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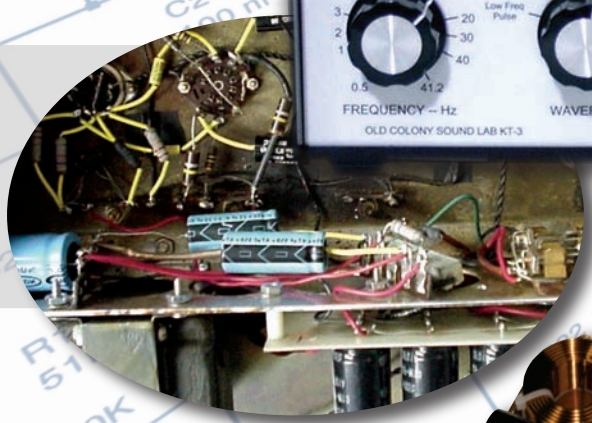

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Upgrading Heath's W6 Power Amp

With these tips from a vintage audio enthusiast, you, too, can successfully tame the Behemoth.

By Bruce Brown, RPh.

In 1957, there were hundreds of tube amplifiers on the market. Heathkit had successfully launched many great amplifiers, including the widely successful, extremely quiet W5. (See *audioXpress*, Dec. 2004, for my rebuild article.)

Responding to "power push" popularity, Heath introduced the W-6M, which produced 70W RMS and 140W peak power, and was Heathkit's most powerful tube amplifier. It was also the first Heathkit audio amplifier that used silicon diodes in a voltage doubler. This resulted in 470V delivered to the plates of the 6550 output tubes. These tubes had been rated for 600V, so the conservative use in these amplifiers assured a long, trouble-free life. But the unit was plagued by stability issues, which I address in this article.

The design of the circuit is different from earlier Heathkit amplifiers, and from most other audio equipment sold during this time. Input signal is fed to the grid of the 12AU7; direct coupling is used to feed this signal to the second triode stage of the tube. This second stage acts as a phase splitter, and the split signal is then fed to the grids of the 12AX7 push-pull amplifier stage and on to the grids of the 12BH7 cathode-follower driver stage. The cathodes of the 12BH7 are connected directly to the grids of the 6550 output tubes.

Bias voltage is applied to the 6550s via the grid return resistors of the 12BH7 driver stage. Variable damping is also provided to adjust the output to match different types of speaker systems. The impedance matching is selected by a plug and socket arrangement on the rear panel. This selects the proper current feedback resistor for the feedback circuit and damping control.[1]

PHOTO 1: Pardon the appearance, but these are the risks of buying online.

Copies of most Heathkit audio manuals are available from W7FG Vintage Manuals.[2] The manual for the W6 contains many frequency response charts and data and some very interesting oscillograms. I recommend you obtain a manual if you plan a rebuild.

THE PROBLEM

The W-6M was plagued with stability issues as the 12BH7 aged. This made it difficult to bias the 6550s; sometimes even a new 12BH7 would not allow proper bias adjustment. Heath introduced a replacement, the W-6A, in 1960.

The W-6A modifications solved these issues. Heath effected slight changes in the circuit with the addition of a bias balancing system and a bias range adjustment. This conveniently allowed for frequent bias checking and adjustment, and allowed the use of less than perfect tubes. Initially Heath eliminated the bias meter from the front panel, but at a later point returned it. (See the schematics for the W-6A and W-6M.)

The 6A was discontinued in 1962, and today the 6M is more commonly seen. The specifications for the W6 series are quite remarkable: 70W of power with a total harmonic distortion level of less than 2% at full rated output, from

20Hz to 20kHz.

Even more impressive, the frequency response at 1W was ±1dB from 5Hz to 80kHz, with a controlled rolloff above 100kHz[1]. This was achieved by the use of a heavy, specially designed Altec/Peerless output transformer, which is still generally regarded as having the widest bandwidth and best square-wave reproduction of any transformer produced,[1] capable of easily handling 140W RMS.[3] The W6 used a 12AU7, 12AX7, 12BH7, and two 6550 tubes. The oversize power supply and output transformer make up most of the weight of this unit and are responsible for its outstanding specifications.

The unit also featured silicon diodes in the high voltage supply. Advertising for the W-6M featured these "new" silicon diode additions as advantages, due to the lack of filament heat, their long life, and small size. (The space gained was pretty inconsequential in light of the size of these 50 lb giants.) It is estimated that fewer than 2000 of these monsters were ever produced.[3]

I obtained my W-6M from an online

PHOTO 2: Working on the Heath W6's innards.

