved the PC boards, so that the fins of the heatsinks protruded from the cabinet sides. This made a tremendous difference. The heatsinks now become only lukewarm, even after many hours of operation. Use of the jig saw has spoiled the looks of the amp, but strongly improved its operation!

Flavio suggests that you can include a bass-boost facility. "If this is required, a series network of a 1k5 resistor and a 150nF capacitor can be connected in parallel to R5 (2k7 resistor). The 1k5 resistor end should go to the emitter of Q1 and the 150nF capacitor end to the collector of Q2." I have not included this facility, as I tend to dislike tone controls. Moreover, in combination with the *Petite* speakers (covered in Part 2), I don't see the need for bass boost.

You should mount the pass transistor within the PSU (2N3055) on an adequate heatsink, using heat-conductive paste and insulating material. Power dissipation in this device (in watts) is equal to the voltage drop across the transistor (difference between the input and output voltages) multiplied by 1.4 (current draw in amperes). The voltage drop under load should be at least 3V (preferably more), so the heatsink must handle at least 5W. I used a black (anodized) heatsink of 100 × 120 × 25mm.

CONCLUSION

Experimenting with these very simple valve and transistor circuits was a nice learning experience and great fun. The sonic results have surpassed my modest expectations. Of course, these amps are not perfect. But most recordings are reproduced in a highly enjoyable way. Isn't that what amp building is all about?

Circuit diagrams were drawn with TubePad5, a free program designed by amateur radio operator WD4NKA. ***aX***

TABLE 3

Amplifier measurements
(courtesy of Flavio Dellepiane)

Sensitivity:	560mV input for 5.6W output (8Ω resistive load) 850mV input for 5.6W output (idem, + bass boost)
Frequency response:	40Hz . . . >50kHz (−3dB) 100Hz . . . 20kHz (0dB)
— with bass boost:	+5dB at 100, + 3.9dB at 200, + 2.5dB at 400Hz −1dB at 10kHz, −1dB at 20kHz
THD at 1kHz:	0.3% at 0.5W 0.45% at 1W 1% at 5.6W

Unconditionally stable on capacitive loads

REFERENCES

1. RED Free Circuits page, http://www.redcircuits. com.
2. Class A Amplifier site, http://www.tcaas.btinternet. co.uk.

solid state

Rescuing Fluke's Top Digital Meter Cheaply

By Charles Hansen

Back

Replacing the battery packs in this digital multimeter puts a $400 tool on your bench for about $100.

The Fluke 8050A provides superior performance and measurement capabilities in a low-cost, benchtop instrument. These 4½ digit DMMs (digital multimeters) have the following measurement functions: DC volts, true RMS AC volts, DC amps, true RMS AC amps, ohms, dBm, conductance, and diode test. They also have a relative-reference feature that works with all measurement functions.

These units feature 0.3% basic DC accuracy and 0.5% true RMS accuracy from 20Hz to 20kHz. The –3dB bandwidth extends to 200kHz, with 5% accuracy from 20kHz to 50kHz. They also have overload protection and selectable impedance correction to reference a 0dBm, 1mW level into 50Ω, 75Ω, 93Ω, 110Ω, 125Ω, 135Ω, 150Ω, 250Ω, 300Ω, 500Ω, 600Ω, 800Ω, 900Ω, 1kΩ, and 1.2kΩ. This DMM is ideal for audio work, unlike many "true RMS" meters whose accuracy is limited to the 50–60Hz power line frequency.

The 8050A basic AC powered DMM cost $389 when new. The 8050A-01 ($439 new, *Photo 1*) is AC powered and also has a rechargeable battery pack that allows portable measurements. Nice 8050A-01 meters are available on eBay for well under $100 these days. The battery packs used proprietary GE nicad batteries that are no longer available.

If the BT indicator in the LCD display is always on, even after having had the batteries on extended charge, one or more cells are bad. Once the batteries go, the DMM will not be as accurate as its specifications—even if powered from the AC line—due to the interoperation of the battery charger and internal DC supply. There is also the danger of eventual battery leakage and damage to the meter.

However, it is quite easy to replace the four internal cells with standard C-size 1.2V nicads or NiMH cells. This article will show you how.

PHOTO 1: FLUKE 8050A DMM.

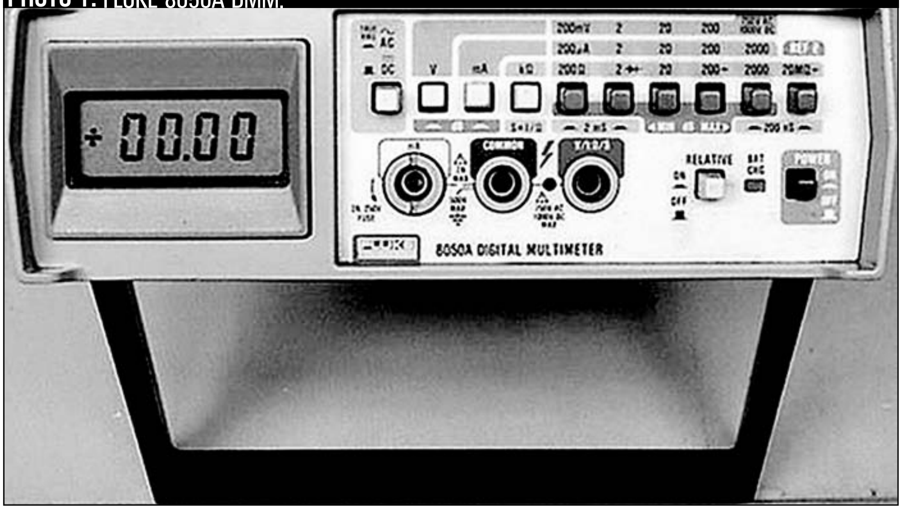


PHOTO 2: OLD BATTERIES REMOVED.

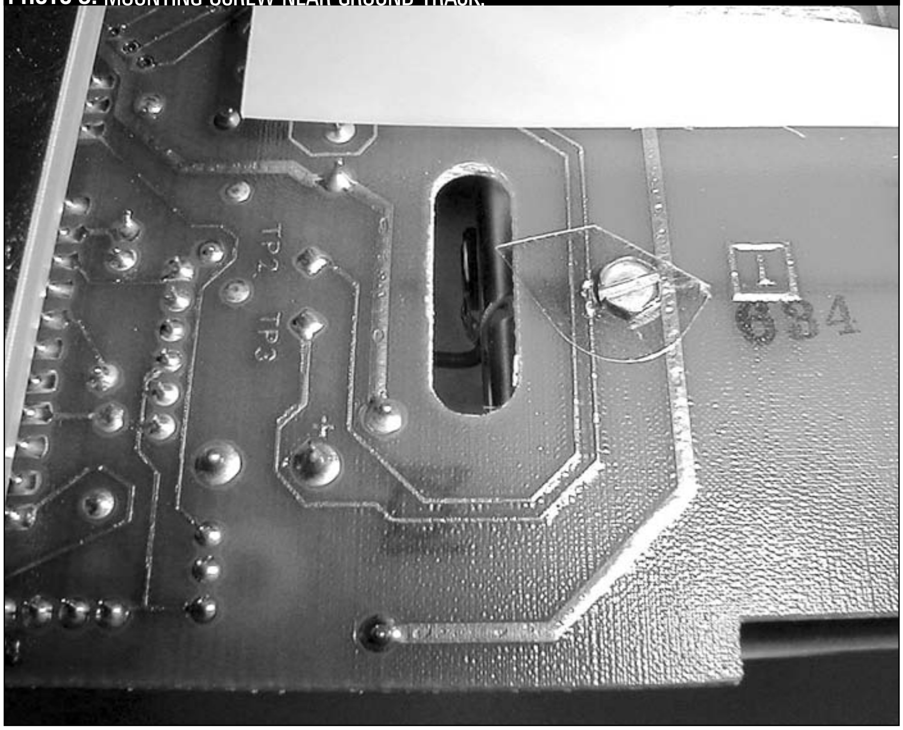
MODIFICATION

You remove the meter cover with one screw on the rear just below the AC receptacle. Place the meter with its display side down, remove the screw, and slide the cover off the meter. The batteries are mounted under two plastic covers on each side of the AC receptacle.

Press the slotted tab on each cover and remove them from the rear of the PC board. Snip the red and black wire from each battery pack, since you will re-use them later (*Photo 2*). The new aluminum battery holders are mounted with 4–40 hardware in holes you drill through the PC board. The right side holder has no PC tracks underneath, so just mark the location of the two holes in the battery holder on the board and drill with a ⅛″ twist drill.

The left rear side of the PC board has three tracks on the bottom of the board. If you are careful, you can drill between the larger track and the closer of the two thinner tracks. It is actually OK to let the screw head touch the wider track (ground), but it must not come close to the thin track (*Photo 3*). You can add a piece of mica insulator from a TO-66 transistor mounting kit. Alternatively, you could use nylon hardware in this location.

PHOTO 3: MOUNTING SCREW NEAR GROUND TRACK.



The other approach is to drill a hole in the left side of the PC board outside all three tracks, then drill a matching hole in the aluminum battery holder. This is a bit more work, but ensures that none of the mounting hardware comes near any of the PC tracks. Make sure the battery holder terminal lugs are facing forward in either case. Mark one lug "+" and the other "–."

Once you have mounted both battery holders, solder the red and black wires to the "+" and "–" lugs, respectively (*Photo 4*). If the wires come loose from the PC board pins, note that the positive pin is marked "+" on the PC board.

Double-check your wiring, and insert four C-size rechargeable cells in the holders. Replace the cover and install the cover screw. Make sure that the power switch is off, then plug the meter into an AC outlet and verify that the red BAT CHG light is lit. Charge the batteries for 24 hours before placing the meter into service. ***aX***

TABLE1

Parts List

QTY	DESCRIPTION
2	Keystone 2C series aluminum battery holders, Mouser P/N 534-149
4	NiMH or nicad rechargeable batteries, size C
4	4–40 × ¼″ machine screws, Mouser P/N 561–P440.25
4	#4 internal tooth lock washers, Mouser P/N 5721–LW1–04
4	4–40 hex nuts, Mouser P/N 561–G440

SOURCES

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

Local Electronics Store

Note: Surplus vendors often have rechargeable batteries and holders. All Electronics, www.allelectronics.com, 800-826-5432.—Ed.

PHOTO 4: BATTERY HOLDER CONNECTIONS.

Software Review

Groove Mechanic 2.5c

Back

By Gary Galo, Regular Contributor

***The Groove Mechanic 2.5c**, Coyote Electronics Inc., 11174 Canyon Cres., Delta, BC, Canada V4E 2R8; http://www.coyotes.bc.ca/;info@coyotes.bc.ca (e-mail), $39, requires Windows 95 or higher.*

I have several friends who make their livings doing audio restoration, including 78-rpm discs, lacquer broadcast transcriptions, LPs, and tapes of every conceivable format. A visit to their studios reveals six-figure investments in Cedar and Sonic Solutions audio restoration equipment, including de-clicking, de-crackling, de-hissing, azimuth correction, digital equalization, and editing. For those of us who can't afford or justify such expensive equipment, the "trickle-down" effect of technology has produced a large quantity of surprisingly affordable audio noise reduction and editing software.