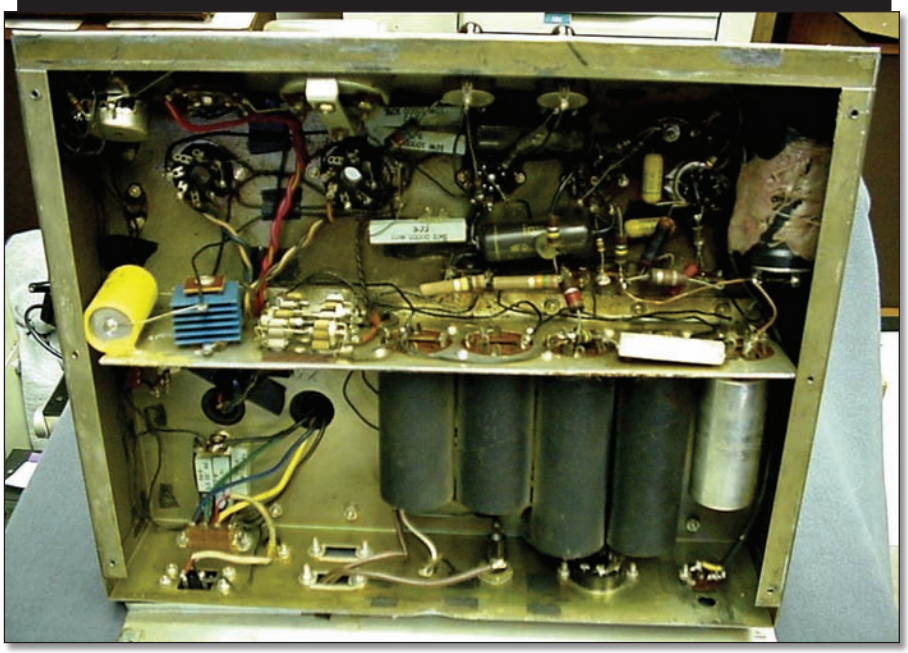
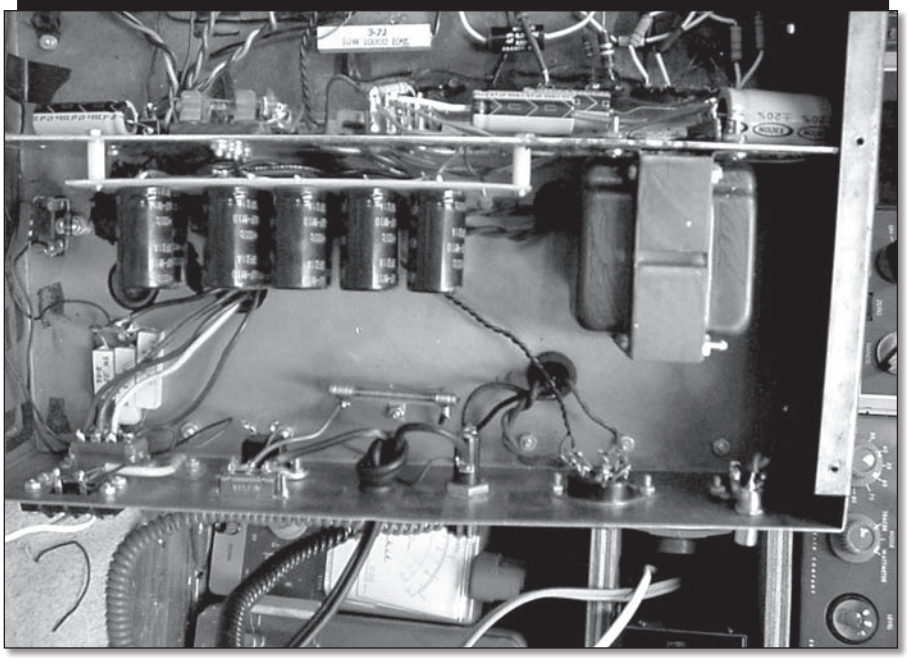


PHOTO 3: Replacing the capacitors.

auction. As you can see (**Photo 1**), it was pretty ugly, with tar or paint splatters covering the front and much of the top of the chassis. (I tried to clean these with a variety of solvents and ended up using a high-quality lacquer thinner from a local body shop, which removed some of the lettering and thinned out the gold paint in some places, but the cosmetics were more presentable.) The seller claimed "it had been used as a guitar amplifier" and had been functioning "before it was removed from service."

This is a great example of why you need to be careful buying online. When I contacted the seller, he was willing to guarantee the transformers, which was good enough for me. As you can see in **Photo 1**, a filter capacitor had been replaced, secured to the chassis with what appeared to be bubble gum. Bare copper wiring was a fire waiting to happen. Two other filter capacitors were shorted, and several resistors were open. This unit would certainly not have functioned, and it would have been a fire hazard. My advice is to contact the seller before you bid on any piece of vintage equipment to verify its working status.

PHOTO 4: Capacitor board installed.



POWER SUPPLY

I prefer to rebuild the power supply first, and once it is up and functioning, move on to the audio sections. I recommend that you draw a diagram of all existing connections and wiring.

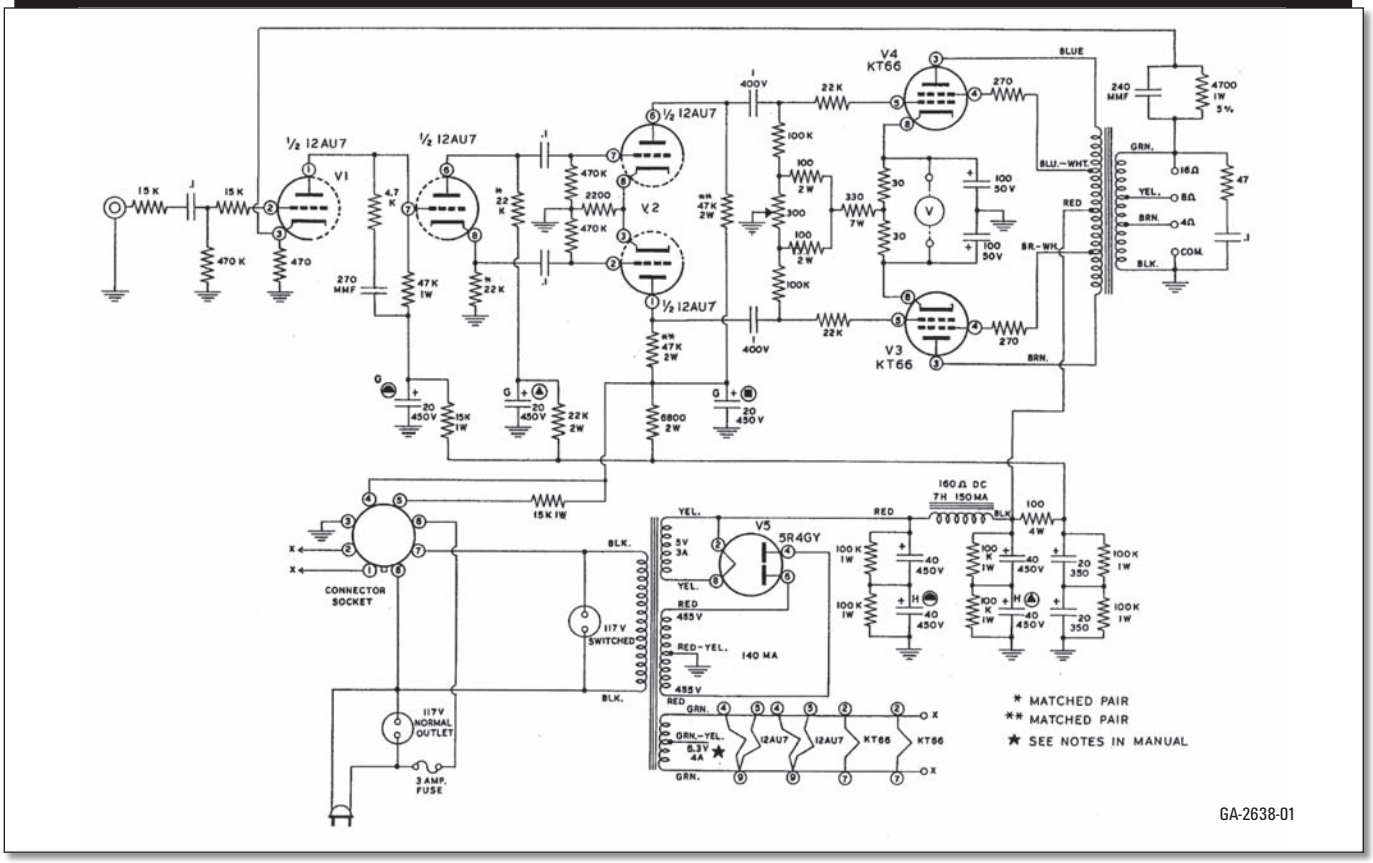
Carefully remove all the wires and resistors from the five can caps, remove the caps from the vertical inner panel, and set them aside (refer to **Photo 2**). Remove the selenium rectifier and associated capacitor and wires. You will now be able to partially remove the vertical panel.

Because most old can filter capacitors are either bad or past their usefulness, and comparable replacements are made out of "unobtainium," I put together a fiberglass board using eight 100µF 450V radial capacitors (see **Fig. 1** and **Photos 3** and **4**). These compact units have been very reliable. (I buy several dozen at a time for my tube audio project restorations.) Using two of these in series gives a voltage rating of 900V and a capacity of 50mF per section. By using eight 100µF 450V capacitors, you will have the same effective capacitance as the original power supply. As a bonus, this bank will take up substantially less chassis space than the original capacitors.

If you wish, you can add two to four more 100mF capacitors to increase the power supply capacity. I replaced everything past the voltage doubler diodes for the high voltage section.

If you size the board to the same width as the vertical panel, it is easy to use several 1″ standoffs to mount your new power supply panel (**Photo 5**).

FIGURE 1: The Heathkit W-5M high fidelity amplifier.



To connect the wiring, you will need a length of wire attached to the positive and negative output points on the panel. These will be run through the upright panel to the new additional capacitors associated with the 12GH7, 12AX7, and 12AU7 tubes.

A common issue with equipment designed in the 50s, described in my rebuild of the W5 in an earlier *audioXpress* issue, was that the line voltage back then was around 110V. Today, line voltages are often 120–127V, producing a problem with many vintage amp rebuilds. Dynaco Mark 3 tube amps commonly exhibit this problem. Sometimes the plate voltage on the 6550s will exceed 500V, and even go as high as 550V before the output tubes warm up. This significantly shortens the life of filter capacitors rated for 500V. This is why Dynaco originally used 525V cans, which are very hard to find today.* I tried adding a 5H 200mA choke between the voltage doubler capacitors and the main filter caps to lower the high voltage to the power tube plates. The resulting voltage drop with the choke was about 35V. This modification to the rebuild assures a long, trouble-free life.