I use Sony Digital Media's Sound Forge 7.0 (formerly made by Sonic Foundry), along with their Noise Reduction 2.0 and CD Architect 5.0 package. These programs once sold separately for close to $1000, but now Sony bundles the three together, throwing in the Noise Reduction and CD Architect packages for the cost of Sound Forge alone. Programs offering similar performance would have cost thousands of dollars only a few years ago.

Groove Mechanic 2.5c is an extremely affordable package designed for removing noise from LPs and tapes. Groove Mechanic has been designed for the non-technical user, and offers remarkable simplicity of operation—you don't need to be an engineer to quickly master the program and achieve good results. The program is optimized for removing clicks and pops from vinyl LP records, hiss from tapes, and rumble from any audio source. The noise-reduction portion of version 2.5c is essentially the same code as 2.5b, but re-compiled with an updated compiler. This corrects a very rare problem in which a prominent click is not repaired. A minor bug dealing with track names has also been fixed.

Groove Mechanic is available only as a download from the manufacturer's website. The complete program is 1.3MB, so it will take just a few seconds with a broadband connection. You can download and try this program for a 15-day evaluation period. After that, de-noising will be limited to file lengths of two minutes. When you purchase the program online, you are given a registration code to enter into the program, which enables full, permanent operation. Groove Mechanic has a good help menu, and a 59-page user manual in .PDF format is also available online. I recommend downloading the manual, which gives a very clear overview of the program.

When you start Groove Mechanic, it defaults to the windowed mode. There's really no need to run it full screen, since there's no advantage to stretching the audio display vertically. In fact, it actually looks better windowed, but I recommend stretching the display so that it fills the screen horizontally.

You can bring audio into Groove Mechanic in two ways. The Capture mode allows you to record directly to the program through the line inputs on your sound card. You normally use the volume controls supplied with your sound card software, or with the Windows volume controls. Groove Mechanic's main display window can be toggled back and forth between Recording Level Meters (*Photo 1*) and Audio Timeline (*Photos 2–4*), which is the main display of your saved file. Groove Mechanic will also open .WAV files produced by other programs, and save processed files in .WAV, .MP3, or .OGG formats (.WAV is the default).

I recorded all of my test selections for evaluating these programs on Mitsui Gold CD-Rs, using a Sony CD-R800 professional CD recorder connected to my main stereo system. I extracted the tracks on the CD-Rs and converted them into .WAV files using Sound

PHOTO 1: THE GROOVE MECHANIC "RECORD LEVEL" METERS. THE PROGRAM USES THE LINE INPUT ON YOUR SOUND CARD FOR RECORDING. RECORDING LEVELS ARE SET USING EITHER THE SOUND CARD OR WINDOWS VOLUME CONTROL SOFTWARE.

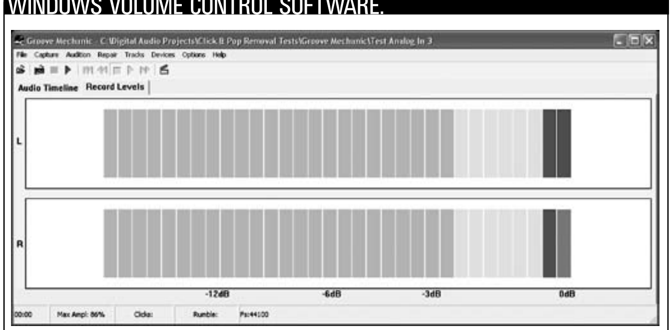


PHOTO 2: THE COLUMBIA IVES RECORDING LOADED INTO GROOVE MECHANIC. THE VERTICAL SPIKES ARE LARGE CLICKS, CAUSED BY A PRESSING DEFECT ON THE RECORDING.

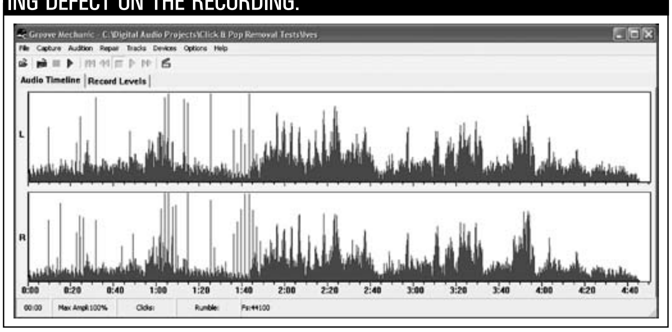
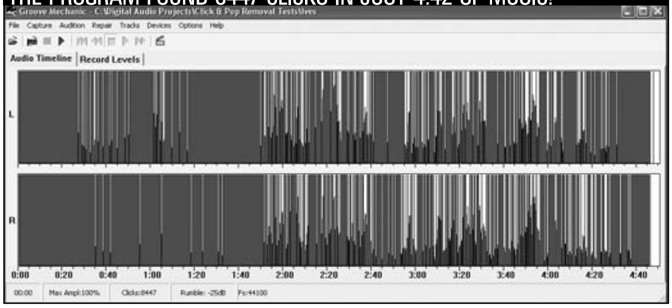


PHOTO 3: THE DISPLAY AFTER ANALYSIS OF THE IVES RECORDING. THE PROGRAM FOUND 8447 CLICKS IN JUST 4:42 OF MUSIC!



Forge (Groove Mechanic won't extract audio from a CD). This is sonically the best route, in my view, even though I have a very good sound card (Creative Labs Audigy 2) in my computer. The oversampling, 20-bit A/D converters in the CD-R800 are excellent, and the analog circuitry in the CD-R800 is powered by a conventional regulated power supply, rather than the switching supply in my computer.

SIMPLE OPERATION

Using Groove Mechanic could hardly be simpler. First, you open the .WAV file that you wish to process. The file will appear in the left and right display windows (*Photo 2*). The window gives you an amplitude display over time, but doesn't show the actual waveform. The loudest clicks will be immediately visible as large vertical spikes louder than the music.

At this point, you can play the file using the transport controls on the toolbar. All noise reduction begins with an analysis of the file—under the Repair menu, select "Analyze Clicks and Rumble." The program will take a few moments (depending on the file length) to analyze the file, and will mark all of the clicks with vertical red lines in the display window (*Photo 3*).

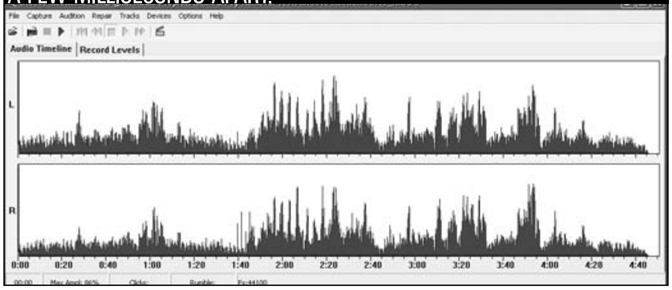
Then, run the appropriate noise-reduction program on the menu, which will be "Fix Clicks and Thumps" for LP records. The program will take a few moments to remove the clicks, and return the screen to its original appearance (the red vertical lines will be gone). You can now play the file and verify how successfully the clicks were removed. Large clicks in the original file will still be visible in the display, making it easy to determine whether or not they were removed when you play the file.

After you've repaired the file, you can save it. Groove Mechanic automatically adds "_fixed" to your original filename, so you won't accidentally overwrite your original file. When you close and then re-open the "_fixed" file, the display shows only remaining clicks—all those removed on the first pass are gone (*Photo 4*).

Hiss reduction begins with the same analysis. In order to reduce hiss, the program must have a sample of the noise without the music. After the initial analysis, you tell the program to get a noise base at the playback cursor, or at the end of the file. Then, select "Fix Hiss." The program will take a few moments to reduce the hiss. Then, you can play the repaired file to see how effective the hiss removal has been.

Under the "Options" menu, Groove Mechanic allows a few user adjust-

PHOTO 4: THE RESTORED FILE AFTER SAVING, AFTER THREE PASSES WITH GROOVE MECHANIC. THE PROGRAM DID WELL ON MOST OF THE CLICKS, BUT A FEW REMAIN AROUND 1:40. THESE REMAINING CLICKS ARE REALLY CRACKLES, SINCE THEY CONSIST OF MULTIPLE CLICKS ONLY A FEW MILLISECONDS APART.



ments. You can select three click settings—large, medium-to-large, and "all." Fine adjustment between these coarse settings is also possible. Three boxes, if checked, select "Detect Pops (Wide Clicks)," "Ensure clicks are not due to voice," and "Don't find clicks that are not audible." I find that selecting "All clicks," along with checking all three boxes, works very well for most material (these are the program defaults). You can set the Rumble Filter for a corner frequency of 20, 25, or 30Hz.

The Hiss Threshold Control choices are "Normal," "Remove less noise," and "Remove more noise." There are also a few General Preferences that most users can leave alone. One that I changed is "Stop updates the playback cursor." If left checked, the playback will resume where it left off if you select Play after Stop. If you uncheck it, the cursor will return to its original position when you select Stop. This is a matter of preference, but I'm glad the program gives you a choice.

Nostalgic for the MS-DOS days? Groove Mechanic can also be run noninteractively from the command prompt. In a single command line operation, you can have Groove Mechanic open a file, analyze it, repair it, save it, and exit. You can also write batch files (remember them?) that will instruct Groove Mechanic to work on several different files at once. The affection I once held for MS-DOS waned years ago, so I did not try running the program from the command prompt.

LP TESTS

Table 1 is a list of the test recordings I made for evaluating both of these programs, along with brief descriptions of the various noise problems. For my LP tests, I copied portions of five different recordings with specific problems, usually two to four minutes long. My "second copy" of the Sheffield direct-to-disc of Wagner excerpts conducted by Erich Leinsdorf has light, random ticks and low-level background crackle. I used a portion of the Prelude to *Tristan und Isolde* for these tests. The first pass removed nearly all of the clicks, and a second pass got most of the remaining ones. Background crackle was lowered, but not eliminated.

A couple of low-level ticks remained, but their high-frequency content was greatly reduced. There's an option in Groove Mechanic called "Force thump detection at cursor." After analysis and repair of clicks and thumps, you can move the cursor to a spot just before a problematic pop, and instruct the program to analyze this one thump. Groove Mechanic will search for remaining thumps for five seconds beyond the cursor. I tried this on the remaining clicks, but the program gave me the message "No thump found near playback cursor."

The Hindemith disc conducted by William Steinberg has moderate, random ticks. With version 2.5b., one pass with Groove Mechanic removed all but one click in the right channel. I tried two more passes, but the right channel click remained, despite the fact that the program continued to find new clicks on every pass. Forcing thump detection at the cursor did not remove the remaining right channel click. This is the rare problem with version 2.5b. Version 2.5c worked flawlessly on this selection—all clicks were detected and removed on the first pass.

On the Furtwängler Beethoven 9th recording, the first pass removed the minor click, but barely touched the very loud one. Here, forcing thump detection at the cursor did the trick. After the first pass, I tried this option, but the program detected a thump in only the right channel—the pop in the left channel remained. Running this option after a second general click removed the remaining left channel pop.