In the first trial of my rebuild, by adding the choke I effectively lowered the plate voltage by about 35V, thus affecting the chassis-mounted bias meter reading, but when I used my DVM to set the bias at –50V, the amplifier meter deflected only three-quarters of the way to the bias mark. I removed the choke to solve this problem, and still managed to keep plate voltage to the 6550s at a reasonable level (452V).

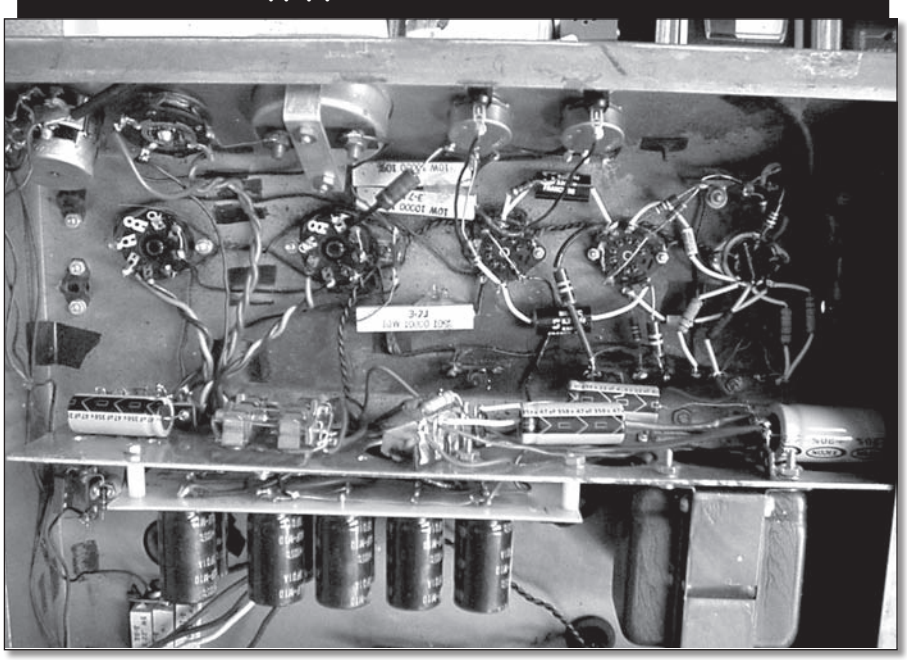
After solving the voltage drop problem, I replaced the selenium rectifier with a silicon diode for the bias circuit, which increased the value of the filter capacitor to 100mF 350V. This helped make the bias voltage more stable during heavy loads.

I replaced one of the rear panel outlets with a DPST slide switch and wired it as an "on-off" power switch. You can use the two wires that run to the rear octal outlet for wiring the switch (refer to the pictorial diagram for the before and after rear panel wiring). The rear octal socket has connections for the AC power to be controlled by an external preamp. You can use an octal plug to connect these two pins, or place a jumper across them.

Some of these amplifiers did not accommodate the surge at power-up; some of the units had a surge resistor on one side of the power line, resembling a small rod wound with wire and a bimetallic strip between either of the ends. As the wire heated up, the bimetallic strip straightened out and shorted across the end terminals, applying full voltage to the power transformer primary.

To reliably suppress the surge when turning the amplifier on, I recommend inserting a three-terminal strip and a

PHOTO 5: Power supply panel mounted.



thyristor (see parts list) mounted in the same position on the chassis. The thyristor works when cold; a several ohm resistance slowly lowers as it warms. This damps a surge that can shorten the life of tube filaments (refer to the pictorial diagram and **Photo 5** for rear panel wiring, before and after).

PARTS LIST

8–10 100mF 450V electrolytic capacitors (Electronic Goldmine)
2–.27mF 630V film capacitors (AES Solen CFSD27-630)
2–.0047mF 630V film capacitors (AES Illinois 6-UD0047-630)
2–100k 2W metal film resistors (AES R-F100K)
2–22k 2W metal film resistors (AES R-F22K)
1–27k 2W metal film resistor (AES R-F27K)
1–1k 2W metal film resistor (AES R-F1K)
1–47mF 450V electrolytic (AES C-ET47-450)
1–47mF 350V electrolytic (AES C-ET47-350)
1–22mF 350V electrolytic (AES C-ET22-350)
1–100mF 160V electrolytic (AES C-ET100-160)
1–600V 1A silicon diode (AES P-01N4005)
1–SPST or DPST slide switch (optional)
1–Thyristor (optional)
1–47k 1W resistor (optional–bias range)
1–2k 2W pot (optional–bias range)

To test the HV power supply, bias supply, and filament voltages, make sure you have no tubes mounted in the amp. If possible, use a Variac to bring the AC supply up slowly, and monitor the DC voltage. When AC voltage is up to line voltage, the DC should be around +500V DC unloaded. The bias voltage should be around –50–60V DC, and the filaments should be 6–7V AC.

Mount the next section of the power supply on the opposite side of the upright panel in the amp. This includes the original 12k 10W resistor, a 1k 2W and 27k 2W, plus a 22µF 350V and two 47µF 450V electrolytics. Follow the schematic diagram and connect everything as necessary. This concludes the power supply rebuild.

AMP DESIGN

The amplifier rebuild starts with a careful check of the wiring, testing all the resistors, and replacing the coupling caps. (In my unit, I was not surprised to find a number of resistors that were open or off value; carbon composition resistors often drift in value as they age. When rebuilding or servicing any vintage piece of equipment, you should measure the value of most resistors and replace them as necessary.) In this W6, the 22k precision resistors were both open, as was the 27k resistor from the power supply, and the 47k –1W on the plate of V1 measured 20kΩ. I replaced the 22k precision units with perfectly matched 2W metal film resistors and replaced all the power supply resistors, most of which were significantly off value.

(I recently discovered an interesting problem with metal film resistors while repairing a vintage amplifier and replacing an open carbon film resistor with a new metal film. On power-up, the amplifier still did not function correctly, and when I measured the replacement, it also was open. I now measure each new component before and after I install it.)

There are four coupling capacitors in this amplifier design. I used .0047mF (4N7) 630 Illinois film caps for the .005s, and .27mF (270H) 630V Solens for the .25µF (250n) "monsters." (I replace all coupling capacitors in vintage equipment because they can be leaky,

a frequent problem with large-value coupling capacitors feeding the grids of output tubes, which results in the tubes failing to hold bias.)

When all the wiring and replacing is finished, it is time to place the tubes and test the fix. Insert all the tubes (use a NOS 12BH7 because of the issues with the early versions of W6 and stability and biasing of the 6550s), turn the bias pots to the fully counterclockwise position, and set the meter switch to off. There is no need to connect speakers or an audio source at this time. Power up the amplifier, and after the tubes have warmed up, connect your DVM to the chassis and to pin 5 of one of the 6550s. Adjust the bias control for that tube to indicate –50V on the DVM. Do the same thing to the other 6550. (If you are not able to set these voltages, you should measure all the voltage test points in the amplifier to determine the source of the problem.)

If the bias holds steady, check all the voltages listed in the Heathkit manual, which should measure to within 10% of their listed value. Check the accuracy of the built-in bias meter by setting the switch to each position and noting whether the meter deflects to the set mark (with your bias set by the DVM). The metering system in the W6 measures the current draw through the center taps of the output transformer from the high voltage supply. If your high voltage power supply differs from factory specs, the meter reading will not accurately reflect the actual bias current.

As described earlier, I lowered the plate voltage with a choke, but decided later to remove it to allow the correct use of the bias meter. You could also increase the capacitance of the voltage doubler section of the power supply and boost the voltage before the choke. Alternately, you could re-mark the meter face with the bias set at –50V as measured by your DVM. Do what works for you, but it is very important to check this bias setting often and compensate for the aging of the 12BH7.

If everything measures correctly, power down the unit, hook up a speaker and input source, and power it back up. You will be amazed at the ability of this amplifier to drive virtually any speaker system, and how cool the unit runs. After several hours of significant use in my shop, the transformers were only slightly warm to the touch; even the 6550s were touchable.

FINE-TUNING

With some trial use, I started having the same issues with bias adjustment as described earlier with the W6Ms. I decided that it made sense to modify this amp to the W-6A style by replacing the 56k 1W resistor between the bias pots and ground with a 47k 1W resistor in series with a 2k 2W pot. This simple modification allows an increase of the available bias voltage to balance aging tubes.

To avoid drilling extra holes in the original chassis, I removed the octal power socket from the back panel

SUPPLIERS

Antique Electronics Supply
www.tubesdandmore.com

Electronic Goldmine
www.goldmine-elec-products.com/home.asp

and made a plate to mount the new control. Run a pair of wires from the front-mounted bias controls where the 56k resistor is connected (necessary because this amplifier is wired with all the ground connections tied together and grounded only at the input connection). This prevents ground loops.

I also added another terminal strip and moved the filament windings and connecting wires to the strip. Because of the original wiring design, the filament winding from the power transformer goes to the back socket and the twisted wires go to the rest of the amplifier. It is also important to ground the center tap of the 6.3V winding. Failure to do this will result in 60Hz hum.

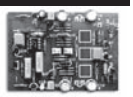
Though not a thing of beauty, the Behemoth, even after extended listening, is one of the most pleasing tube amps I have ever heard. It is extremely musical, and listening for long periods of time is effortless. Now I would like a second one, to listen in stereo! aX

* Keep in mind that domestic voltages in the US routinely vary with demand over the course of a day. Use a Variac to set bias properly.—ETD.

REFERENCES

1. Heathkit High Fidelity Amplifier Model W-6M Assembly and owners manual.
2. www.w7fg.com.
3. *Vacuum Tube Valley*, Issue 2, Fall 1995.