On the Ives LP Groove Mechanic was pretty effective in reducing the low-level crackles caused by a pressing defect, and most of the large clicks were caught the first time through. But, several of the clicks were not removed even on the fourth pass. I brought the restored file into Sound Forge to zoom in on the clicks. I found that these clicks were more like crackles, in that each one consisted of multiple clicks only milliseconds apart. Groove Mechanic makes no claims for removal of crackle, which is a much more difficult proposition than single, isolated clicks (Sound Forge's Vinyl Restoration will fix these problematic crackles).

I deliberately ruined an extra copy of the *Belshazzar's Feast* record that I made about 20 years ago. Using a compass point, I put three evenly spaced scratches on side one, and three scratches about ½″ apart on side two. These are very loud scratches with a lot of high-frequency content. Side one required two passes to get them all. The program did not remove all traces of the scratches, but left behind only a very low-level, low-frequency bump.

Side two required three passes to

TABLE1

Recordings Used to Evaluate Groove Mechanic 2.5c

LP CLICKS AND POPS

Charles Ives: *The Things Our Fathers Loved*— "Slow March," "Canon," and "There's a Certain Garden." Helen Boatwright, Soprano; John Kirkpatrick, piano. Columbia Masterworks M4 32504, Record 3, Side 1, Band 1. (Loud crackles and pops at the beginning due to a pressing defect).

Beethoven: Symphony No. 9 in D minor. Berlin Philharmonic Orchestra conducted by Wilhelm Furtwängler, Rec. London May 1, 1937. Toshiba/EMI WF-60073–74. Excerpt from 2nd Mvt. (There's a gouge in these otherwise pristine records on Side 2, causing two ticks, one minor and one very loud).

Wagner: *Tristan und Isolde*—Prelude to Act 1. Los Angeles Philharmonic Orchestra conducted by Erich Leindsdorf. Sheffield Lab LAB-7 (Direct-to-Disc). (Light, random ticks).

Hindemith: *Symphonie Mathis der Maler*, Mvt. 1 Introduction. Boston Symphony Orchestra conducted by William Steinberg. Deutsche Grammophon 2530 246. (Moderate, random ticks). (An outstanding performance by an underrated conductor that deserves a CD re-mastering by DG).

Walton: *Belshazzar's Feast*. Crane Chorus and Crane Symphony Orchestra conducted by Brock McElheran. CSM-1003, a record produced privately by The Crane School of Music, SUNY Potsdam, from analog tapes I recorded in 1984. (I had some extra copies of this record lying around, so I sacrificed one for these tests. I put three large, evenly-spaced scratches at the beginning of the piece, and three more large scratches ½″ apart at the beginning of Side 2.)

TAPE HISS

Fauré: *Requiem*—Libera me. Crane Chorus and Crane Symphony Orchestra conducted by Nadia Boulanger. Spring Festival of the Arts, 1958. 15-ips full-track mono original tape.

78-RPM TESTS

(These discs were transferred to digital using a Technics SP-15 turntable, SME-3012R Tonearm, Stanton 500AL Cartridge with truncated elliptical styli of various sizes, and my modified McIntosh C-8 preamplifier. Playback speeds and phono equalization are noted for each record. A modified Phoenix Systems Parametric Equalizer was also used for the acoustical recording).

Verdi: *La Traviata*—Prelude to Act I. Philharmonic-Symphony Orchestra of New York conducted by Arturo Toscanini. Victor 6994-A, rec. 1929. 77.0-rpm, 300Hz bass turnover, –5dB at 10kHz. (An unworn copy of a late-1930s pressing, but there are a large number of clicks and pops, plus the usual steady-state surface noise.)

Cohan: *Over There*. Enrico Caruso, tenor; Victor Orchestra conducted by Josef Pasternack. Victor 87294, rec. 1918 (acoustical recording). 75.00-rpm, flat phono EQ, parametric equalizer set to +6dB at 180Hz, 0.7 octaves. (A pristine, single-faced copy with virtually no clicks or pops, but the usual steady-state surface noise from the abrasive fillers used in Victor's shellac formula.)

Verdi: *Il Trovatore*—"Miserere." Rosa Ponselle, soprano (Leonore); Giovanni Martinelli, tenor (Manrico); Victor Orchestra conducted by Giulio Setti. Victor 8097-A, 77.0-rpm. 400Hz bass turnover, flat treble. (The usual steady-state surface noise present on scroll-label Victors from the late 1920s, plus some clicks and pops, and distortion on the final chord caused by too high a recording level.)

get everything, again leaving only a low-level, low-frequency component. On both sides, the second and third passes did not further improve the low-frequency bumps left behind—they only helped on scratches missed by the first two passes. Scratches like this make a record nearly useless, but Groove Mechanic restored this recording to a very listenable state.

Even with professional restoration equipment, it is often necessary to run more than one pass in order to remove or minimize noise. My friends in the restoration business tell me that it is often better to run multiple "light" passes rather than a single "brute force" treatment. Groove Mechanic's algorithms have been intelligently designed, with multiple passes in mind. The manual notes that you usually hit the point of diminishing returns after two or three passes, which I found to be the case.

TAPE HISS REDUCTION

I recently made a digital transfer of a 1958 Crane School of Music performance of Gabriel Fauré's *Requiem* conducted by Nadia Boulanger. I did this for a colleague on our faculty who was presenting a paper on Boulanger's association with our school (Boulanger was a close friend, and a former teacher, of Helen M. Hosmer, director of our school from 1930–1966; Hosmer brought Boulanger to Potsdam several times as guest conductor of our Spring Festival performances by Crane Chorus and Crane Symphony Orchestra). This recording is a 15ips, full-track mono tape made on equipment well below the state-of-the-art for 1958, and it is quite hissy. I selected the "Libera me" movement for this test.

Groove Mechanic did a surprisingly good job reducing the hiss on this recording. I found that the "remove less noise" option worked best. If I select the "Normal" option, I begin to hear some subtle, high-frequency swishing artifacts around the strings. The "Remove more hiss" option increases the audibility of those artifacts, and also introduces some subtle pumping of dynamics. This is no surprise, since even the multi-thousand dollar hiss reduction programs will leave artifacts if overused.

I was quite impressed by how well Groove Mechanic handled this tape using the most conservative setting. Other settings may be appropriate for different program material. Groove Mechanic does not audibly degrade the high-frequency response of the original file, which is very impressive. Surprisingly, Groove Mechanic also found a large number of clicks on this tape, but I didn't find any benefit in removing them, since they are not audible.

78-RPM DISCS

Groove Mechanic wasn't specifically designed to handle noise from 78-rpm discs, and the manufacturer doesn't make any claims for 78s. But, I decided to try a couple anyway. My Victor copy of the *Traviata* Prelude with Toscanini and the Philharmonic-Symphony Orchestra of New York has minimal wear, but a lot of clicks and hiss. Two passes with Groove Mechanic removed all of the clicks, leaving only a steady background of hiss. These hiss-removal algorithms are not compatible with the hiss encountered on 78s. Even the most conservative setting caused dynamic pumping and high-frequency artifacts, but the click removal worked quite well.

On the other 78-rpm discs, the program was less effective. Groove Mechanic seems to interpret high levels of background hiss as very closely spaced clicks, and can introduce crackling sounds where none existed before. This was especially true in a few spots on the "Miserere" from *Il Trovatore* with Ponselle and Martinelli.

Caruso's "Over There" is an extremely clean copy, but with the normal amount of 78 background noise. Click removal actually introduced a layer of noise around loud, high notes. Again, Groove Mechanic wasn't designed for 78s, but discs with moderate, random ticks, such as the Toscanini *Traviata* recording, it works pretty well.

LIMITATIONS

Groove Mechanic has a couple of limitations. The program allows you to produce boundary markers in the display, which are ultimately used to produce tracks on the CD. But, it isn't possible to zoom in on the display window, so an entire CD's worth of audio will be displayed across the width of your computer screen. This makes it difficult to navigate within the window on files as long as a full CD, and rather tricky to precisely place the boundary markers that will be converted into CD tracks. The manufacturer tells me that a zoom feature will be added to a future release.

Groove Mechanic doesn't support CD burning. Most computers come with CD burning software (my Dell Dimension 4600 came with Sonic Record Now!) that will burn CDs from .WAV files. Groove Mechanic will not embed track information into .WAV files. When you extract your audio to a file, Groove Mechanic actually extracts each track to a separate .WAV file, the same way .MP3 files work.

One of the nice things about the .WAV format is that it doesn't have the "one-track-per-file" limitation of the .MP3 format. The .MP3 format—sonically and functionally—was designed with pop music in mind, but the designers of the .WAV format had the foresight to consider the needs of classical music as well. Since Groove Mechanic does not save the track boundary information with your .WAV file, you have only one chance to get it right before you close the file and lose all of your track information. If you find that one track location is out of place after you've extracted all of the tracks to individual files, you must go back into Groove Mechanic and mark the boundaries for the entire CD again.

When you use your CD burning software to burn the files to a CD, a two-second pause is usually placed between each track, unless you tell the program to do otherwise. With many classical works, especially operas, there are often multiple tracks within continuous music. The two-second pause between tracks—an inherent limitation with "track-at-once" CD burning—is no good for classical music. With Sonic Record Now!, you select burning in the Disc at Once mode from the options menu, rather than track at once.

To my surprise, this actually works fine. Multiple tracks—even if they occur within continuous music—are reassembled seamlessly this way. If your interests include classical music, you'll need a CD burning program that allows "Disc at Once" burning.

Your CD burning software may also make it possible to overcome the lack of zoom capability with Groove Mechanic. For longer CDs, it makes sense to work on only a few tracks—or a single LP side—at a time, minimizing the time in the display window. After processing all of the .WAV files you need for one CD, use your burning software to assemble the various tracks in the desired order.

CONCLUSIONS

Every audiophile probably wishes to know whether this software "degrades" the sound quality. If you are a lover of vinyl who insists that the LP is still superior to the CD, no computer program for digitizing LPs will satisfy you. Programs like this were not designed for you. Keep your LP playback equipment in good working order and continue to play those LPs.

The process of digitizing LPs will invariably introduce some change to the original sound, especially at the 44.1kHz/16-bit CD standard. There are many places where this degradation can occur, especially in the analog circuitry and A/D converters in your sound card. There are also numerous opportunities for the introduction of clock jitter, including the noise caused by the switching power supplies used in all computers.

If you perceive a change in the sound of your LPs, there are many factors that can account for this. Groove Mechanic does a good job of minimizing or removing noise without introducing audible degradation of the original .WAV file. If the program errs, it is on the conservative side, leaving a bit of noise behind rather than introducing audible artifacts.

Groove Mechanic is an effective program for click, hiss, and rumble removal. It isn't perfect, but it offers extremely good performance at a bargain price. Even with expensive professional restoration equipment, it often is not possible to remove all of the noise from a recording, at least not without degrading the original sound. Groove Mechanic's algorithms have been intelligently designed to deliver good results, while keeping the user interface simple. If you are looking for an easy, cost-effective way to convert your LPs to CD, cleaning them up in the process, Groove Mechanic will do the trick. *aX*

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CAP SOLUTION

I wish to share with you my experience in replacing my tweeters and making a crossover for them. These are for my BES (Bertagni Electroacoustic Systems) D-280 speakers, which I bought from Monterey Stereo (Calif.) in 1980. They radiate sound in a 360° pattern—a large, wide, and deep sound, which is a quality I desire in a speaker. I did not entirely like the highest frequencies they produced through the use of cheap piezos.

I purchased a pair of neo 8-PDR planar transducers by Bohlender-Graebener for $62.68 each from Parts Express. They have very good horizontal coverage and project sound both front and rear with effortless ease and naturalness. They blend well.

On making the crossover I experimented with several values before settling on a 9.1µF 250V metallized cap by AEON. I put in a 0.50mH copper foil air-core inductor to filter the lower frequencies, under about 18kHz. The frequencies blend well with no noticeable spikes. The BES speakers have an upper midrange, which goes very high, so I needed to consider that when I listened to many cap values for the crossover. I have only my ears for the test equipment, so I need to trust them.

After a few days of listening, I thought I needed more extension on top. I put in a .10µF 600V polystyrene film and foil MIT multicap and two 10nF 600V signature film and foil versions by Infini. (All caps and inductor from Michael Percy Audio.) I got extension in spades.

This is where the fun is, or begins. After more nights of listening, I realized that not all was perfect with the crossover. I'm hearing a smear of the high frequencies. It's like many parts of the highs are overlaying—multiples of the same frequencies are being overlapped, one upon another.

I then remembered a white paper by Infini on bypassing: When bypassing do not exceed 1/100 of the value of the main capacitor! This is to avoid spectral overlap. The 10nF cap exceeds the 9.1µF by 10%.

I replaced the 10nF with a 15nF 600V MIT multicap together with the two 10nF bypasses and was well within the safety zone (no spectral overlap, only good extension, and the sound is very coherent). I feel much better now. I may put in a couple more 10nF caps for even more extension. I'm sure there is a limit on how many you can use, even while staying within Infini's safety zone. For fun, I want to test the outer limits of this, though, before I settle down.

I recall a similar problem when I bypassed a cap in my preamp's power supply. In a few days I wondered what was going on with the sound. By changing to a lower valve cap in accordance with Infini's recommendation, all was well. It was easy to hear.

William R. Harren
Ridgefield, Wash.

AUDIO DIVERSITY

Your publication continues to improve in its service to the audio enthusiast. In an increasingly digital world, *audioXpress* retains the necessary dedication to the analog foundation of recorded and reproduced sound. Despite a few complaints to the contrary, I believe that your consolidation of the three former, independent magazines is the only option in maintaining an economically viable product. I look forward to each issue and the diversity it brings to this hobby. In the October issue, I found Mr. Still's approach to "bass enhancement" ("High-Quality Tube Type Control Unit," p. 26) of particular interest.

Jim Wood, Pres./Chf.Engr.
INOVONICS, INC.

SIGNAL WIZARD

I've enjoyed reading "Room Correction" by Rune Aleksandersen in the Aug.–Nov. issues of *audioXpress*. Can you provide me with the name of a supplier for the Signal Wizard 2?

Don Anstedt
sparks33@mc.net

Rune Aleksandersen responds:

There is a new and improved version of the Signal Wizard DSP card now available. I received this e-mail from Brian Jones in San Diego, Calif.:

I met Patrick (designer of the program) on the Internet after purchasing Signal Wizard 1.6, which I wanted to use for arbitrary equalization and linear phase crossovers for loudspeakers—a hobby. As you mentioned in your article, Signal Wizard 1.6 has some limitations for this type of work—no flash, independent filtering, and so on. Now all that has changed.

I have been testing Signal Wizard 2 for Patrick, putting it through a whole battery of tests, and the audio quality and the filtering/EQ capabilities are really very exciting. Patrick has also added the delay feature that allows the crossover designer to completely compensate for the acoustic offset between drivers. I tried this on a simple KEF UniQ bookshelf speaker two weeks ago, using 96dB/OCT FIR filters. Signal Wizard can handle each two-way loudspeaker.

In an effort to see the benefits of Signal Wizard reach the marketplace, I am arranging the manufacture, testing, and supply of the complete package. Shortly, we should have a way for customers to easily purchase and receive the product directly from here. Production of the boards started at the end of October, and the first kit (PSU, cables, S/W) will be available toward the end of November.

Brian Jones
San Diego, Calif.

This is information from Patrick Gaydecki, the designer of the Signal Wizard:

Signal Wizard 2 has many more features than Signal

Wizard 1.6, including flash memory for storage and automatic execution of filters from power-up, true dual-channel operation, FIR and IIR filters, adaptive filters, channel mixing, delay functions, and a real-time digital oscilloscope and spectrum-analyzer feature. It is also some six times more powerful.

If you wish to order a system, please send a check made payable to "The University of Manchester" to the address below. Checks will only be processed once systems have been dispatched. The cost of the Signal Wizard 2 system is £250 + VAT (362 euros or $448).

For a full description of Signal Wizard 2's capabilities, please visit the website: http://www2.umist.ac.uk/dias/pag/signalwizard.htm

Patrick Gaydecki, BSc, PhD, FIEE, MInstP, MInstNDT, CPhys, Ceng Sensing, Imaging and Signal Processing Group, School of Electrical and Electronic Engineering
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United Kingdom

DVD JITTER

During the sound and video experimentation with my Sony DVP-300 DVD player, it was sometimes necessary to operate the player in the Factory Diagnostics mode[1]. This easy-to-use resource allows you to look up stored error codes, clear them after repairs are made[2], and request the diagnostics program to optimize all electromechanical adjustments to minimize measured jitter[3].

The diagnostic menus and their attendant results are displayed on the TV monitor hooked up to the player. The service literature, in sometimes strained English, describes how to operate the diagnostic resources by using the remote control from a comfortable couch.

The reduction of measured jitter is always accompanied by noticeably sharper images, better color rendering, and more accurate audio. The process is specialized for CDs, single-layer, or double-layer DVDs. Unfortunately, the written instructions fail to clarify a crucial step in the jitter optimization process, so I offer text to overcome that lack and to inform readers possibly interested in optimizing their own players.

Automated Jitter Reduction

After you identify the kind of disk just mounted, the diagnostic program runs through a sequence of adjustments to fine-tune the disk-reading process by varying the gain and offset voltages of each servo-driven device used to read the disk. One such device tilts the disk to achieve the best possible angle between the reading laser beam and the disk surface, and also uses measured jitter as the payoff variable. When the optimization program is finished, all settings are stored as binary numbers in EEPROM until another adjustment is performed. Literally, better it could not get.

First Adjustment Cycle

Upon completing the first optimization pass, real-time measured jitter is displayed while the disk continues to spin, and the user is prompted to press the "stop" button or let the program continue. Here the service literature omits a critical piece of text. HIT THE STOP BUTTON to store the OPTIMIZED adjustments in EEPROM. You are now finished.

Final Adjustment Cycle

If you let the program continue, the tilt servo will be put into the center of its range, then held there to allow you to adjust a screw inside the machine to minimize the jitter displayed on the monitor. The only time this adjustment is needed is when a new read-head has been installed (for example, during manufacture), and is otherwise unnecessary.

It is imperative that you hit "stop" after the first jitter optimization cycle. This avoids the fact that when the second cycle is done, it leaves the tilt adjustment in the center of its stroke, perhaps far from the optimum position, causing very crude adjustment numbers to be stored in the EEPROM when the diagnostics end. Soft images and mid-fi audio are sure to result.

Peak Tuning Forever

When loading a CD or DVD, I now observe the practice of running the jitter optimization sequence before I play each disk. That way I'm assured that every disk starts out from the best adjustment, which consumes perhaps 30 seconds.

The displayed jitter remaining after

optimization is a telling indicator of the manufacturing quality that particular disk has received. Not only do disks vary widely, their sonic and graphic performance is correlated with their jitter. It's nice to know the player has done everything possible to make the best of it. I'm beginning to like this player, it really tries hard to please.

Darcy E. Staggs
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REFERENCES

1. The Sony service literature contains assembly illustrations, operation details, schematics, parts layouts and specifications, diagnostics, and pin descriptions of all major integrated circuits. Sony also sells spare parts (I replaced a ribbon cable) from a central distribution center in Kansas City, Mo. (1-800-488-7669).
2. Causes of repairs have been broken foil traces, short circuits (blown fuses), cold solder joints, incorrect component values, failed ribbon cable conductors, contaminated reading optics, and a disk jammed in the drawer.
3. Jitter can be described as the statistical variation in the time between ticks of a digital clock inherent in the state changes in a stream of digital data. Depending on its magnitude, both audio and video details become smeared.

BOOK REVIEW

I'm finally getting around to writing and saying thank you for reviewing my book (*Sound Recording Advice*) in your Sept. 2004 issue! That was an excellent review by Richard Honeycutt. Wish I could have had him do a once-over before I sent the manuscript to the printer. That was my first book effort, and I started from square zero in self-publishing and authoring with no real background in either.

PHOTO 1: 6550 TUBE-BASED AMP.

As for the audio cables that only "pass signals in one direction only," they really do pass signals in both directions. The manufacturer of the cables puts the arrows on the cable jacket to alert the user that the shielding is more effective when the source signal flows to the destination in the direction of the arrows. Is this going to make a huge audible difference for your average Joe? I suspect not.

John Volanski
soundadvice@johnvolanski.com

Richard Honeycutt responds:

I am grateful for Mr. Volanski's kind words regarding my review of his book. I still suspect reptile lubricant when manufacturers indicate preferred "source" and "destination" ends of cables. The only case in which cable orientation in relation to source and destination should matter is for a line-level balanced cable, in which the shield is sometimes disconnected at the source end, but should be connected to the chassis at the input to the destination device.

I wish Mr. Volanski much success in his current and future publishing efforts.

WORKING WITH TUBES

As a long-time subscriber to *Glass Audio* and *audioXpress*, I thought it was time to share with you a recent project I completed. *Photo 1* shows my 6550-based amplifier, which is based on Joseph Norwood Still's original design, but uses 6SN7s in the preamp/phase inverter sections. I corresponded with Mr. Still on the revised design.

I have also constructed Mr. Still's 6L6-based amp and Rick Spencer's 6V6 amp. Both designs work exceptionally well.

Now for a bit of constructive criticism. I enjoyed Graham Dicker's series for beginning amp builders ("Tube Audio Construction Tips, Parts 1–4," April–July '04 *aX*), but had to chuckle on some points. After buying and selling numerous tube caddies via eBay and locally, I should warn beginners that they contain mostly useless and disappointing tubes of little or no value to serious tube amp builders. Encouraging beginners to pursue these as a source of useful tubes is leading them in the wrong direction.

The second point relative to that series deals with metal working for an amplifier chassis. Whatever happened to good old chassis punches? They are available on Mouser or eBay. I could not imagine tackling a project such as my 6550 amp armed with a hole saw!

Carl Hartman
carl_hartman@hotmail.com

Graham Dicker responds:

As Mr. Hartman points out, the content of tube caddies will not help the seasoned serious amplifier builder find output valves such as 2A3s, WE300Bs, 6L6s, KT88s, KT66s, EL34s, and other much used audio tubes.