Bruce Brown *is a registered pharmacist, working in the medical research area for a major pharmaceutical company. He has been experimenting with electronics for over 30 years. His earliest interests involved tubes salvaged from castoff TVs. He serviced audio equipment during college. Over the years he has remained actively interested in electronics, building kits and "home brew" audio. Online auctions have stimulated new interests in restoring vintage audio equipment. Only recently has he become interested in writing articles to assist other hobbyists. His wife, Kristin, has been an invaluable help with editing his ramblings. He currently uses many restored and home-built audio amplifiers. He welcomes communication about vintage equipment and restorations. He can be reached at Tuninfork@aol.com.*



Loading Step-Up Transformers for MC Cartridges

Here is an easy way to load a moving-coil step-up transformer to match your cartridge load requirements.

By Cornelius R. Morton

For this project, all you need are two RCA f/m/f adaptors, mono to mono plug adaptor at Radio Shack, a handful of solder on RCA plugs (insulated), and a selection of resistors. **Photo 1** shows an adaptor and plugs for reference. The load resistor is soldered between the center pin and the ground of the RCA plug.

The adaptor is plugged into the MM phono input, the load plug (with the desired load resistor) into one of the female sockets on the adaptor, and the appropriate left or right cable from the MC transformer into the other. Then repeat for the other channel. Oh, that desired load resistor!

THE REST OF THE STORY

MC cartridges have loading requirements varying from 100Ω to 30Ω or less, while the standard MM input stage has an input impedance of 47kΩ. The MC step-up transformer has two functions. The first, and most obvious, is to step-up the output of the MC cartridge, between 0.5 and 0.05mV or less, to the 3 to 5mV level of a typical MM cartridge.

PHOTO 1: Adaptor and plugs.

The second function is to transform the high input impedance of the MM input to the relatively low impedance required by the MC cartridge. MC step-up transformers are usually rated in dBs of gain. Unless you get a transformer designed for your cartridge or one of the few expensive ones that incorporate a load select switch, how do you match everything up?

A look at an MC transformer will help you find an answer. The ratio of primary to secondary turns determines the voltage step-up of the transformer and is identified as "n." If the secondary has ten turns for every turn on the primary, then n = 10 and 1mV in results in 10mV out. When you take the voltages across the same impedance—in this case the preamp output—the power gain is equal to the square of the voltage gain, or n^2, which is also the impedance ratio of the transformer. Thus the transformer with a turns ratio, n, of 10 has an impedance ratio of 100, so 1Ω across the primary looks like 100Ω across the output or, conversely, 100Ω across the output looks like 1Ω across the input.

If the cartridge requires a load of 100Ω, then the resistance required across the secondary is 100 × 100Ω = 10kΩ. This is great, but the transformer is rated in dBs of gain! Well, the gain in dB is equal to 10×log(power gain) for the transformer with a power gain of 100, which is equal to 10×log(100) or 10 × 2 = 20dB.

To obtain the power gain or, more important, the impedance ratio of the transformer from the gain in dB, you simply divide the gain in dB by 10 (20/10 = 2) and use the resulting number as a power of 10, 10^2 = 100. For a gain of 26dB the impedance ratio is $10^{2.6}$ = 398.1. To properly load a MC cartridge, the resistance across the secondary must equal the desired cartridge load multiplied by the impedance ratio of the transformer.

Continued on p.49

TABLE 1.

Transformer	Transformer	Phono in	Load Resistance for Desired Input Z		
			Z in	*Z in*	*Z in*
Gain/db	Z ratio	47k	100Ω	50Ω	32Ω
21	125.9	373Ω	17.2kΩ	7.27kΩ	4.4kΩ
22	158.5	296.5	23.9k	9.53k	5.69k
23	199.5	236.6	34.7k	12.7k	7.39k
24	251.2	187.1	53.96k	17.1k	9.7k
25	316.2	148.6	96.2k	23.8k	12.9k
26	398	118	259.8k	34.5k	17.5k
27	501	93.8	——	53.6k	24.3k
28	631	74.5	——	96k	35.4k
29	794	59	——	340k	55.3k
30	1000	47	——	——	100k

Continued from p.16

The typical input impedance for a MM preamp stage is 47kΩ. Generally, inserting a resistance in parallel with the preamplifier input so that the resulting parallel combination equals the desired load times the impedance ratio will provide the desired results. **Table 1** lists gain in dB, impedance gain, the reflected load of the 47k input resistance, and load resistors required for cartridge loads of 100, 50, and 32Ω, for gains of 21 to 30dB in 1dB steps.

The raw input impedance of the MM input stage needs help until the transformer gain reaches 26dB, where the transformer input impedance is 118Ω and 93.8Ω at 27dB gain. Many MC cartridges list 100Ω or less, so the 118Ω may be a little high.

Table 1 lists gains in whole values; fractional values such as 21.3dB are easily interpolated. For example, 21.3 is 0.3 of the value between 21 and 22. For the Z ratio then, it would be 0.3×(158.5 – 125.9) + 125.9 = 9.78 + 125.9 = 135.68; the calculated value is 134.9 and the error is less than 1%.