As a long-time tube experimenter, and in the tradition of ham radio operators worldwide, I never underestimate the value of a good junk box, especially one with lots of tubes. For example, if you need half of a 12AX7, the EC91 is a good alternative; commonly found TV tubes such as the 6GW8 and the 6BM8 make a nice 10W per channel push-pull amplifier; and even the 6GV8 vertical deflection tube can serve as a nice audio amplifier. The 6M5, 6CW5, and EL86 vertical output tubes are close replacements for the 6BQ5.

There are literally hundreds of usable TV-type replacement tubes that you can use for audio. My favorites, the 6CM5 and 6DQ6A, make great audio output tubes, even though they were designed for horizontal TV use. Have a look at the following website for some interesting applications:

http://www.alphalink.com.au/~cambie/EL36.htm

Mr. Hartman also provides some good sources for chassis punches. I have a set of these, but I have not used them in over three years on aluminum and bakelite chassis; instead, I use a standard spade type woodworking bit and a little cutting oil. It takes less time to cut a hole, the edge of the hole is as good as a punched one, and a set of six titanium tipped spade bits will only cost around $3, as opposed to $25, per punch size for Qmax ones.

JFET AVAILABILITY

In the Dec. 2004 issue of *aX*, Norman Thagard declares that the 2SK389 is no longer available ("Single-Ended to Differential Mode Made Easy," p. 13). As I have mentioned before, MCM (who, BTW, has an ad on page 57 of this issue) lists many "Japanese" transistors. I called them a couple of days ago and spoke to Barbara. It was like Ripley describing the eggs in "Aliens": "There are hundreds of them," she said.

John Nickerson
johnnickerson@verizon.net

Norman Thagard responds:

Although it has been quite a while since I wrote the comment to which Mr. Nickerson refers, I believe that it was based on an even earlier e-mail inquiry from a reader who asked whether there was a suitable substitute for the 2SK389 and 2SJ109, both of which were used in my phono preamp design published in Jan. and Feb. 2001 and were no longer available. I did nothing to corroborate the writer's assertion and simply mentioned it in my recent article.

This morning I searched Toshiba's website without finding either transistor. I believe Sanyo second-sourced these dual JFETs, but a similar search of their website also gave zero matches. A web search for the 2SK389 and 2SJ109 did reveal that some are still available from distributors, which is good because perhaps additional readers will build my phono preamp.

Shop carefully because some vendors were not located in the United States and were asking as much as $4 for the 389 and $10 for the 109. I also found some prices as low as $0.72 and $1.80, respectively, but only six of the former and three of the latter were in stock at those prices.

I am unmotivated to search further to see whether these devices are still under manufacture, but it looks doubtful to me. My advice is to buy now rather than later.

CIRCUIT BASICS

In "Line Stage Odyssey Continues" by David Davenport (Nov. '04 aX), I cannot find any info about real anode voltage B+ (Fig. 9, p. 59). And what is a turns ratio and the inductance of T1 and T2 transformers in this circuit? Also, what is the voltage gain (in/out) of Fig. 9, and what are frequency response limits?

Nick
nick2bike@mail.ru

David Davenport responds:

In an amplifier you can set either the plate voltage or plate current but not both. As the name implies, a constant current source (CCS) sets the current in the circuit and allows the plate voltage to adjust itself to maintain that current. The current is set at about 8mA, and for that current the plate voltage typically is about 165V. The B+ input to the CCS is about 200V, but this voltage is not critical as long as it is high enough to keep the CCS in regulation.

The ratio chosen for T1 is 2:1, and the ratio chosen for T2 is 4:1. These determine the input and output impedances, gain, and frequency response of the circuit. Many—if not most—preamplifiers have too much gain, with the user needing to operate the volume control at 7 or 8 o'clock. I have set the gain of the circuit to 6.5dB, which provides reasonable adjustment of the volume control for most amplifiers. The frequency response with this gain is 6Hz to 40kHz.

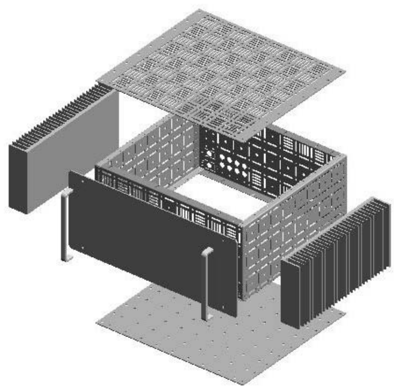
TANG BAND REMAKE

The article "Remaking Tang Band's W3-881S" in the December issue (p. 54) was thoroughly interesting. I was left with two questions after reading it. How did Dr. McKenzie come up with the improvement measures, and how is it that the manufacturer wouldn't have done things like this to start with?

Mark Rumreich
mark.rumreich@thomson.net

Mark McKenzie responds:

I developed the driver modifications by applying my knowledge of loudspeaker cone design and a proprietary process for mapping cone and diaphragm vibration modes under dynamic conditions. As to your second question, it is not really my place to pronounce on the decisions of commercial manufacturers. This is a case of "your guess is as good as mine." Looking at the range of possibilities, however, the word contractions "won't" or "can't" seem to cover most possibilities. **XM**



solid state

NTSC Video Fulfilled:
POOGE Works For TVs Too

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By Darcy Staggs

In past issues of *audioXpress*[1,2,3] I chronicled the changes made in an NTSC TV and consumer-grade DVD player, claiming that this work resulted in "fully realized" NTSC imagery. Here I offer photographic examples to illustrate this application of video circuit enhancement through Progressive Optimization of Generic Electronics[4] (POOGE).

The images presented come with the kind permission of Joe Kane, producer of the DVD from which they are reproduced, *Video Essentials*[5], which I recommend to anyone interested in accurate video reproduction. It has helped me with every phase of monitor convergence and proper adjustment.

The DVD player is connected to the 27″ Zenith TV set auxiliary input with a coax cable carrying a composite signal, meaning all video must pass through the TV's comb filter to be separated into chroma and luminance. I have disabled the set's discrete 3.58MHz color sub-carrier filter and turned on the equivalent filter built into the video processor with noticeable improvement. Newer TV sets offer component and/or S-video (separate chroma, luminance) inputs which bypass the comb filter, resulting in improved sharpness.

GRAPHIC EXAMPLES

Photos 1 and *2* are presented as examples of color rendering. The processes of film photography, scanning into a computer, and printing the image in a magazine have affected color accuracy. From my couch, however, colors are pure and brilliant enough to elicit a strong emotional reaction from me and my guests.

Photos 3 and *4* indicate the kind of sharpness now present. However, all photos were made from moving subjects containing dithered video, so viewed sharpness appears better than shown here.

PURPLE PROSE

Improvements in image quality can be summarized in text, although words can't really convey the sensation of a brilliant picture on a CRT screen. Notwithstanding, I offer the following:

Sharpness – easy to follow during the modification process. Removal of halos as well as the arrival of natural edges appear as sharpness. Truly fast capacitors and the elimination of noisy electrolytics are the main contributors.

3-D Shading – the tonal gradation of light and shadow across a 3-D object has replaced a "cookie cutter" quality the image once had. Cinematographers excel at their trade, and generally expose each scene with only the tiniest areas bleached white, only a small part of the subject totally black, exploiting the entire contrast ratio of the film, screen, CRT, and so on.

Color – it always struck me that it shouldn't be impossible to combine three primary colors in a CRT and come up with some really good ones. It's happening now. The best colors are from nature, where flowers, sky, people, and other familiar objects are rendered. Live TV broadcasts can be stunning; for example, field reports from a doughnut shop, where the close-ups of fresh goodies look irresistible, sprinkles and all, and from awards nights and sports broadcasts, which also offer accurate live color.

It took a while before certain colors "came online," particularly yellow and orange, which began as very anemic and washed out, finally bursting forth.

PHOTO 1: ENHANCED COLOR SAMPLE (1).

PHOTO 2: ENHANCED COLOR SAMPLE (2).

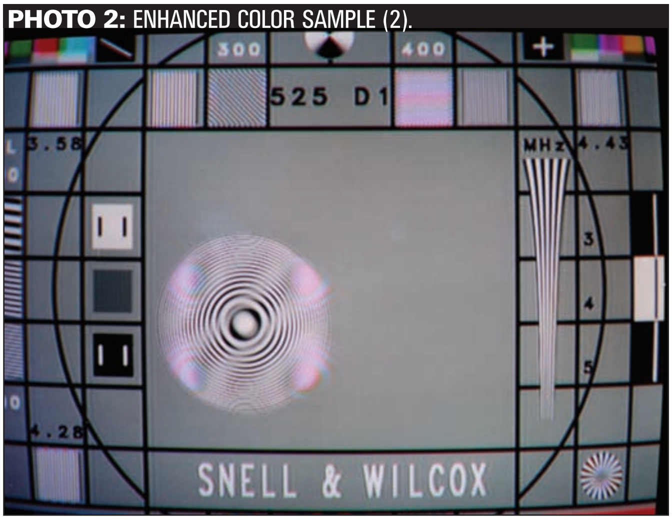

PHOTO 3: ENHANCED SHARPNESS SAMPLE (1).

PHOTO 4: ENHANCED SHARPNESS SAMPLE (2).

The precise summation of primaries does the trick.

Transparency – between objects in an image is clear air devoid of fog or subtle, grainy noise. This contributes to the realism of an image. Live broadcasts look truly amazing with this quality, with DVDs a very close second.

Snap – you could argue that turning up the contrast would yield this improvement in transient response, but this isn't the case—it's only the peaks that needed help. The line cord and improved power supply and signal caps make nighttime scenes look very realistic. Neon signs, street lighting, and so on look very sharp and brilliant, with a searing, pinpoint light and precise, vibrant colors.

Tonal Gradation – subtle variations in color and its intensity add just as much detail to a scene as detail of physical size. Architectural photography, clothing, automobiles, and so on look much more interesting with this improvement.