Interpolation of loading resistor values for Z in values between those given will be fair up to a gain value of 24dB. You can determine accurate values by the following:

$$R_{load} = (1/(Z_{ratio} \times Z_{in}) - 1/47000)^{-1}$$

for example, if the desired Zin = 70Ω and the Zratio = 398, gain = 26dB

then

$$R_{load} = (1/(70 \times 398) - 1/47000)^{-1} = 68.413k\Omega$$

This is easier to do on a calculator than it is to write.

LOAD VALUES AND RESISTORS

• MC cartridges tend to have a slight high-frequency peak; therefore, a load slightly on the low side may provide better high-frequency control.

• 1% metal film resistors are recommended to maintain channel balance and noise floors.

• ¼W resistors are fine and easily fit the solder type RCA plugs. **Photo 1** shows a ½W in my plug, which was handy. *aX*

PARTS LIST

Gold Plated MonoAdaptor, Radio Shack 274-881, $4.99 ea., two required.

Insulated phono plugs (8), color-coded (red, blue, yellow and white), Radio Shack 274-321, $3.99 per package.

1% metal film resistors are available from Antique Electronics, Digi-Key, Mouser Electronics, and other suppliers. The gold-plated connectors look nice and do not tarnish. Gold does have a higher contact resistance than some of the standard platings, thus there is no electrical gain. So my connectors look nice.

The Sound Strobe

With this simple diagnostic tool, you can test the sound of your loudspeakers. . . without hauling them to the lab.

By Dennis Colin

Listening to these pulse signals, with six selectable spectrum shapes, is very revealing of loudspeaker time smear (driver "hangover," crossover misalignment, cabinet diffraction, and so on), frequency response colorations (particularly from resonances), and room effects such as discrete echoes, standing-wave bass modes, and acoustic colorations in general. You can hear quickly, and with detailed clarity, the speaker's degree of image focus, transient precision, and tonal neutrality.

OVERVIEW

Six pulse shapes are generated with a pulse repetition frequency (PRF) of 0.5-41.2Hz (see unit in **Photos 1–4**). The lower (rhythmic) rates allow hearing detailed decay patterns, while the audio rates (41.2Hz is the low "E" on a 4-string bass) provide harmonically-rich tones very sensitive to response and coherence anomalies. The selectable spectral distributions range from linear to non-resonantly shaped, allowing a focus on various frequency ranges. The impulse waveform and a linear ramp sawtooth are also provided at additional outputs, useful for oscilloscope triggering and X-Y plots.

The compact (8¼″ × 5¾″ × 3½″) low cost generator is powered by two rechargeable 9V batteries or an AC line "wall wart" (**Photos 5** and **6**); the batteries will power the unit for about 10 hours/charge.

APPLICATIONS

Listening to test signals is, of course, no substitute for music evaluation. Rather, these wideband coherent-transient pulses serve as an auditory diagnostic tool; repeatable signals allow sonic identification of a wide variety of speaker and room anomalies. The time-coherent nature of the pulses allows you to simultaneously hear and distinguish both the direct speaker output's image focus and a plethora of room effects. Walking around the room, you can identify reflective surfaces contributing colorations, discrete echoes, and so on.

An additional application is peak power testing for distortion and compression: with the short-duration (14.5µs) impulse waveform, you can even test a tweeter at 1kW peak, because even at the maximum PRF of 41.2Hz the average power is only 0.5W.

WAVEFORM DATA

Figure 1 shows the six waveforms, spectral distributions, pulse widths, and RMS to peak voltage ratios at two PRFs, 5Hz, and 40Hz. Squaring these

PHOTO 1: Sound Strobe front .

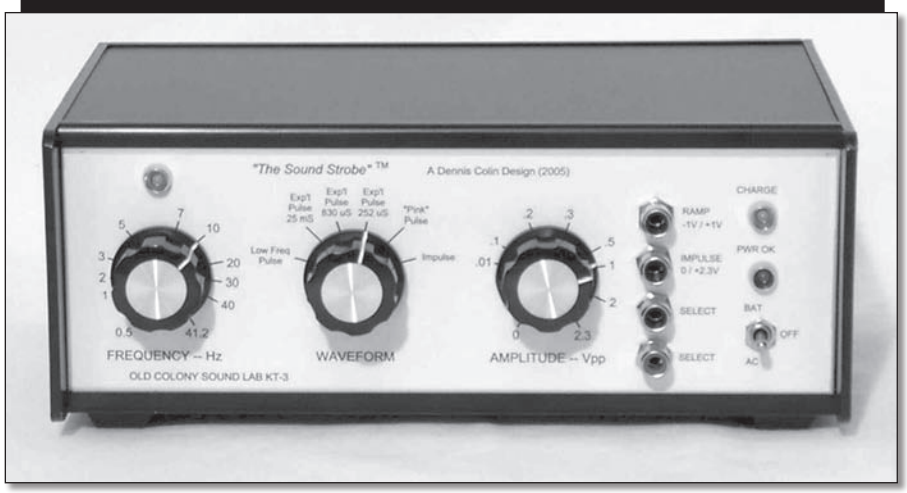


PHOTO 2: Rear view of unit.