CONCLUSION

Here you've seen a small taste of the video facet of electronic upgrades. Except for working with surface-mount devices, this effort was no more difficult than past audio efforts, and certainly yields enduring pleasure. I have applied nothing more than the audio POOGE principles appearing more or less continuously in this magazine, and meanwhile have learned many interesting things about TV[7] and DVD technology. ***aX***

REFERENCES

1. "A DVD Rescue," *audioXpress,* July 2002, p. 24–27.
2. Guest Editorial, "Detrashing Television," *audioXpress,* June 2003, p. 4.
3. Xpress Mail, "Capacitor Violence for the Unintimidated," *audioXpress,* July 2003, p. 67–68.
4. The POOGE series was published in *The Audio Amateur,* 1/81, 4/81, 4/85, 1/88, 2/88, 2/92, 3, 92, by Walt Jung, Chuck Hollander, Dick Marsh, Gary Galo, and Hampton Childress.
5. *Video Essentials, Optimizing Your Audio/Video System* (DVD), Joe Kane Productions. This is the NTSC version, highly recommended, but out of print. A new version is available, entitled *Digital Video Essentials,* http://www.videoessentials.com/.
6. Photos were made in a darkened room on 120-size Kodak Porta 160 NC color negative film with a Rolleicord V equipped with a 75mm F3.5 Schneider-Kreuznach Xenar lens. The light meter is a Weston Master 6. Prints were made by Technics Photo Service Inc., Orange, Calif., on 6″ × 7″ Kodak printing paper.
7. Useful TV reference data is found at: http://www.ntsc-tv.com/ntsc-index-01.htm (an authoritative source of the NTSC broadcast television signal format standards), http://www.arcadecontrols.com/files/Miscellaneous/crtfaq.htm (informative CRT technology discussion), http://www.cybertheater.com/ (rich source of introductory-to-intermediate level NTSC video technology), and http://www.smpte.org/ (official site of the Society of Motion Picture and Television Engineers).

classified

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VENDORS

Mic and phono preamps, meters, filters, etc. all US built. **TDL Technology, Inc., 505-382-3173** **www.zianet.com/tdl**

Principles of Power, tube audio books, kits, FAQ **www.londonpower.com**

DIY Starter Kits
www.borbelyaudio.com
Selected Borbely Audio kits in North America:
www.audiokits.com
Selected Borbely Audio kits in Japan:
http://homepage3.nifty.com/sk-audio/

www.stillaudio.com
18 watt 0.6% distortion (no loop feedback) single-ended amplifier

Serious Heatsinks! 12″ long, machined, cast aluminum. Also matches ezChassis prepunched cabinet. **www.designbuildlisten.com**

Yard Sale

WANTED
Selector switch (5-position) for Realistic STA-95 stereo receiver or info about an equivalent. roncollier@erols.com or call (804) 780-0331.

I need the schematic for the PS Audio Model One Power Amplifier. Any help is appreciated. Please contact me by e-mail at mageorge@sbcglobal.net.

"Yard Sale" is published in each issue of aX. For guidelines on how subscribers can publish their free ad, see our website.

Product Review

Testing the Parts Express MTM Kit

By Joe D'Appolito
Listening Critique by Dennis Colin

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FIGURE 1: PE MTM KIT IMPEDANCE MAGNITUDE (BOTTOM) AND PHASE (TOP).

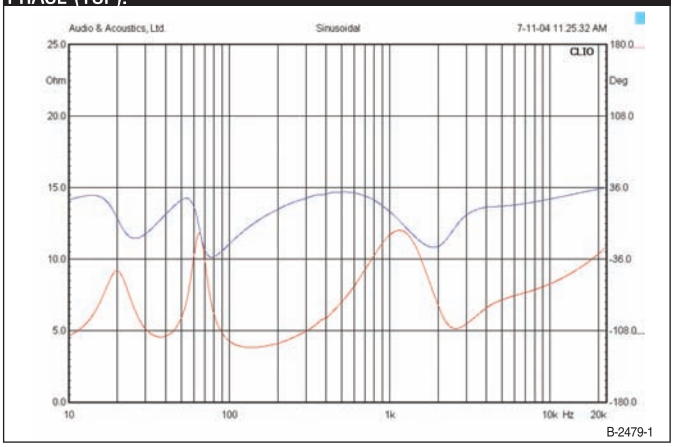
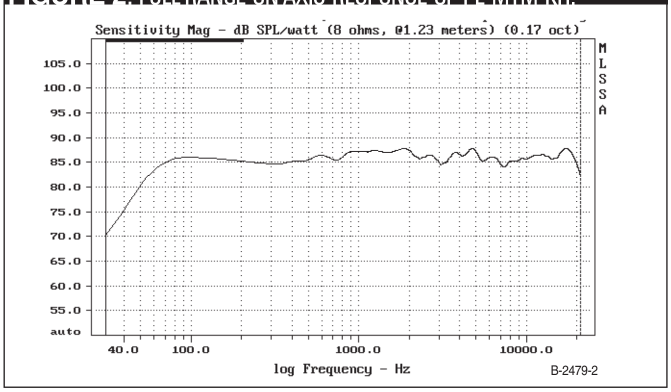


FIGURE 2: FULL-RANGE ON-AXIS RESPONSE OF PE MTM KIT.



I ran a series of impedance, frequency response, polar response, and distortion tests on the Parts Express MTM kit that Ed Dell built (March '05 *aX*).

IMPEDANCE

Figure 1 is a plot of system impedance magnitude (bottom) and phase (top). At low frequencies the plot displays the double peaked curve typical of vented systems. The impedance minimum between the peaks at 39.3Hz indicates the vented-box tuning frequency (Fb). There is an absolute impedance minimum of 3.12Ω at 136Hz. Impedance phase angles range from +32° to –34°. I would rate this a 4Ω speaker.

FREQUENCY RESPONSE

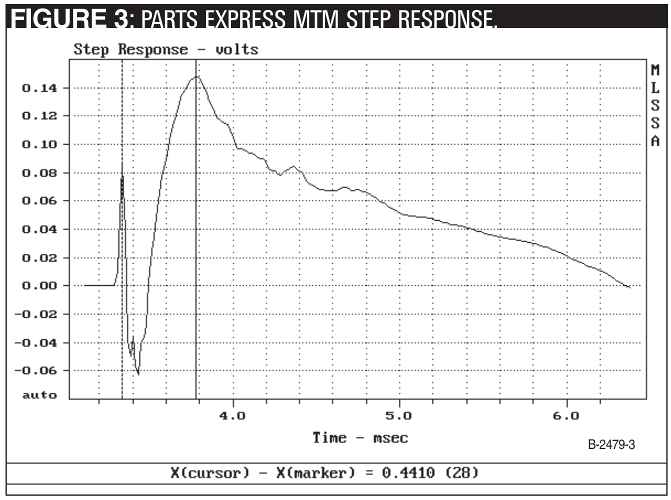
Figure 2 shows the full-range frequency response of one of the pair of MTM speakers. This response is obtained as a combination of the far-field quasi-anechoic response in the freestanding position and a ground plane measurement[1]. I placed the microphone along the tweeter axis at a distance of 1.23m to produce the far-field freestanding response. I took the ground plane measurement with the mike placed at 2m on the tweeter axis. I then spliced ground plane and far-field responses together at 200Hz to produce the full-range response[1]. The response data is ⅙ octave smoothed. In the two octaves between 500Hz and 2kHz sensitivity averages 87dB SPL/2.83V/1m. Relative to this level, response varies by +0.9dB and –3dB over the range of 100Hz to 20kHz. The low-frequency point –3dB is at 58Hz.

The PE MTM is not configured for bi-wiring, so I could not determine responses for the individual drivers. But from data to be shown later, the crossover point appears to be somewhere in the range of 2000 to 2500Hz.

WOOFER/TWEETER TIMING

The PE MTM step response is plotted in *Fig. 3*, showing two separate arrivals of acoustic energy. The initial sharper positive spike is the tweeter arrival. It is followed by the woofer arrival, also positive, beginning about 160μs (0.16ms) later. The woofer peak is delayed relative to the tweeter peak by 440μs. These peaks are not, however, a good measure of the average tweeter/woofer delay. Although not shown, a detail examination of the excess group delay plot[1] shows the woofer pair to be exactly 290μs (0.29ms) behind the tweeter. Although both drivers connected with positive polarity, the system is not time-coherent.

FIGURE 3: PARTS EXPRESS MTM STEP RESPONSE.



CUMULATIVE SPECTRAL DECAY

The PE MTM cumulative spectral decay (CSD) response is presented in *Fig. 4*. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.2ms increment of time. The total vertical scale covers a 30dB dynamic range.

Ideally the response should decay to zero instantaneously. Inertia and stored energy that take a finite amount of time to die away, however, characterize real loudspeakers. A prominent ridge parallel to the time axis indicates the presence of a strong system resonance.

The first time slice in *Fig. 4* (0.00ms) represents the system frequency response. There are major response ridges at 3.1kHz and 5kHz decaying slowly and extending out to 5ms. I suspect these come from the woofer pair. The tweeter generates a fair amount of "hash" above 10kHz extending out to 2ms followed by a series of ridges disappearing at about 4.5ms. This is characteristic of many metal dome tweeters. Compared to other loudspeakers I have tested for *audioXpress*, this is rather poor decay performance. See Dennis Colin's reaction to this performance in the accompanying Listening Critique.

HORIZONTAL POLAR RESPONSE

Horizontal polar response is examined in *Figs. 5* and *6*. *Figure 5* is a waterfall plot of horizontal polar response in 10° increments from 60° right (+60°) to 60° left (–60°) when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00°. For this reason, the plotted curves show the *change* in response as you move off-axis. For good stereo imaging the off-axis curves should be smooth, evenly spaced replicas of the on-axis response with the possible exception of some tweeter rolloff at higher frequencies and larger off-axis angles.

Within ±30°, the off-axis curves are indeed fairly smooth replicas of the on-axis response. The –3dB coverage at 15kHz is ±22°. At 10kHz the –3dB coverage extends to ±30°. This is typical of 25mm

FIGURE 4: PE MTM KIT CUMULATIVE SPECTRAL DISPLAY.

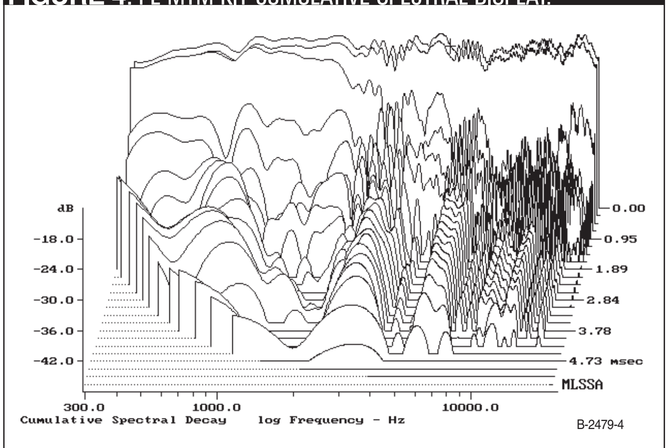
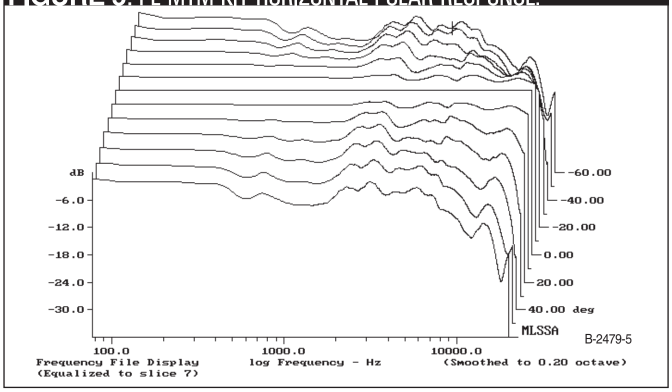


FIGURE 5: PE MTM KIT HORIZONTAL POLAR RESPONSE.



dome tweeters. The 60° curve shows the transition from the woofer pair to the tweeter starting just above 2kHz. Here you can see change from the narrowing response of the woofers to the broader response of the tweeter.

Within the woofer pair active range the worst case –3dB width occurs at 1.4kHz and ±35°. This is due to the woofer pair directivity at that frequency. At ±60° off-axis and 1.44kHz response is down 7.2dB. By 3kHz the system output has fully transitioned from the woofer pair to the tweeter. At 3kHz and ±60°, response is down only 2.1dB. The response at large off-axis angles is typical of two-way systems.

The average response over a 60° horizontal window (±30°) in the forward direction is a good approximation of the way a speaker will sound in a typical listening environment. This response is plotted in *Fig. 6* (solid line) and compared to the on-

FIGURE 6: AVERAGE HORIZONTAL RESPONSE OVER ±30°.

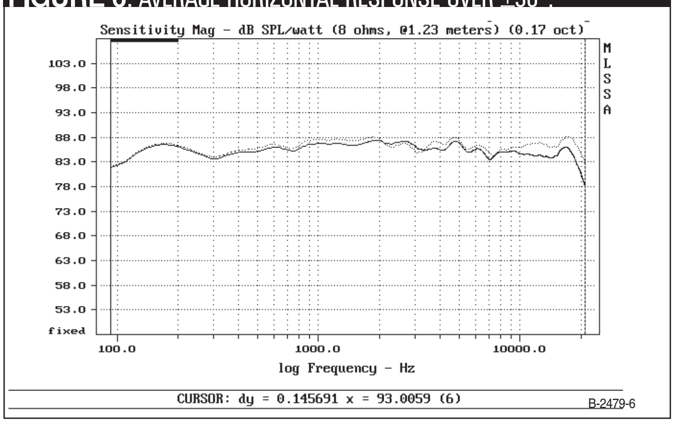


FIGURE 7: PE MTM KIT VERTICAL POLAR RESPONSE.

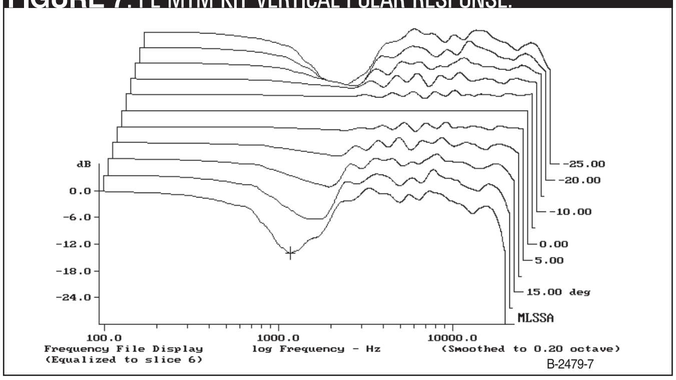
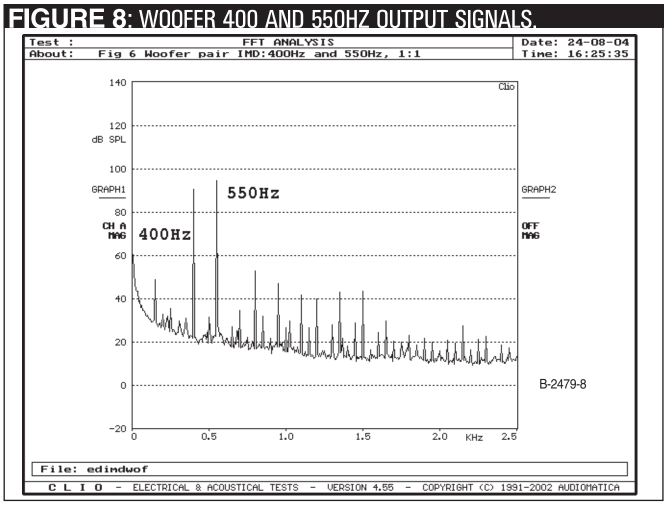


FIGURE 8: WOOFER 400 AND 550HZ OUTPUT SIGNALS.



LISTENING CRITIQUE

By Dennis Colin

My first impression (and one that persisted) was that this speaker has a reasonably well-balanced sound, smooth and extended highs, reasonable bass (for its size), and the very good image focus and staging that you expect from minimum-width-baffled MTM arrays. But there's something about the midrange that I would describe as an indistinctness, a lack of clarity and immediacy. Without benefit of reviewing Joe D'Appolito's measurements, I'd say the midrange response sounded somewhat uneven, and also emphasized in the 1–3kHz "presence" region.

These characteristics were not severe enough to be constantly disturbing. Depending on the music, the midrange coloration could be moderately noticeable, able to be ignored, or almost inaudible. Strangely perhaps—considering that 1–3kHz is where the ear is most sensitive and where voice detail and harmonics are most prevalent—voice was not unpleasantly or unnaturally affected. Voices sounded good. Most affected were violins and massed strings, horns, clarinet, and (to a lesser degree) piano and lower pitch-range percussion.

SPECIFIC RECORDING IMPRESSIONS

Turtle Creek Chorale, **Reference Recording RR-905CD:**
I've waxed poetic about the haunting depth and quality of these male voices singing "Alleluia" on the Thor speakers (May '02 *aX*). I wrote, "I've never heard more natural-sounding voice reproduction, period." Well, that's a tough act to follow, but the MTM units did a good job with Turtle Creek—the voices were smoothly reproduced and focused, though of course not stunningly so, as on the Thors. I noticed only a slight midrange emphasis.

Jacintha, **FIM XRCD020**
There was a slight sibilance (emphasized "s" sound), but otherwise her voice and the instruments, including string bass, were good-sounding in tonality and spatial depth.

Carmen Fantasy, Percussion Fantasia, **FIMCD017**
From deeply pitched drums to tiny triangles, this recording is an excellent test track for transient clarity, impact, and frequency response extremes. With these speakers, the HF response and transients were very good; midrange was somewhat indistinct and "presence region" emphasized; and the bass drums were reasonably solid and deep.

Swing Live with Bucky Pizzarelli, **Chesky SACD223**
This is a hybrid multichannel SACD of very high quality, recorded in a New York City jazz club. Instruments are guitar, vibes, drums, clarinet, and acoustic bass. Everything sounded good except the clarinet, whose spectrum is apparently too sensitive to these speakers' (as I perceived) midrange emphasis. The intimate, well-recorded venue space was very well-reproduced. The audience's uninhibited applause and conversations were realistic, if somewhat overly, present. Overall, the colorations could be forgotten, because the sensation of being there in that warm and intimate jazz club was very convincing.

K622, **Musical Fidelity MFSACD017**
The speakers' (as I perceived) midrange indistinctness and emphasis unfortunately affected the beautifully played and recorded strings and clarinet. Not so much, however, to prevent Ed Dell, Joe D'Appolito, and me from becoming lost in the ethereal beauty of this music. Incidentally, this is a first-rate recording (if you have an SACD player) for auditioning these format differences.

RECOMMENDATIONS

I express some reservations with orchestral music, regarding string tonality. The effect, however, is neither serious nor unpleasant. For jazz, rock, and much vocal and popular music, these speakers can be quite satisfying, and appear to be worth their modest price. I'd guess that the tweeter is of good quality but the mid/woofers are not as good in their midrange. Considering the excellent and beautiful cabinets, it might be worth your while to replace the mid/woofers (if my guess is correct) with units commensurate in smoothness with the tweeter.

COMMENTS ON MEASUREMENTS

Looking at the frequency responses on-axis and ±30° averaged, I'm happy for the speaker, but not for my ability to correlate these responses with my sonic perceptions. I had guessed that the midrange area was uneven and emphasized around 1–3kHz. On-axis,

axis response (dotted line). The average horizontal response is within 1.2dB of the on-axis response out to 10kHz and is only 2dB down at 15kHz. This is excellent horizontal performance and suggests good direct field coverage in the primary listening area. There will be little change in spectral balance with changing position. Image stability should be very good.

VERTICAL POLAR RESPONSE

Figure 7 is the waterfall plot of vertical polar response. Responses are shown in 5° increments from 25° below (–25°) the tweeter axis to 25° above it. Off-axis responses out to ±5° track the on-axis response with little error. At ±10°, off-axis vertical response is down 3dB at 1845Hz. As angles approach ±20° and more, deep symmetric notches develop just below 2kHz. This performance is typical of the MTM geometry, and is actually one of its major advantages, greatly reducing floor and ceiling bounces that may confuse imaging.

INTERMODULATION DISTORTION

In past reports I have listed results for both harmonic and intermodulation distortion (IMD) tests. However, I believe IMD tests are more revealing of loudspeaker performance and more demanding of loudspeaker linearity. So in this and future reports I will list only IMD results.

In this test two frequencies are input to the speaker. IMD produces output frequencies that are not harmonically related to the input. These frequencies are much more

1–3kHz *is* elevated, but by only about 2.5dB; what I heard sounded like about 5dB.

The CSD might explain this: Joe said, “rather poor decay performance.” The prominent 3kHz ridge “looks like what I heard”—the indistinctness, lack of clarity. The time-domain “hanging on” of 3kHz and higher-frequency “hash” would explain not only a loss of clarity, but perhaps also a sense of tonal emphasis.

Furthermore, both horizontal and vertical off-axis responses show the 2–4kHz region elevated relative to the depressed 1–2kHz octave.

This is an (unusual) example of sound not correlating well with the ±30° horizontal averaged frequency response. But the overall measurement picture, including the CSD and dispersion responses, does appear to correlate with what I heard. I have yet to hear a speaker whose predominant sonic characteristics can’t be related to some physically measurable phenomenon visible somewhere within Joe’s extensive acoustic tests.

Manufacturer’s response:

Part I: Assembly Process

Unfortunately, Mr. Dell (*aX* 3/05) was under the impression that the kit sent to him had the front baffle already permanently installed. This caused numerous problems throughout the assembly process related to driver, crossover, and foam installation. The truth of the matter is the cabinets are indeed shipped with the baffle in place on the front of the cabinet, but they are not *permanently* attached. The cabinets have the baffle merely screwed in place with four Philips-head screws for shipping purposes (the same holds true with all of the PE kits). As it states in the “getting organized” portion of the manual, the baffles should be removed upon unpacking of the cabinets. It also states within the first few paragraphs of the instructions: “if there are any questions during the assembly process, please feel free and contact our Technical Support staff at 1-800-338-0531.” If Mr. Dell would have taken a moment to inquire about his baffle situation, he could have quickly cleared up the issue and saved a lot of unnecessary difficulties.

A few other minor inaccuracies within the article are worth commenting on, even though they are somewhat insignificant. First, the cabinets are constructed out of Medium Density Fiberboard, not particleboard as stated in the article. They have ¾″ panels for the top, bottom, sides, and rear, but use a 1″ thick front baffle. The inductors that are shipped with the kit do come with pre-stripped and tinned leads, so no extraneous stripping is necessary as long as you don’t insert the leads too far. Regarding the woofer hookup to the crossover boards, the woofer “plus” and “minus” terminals are clearly marked—the cause for confusion on this matter is unclear.

Overall, we feel that Mr. Dell did a good job keeping the review unbiased, and in describing the assembly process in matter-of-fact terms. It is a shame that the baffle installation issue was not brought up earlier in the process, as Mr. Dell’s experience would likely have been quite audible and annoying than harmonic distortion. Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

First, I examined woofer intermods by inputting 400Hz and 550Hz signals at equal levels. These frequencies should appear predominantly in the woofer output. Total SPL with the two signals was adjusted to 90dB at 1m. *Figure 8* is a plot of the resulting output spectrum. The principal output frequencies of 400 and 550Hz are highlighted on the graph.

In addition to these frequencies, you will see many additional vertical lines representing both IMD products and harmonics of the 400 and 550Hz signals. Counting only the IMD products, there are significant lines at 150, 950, 1350, and 1500Hz. The overall level is only 0.66%. Although not bad in an absolute sense, this level of IMD is higher than most other woofers I have tested for *audioXpress*.

Next, I measured tweeter intermods with a 10kHz and 11kHz input

FIGURE 9: TWEETER IMD PLOT.

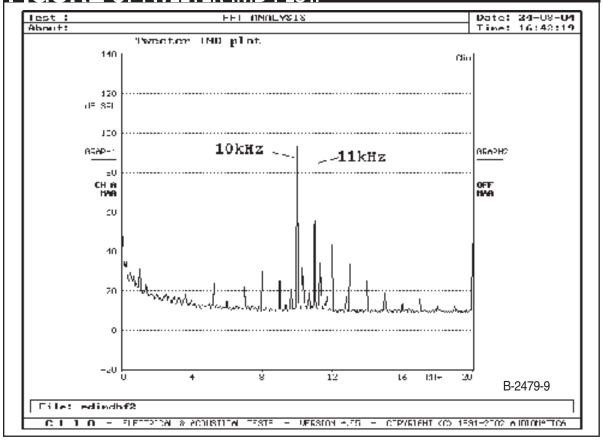
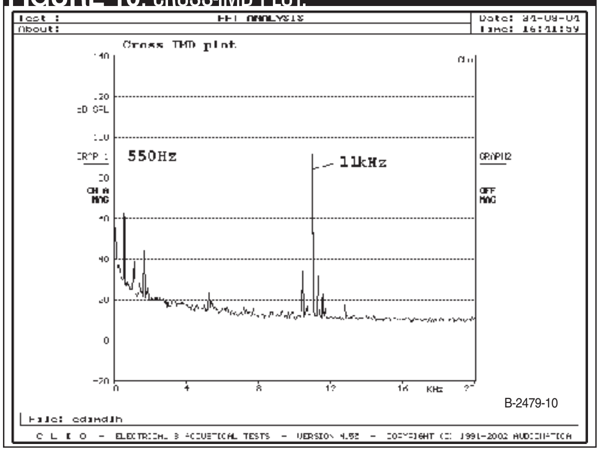


FIGURE 10: CROSS-IMD PLOT.



different without the extra difficulties the baffle presented. Nevertheless, Mr. Dell's main ideas still come through: these kits are relatively easy to put together, require only a few tools, and have a minimal chance for assembly error.

Part II: Measurement and Listening

Mr. D'Appolito has, of course, done a very thorough job with the measurements of this speaker system. There is little doubt of the accuracy of the information he has provided, but some of the data still warrants a few comments and interpretations. First, these speakers are indeed not time-coherent, and nor are they claimed to be. In reality, it is next to impossible for any speaker of this same physical configuration to be time-coherent. This is not a flaw of the Vifa/Dayton MTM kit design, but merely a characteristic of its construction. Second, it is worth considering whether the less-than-stellar performance in the spectral decay plots may be related to the poorly secured front baffles. Our own in-house CLIO waterfall plots of the kit prototypes do not show issues at the cited frequencies, nor do the individual responses of the aluminum cone woofers.

Mr. Colin mentions that the Vifa/Dayton MTM speakers are being auditioned on the heels of the Seas Thors. The Thors, being designed by Joe D'Appolito and with a cost of over $1600 including cabinets, are indeed a tough act to follow. It is important to keep in mind that the Vifa/Dayton speakers cost less than one-third of the Thor system and were designed to maximize the sound quality, ease of assembly, and aesthetic appeal *for their price*.

Some of the issues with the tonal balance of the Vifa/Dayton MTM could be related to the speakers' room positioning and the characteristics of the listening environment. The speakers do exhibit a slight rise in the presence region when measured anechoically, but this can help provide a more balanced sound and greater clarity when used in a typical home listening environment. Since the Thors measure extremely flat anechoically, and Mr. Colin very much likes their sound, it stands to reason that unlike a typical living room, his listening room has very little low-frequency reinforcement.

As with any speaker that you would build from a kit, build from scratch, or purchase from a store, each speaker has its own strengths and weaknesses. Mr. Dell chose this particular kit for his own reasons, and Parts Express did not offer any advice or suggestions on which kit to review. If the specific goal was to have a very clean midrange presentation, then a different kit might have been the better choice. But, by looking at the individual sonic ratings done by Mr. Colin, you will see that the Vifa/Dayton MTM speaker system does very well in most categories, and when you consider the price, is an excellent value.

Darren Kuzma
Parts Express Product Manager of Speaker Building

pair adjusted to produce a 90dB SPL at 1m. Results are plotted in *Fig. 9*. The major IM products occurred at 8, 9, 12, and 13kHz. Total IMD was 0.26%, a result in line with many other tweeters I have tested.

The last IMD test examines cross-intermodulation distortion between the woofer and tweeter using frequencies of 550Hz and 11kHz, again at a 90dB level. Results for this test are graphed in *Fig. 10*. Ideally, the crossover should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. Tweeter sidebands appeared at 10,450 and 11,550Hz at an overall level of 0.11%. This is a very good result and indicates good inter-driver isolation by the crossover.

SPEAKER MATCHING

All of the test results reported so far were obtained from a single sample. Now look at how well the two speakers match in frequency response. A frequency response plot of the *difference* between the two samples is shown in *Fig. 11*.

The two PE MTM samples match quite well below 2kHz. Above 2kHz the second system is within ±1.4dB of the first out to 20kHz. This is fairly good matching, especially for a kit, and bodes well for image stability.

GRILLE EFFECTS

I conducted all of these tests with the grille off. *Figure 12* shows the PE MTM response with the grille on, but referenced to the response with grille off. That is, it plots the *change* in response under the two conditions.

The grille has a fairly substantial effect on response at all frequencies above 300Hz. Overall response variations of +2.5dB to −1.4dB are shown. Response with the grille on is particularly ragged above 2kHz. As usual, the grille is there only for cosmetic effect and to protect the drivers from prying fingers.

A note on testing: The PE MTM loudspeaker was tested in the laboratories of Audio and Acoustics, Ltd. using the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems. Acoustic data were measured with an ACO 7016 ¼″ laboratory grade condenser microphone and a custom designed wideband, low-noise preamp. Polar response tests were performed with a computer controlled OUTLINE turntable on loan from Old Colony Sound Laboratory. *aX*

FIGURE 11: MATCH BETWEEN TWO PE MTM KITS.

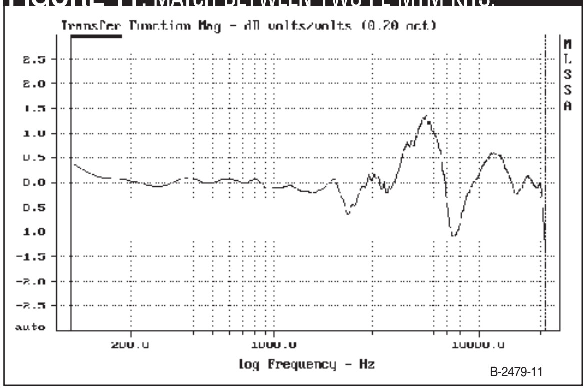


FIGURE 12: EFFECT OF GRILLE ON PE MTM KIT RESPONSE.

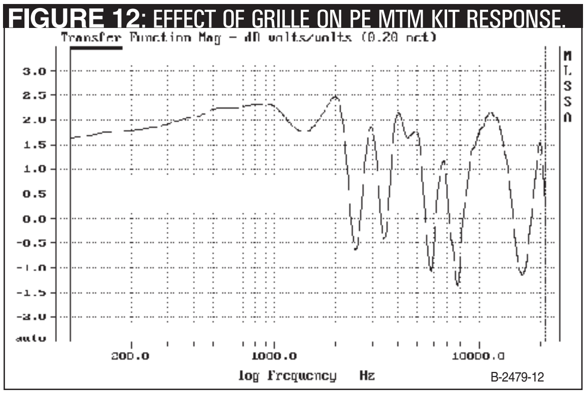


TABLE 1

Sonic Characteristics Ratings	
Presence	8
Freedom from distortion	8
Frequency response smoothness	7
L-M-H balance	7
Treble quality	8
Midrange quality	6
Bass quality	8
Bass extension	7
Immed. transient response	7
Image focus	8
Stereo soundstage realism	8
Ambience	8

REFERENCE

1. J. D'Appolito, "Testing Loudspeakers," Old Colony Sound Laboratory (PO Box 876, Peterborough, NH, 888-924-9465, custserv@audioXpress.com), 1998.

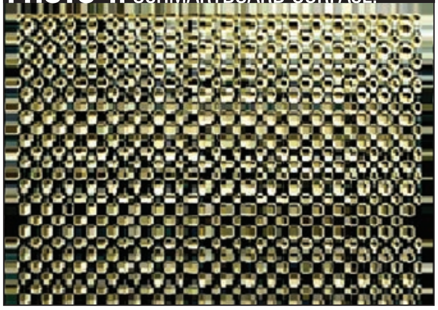
solid state

New Chips On The Block:

SchmartBoard

By Charles Hansen

PHOTO 1: SCHMARTBOARD SURFACE.



SchmartBoards are pre-traced printed circuit boards that are connectable like building blocks. They allow you to build electronic circuits block by block, and then connect the blocks together to form a functional board.

SchmartBoard is segmented into throughhole and surface-mount boards. In addition, they have a library of smaller I/O and power solutions. The boards are available in pre-defined combination packs or as individual units.

Because the boards are pre-traced (*Photo 1*), the need for wire jumpers is minimized, as opposed to breadboards or wire-wrap boards. The design process is quite simple:

1. **Select boards**—Select the boards that you will need for the project.
2. **Connect components**—Solder the components to each board. Segment your project into blocks to take advantage of the pre-routed traces on SchmartBoard. This minimizes the number of wires needed on each SchmartBoard.
3. **Connect boards**—Using the SchmartBridge mechanical bridges, connect the boards together. There are two slots on each side of the SchmartBoard. The bridges have two notches that fit into the slots (*Photo 2*). This plastic piece, in conjunction with the holes on the sides of SchmartBoards, allows you to physically connect SchmartBoards together.
4. **Connect wires**—Connect the wires. They have included wires and male headers in the combination and multi-packs. Use the headers to connect the wires.
5. **Test and make modifications**—You can make modification simply as needed. Use glue to make the connected boards more rigid. Do this only after you are satisfied with the final configuration.

SchmartBoard offers services as well as products. Once you are ready to take your SchmartBoard product to an actual PCB assembly, they can do it for you. Boards are priced from $5 to $48, depending on complexity.

The makers of SchmartBoard have a program to give free samples of the board, which can be found at http://www.schmartboard.com/index.asp?page=SchmartLAND_samples. There is also a contest where the company gives away six combination packs each month, at http://www.schmartboard.com/index.asp?page=SchmartLAND_contest. *aX*

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PHOTO 2: CONNECTING THE SCHMARTBOARDS.