FIGURE 1: Waveform data.

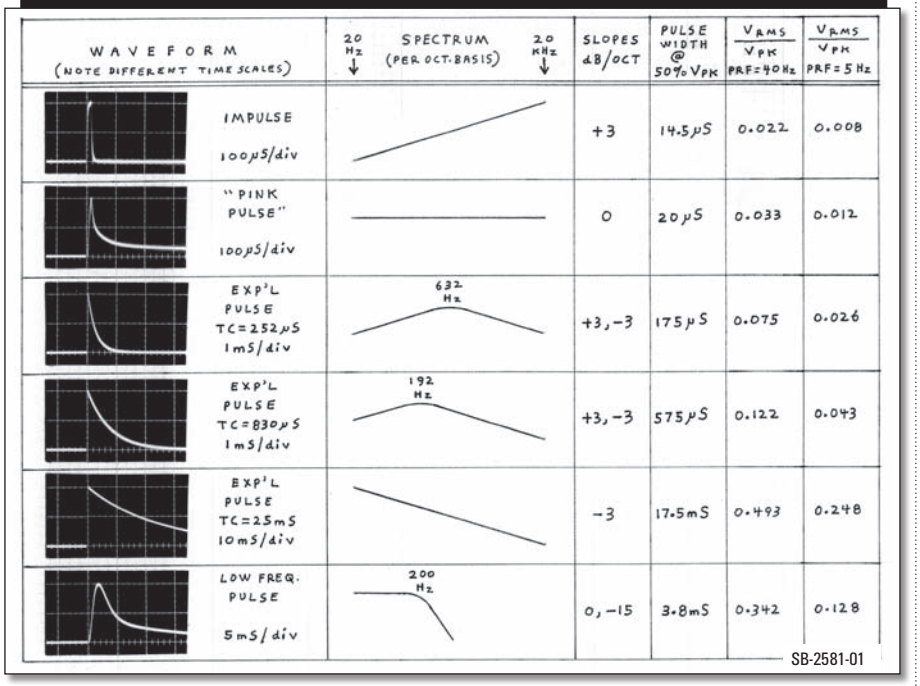
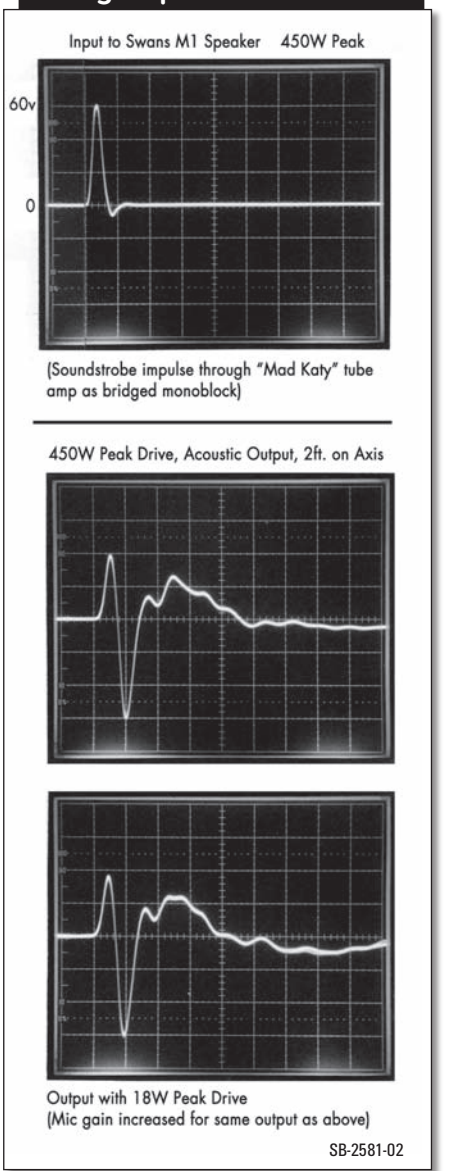


FIGURE 2: High power impulse testing of speaker.



ratios gives the ratios of average to peak power delivered to the speaker, useful for ensuring safe high-peak-power testing for compression and distortion. **Figure 2** shows an example of this high power testing. The impulse signal was amplified by "Mad Katy" (**Photo 7**), a 125W per channel stereo tube amp I designed. Driven to just below clipping in bridged monoblock mode, a peak impulse power of 450W was delivered to a Swans M1 speaker (average impedance about 8Ω). This excellent mini-monitor with ribbon tweeter (which I reviewed in *SB* 3/99) had no problem handling the 450W peak impulse, as shown by the very small change regarding the response with 18W peak power.

The 450W peak impulse, at the 20Hz PRF I used, produced an average power of only 0.11W. Perceived loudness was closer to the 113dB peak SPL than the 77dB average SPL. The sound was very "snappy," similar to that of electric sparks. It was very easy to distinguish the direct speaker output (crisp "snaps" with almost no tonal color) from the room sound (an enveloping "ocean" of thousands of pulse harmonics with tonality sustained by standing waves, and discrete pulse reflections originating from localizable reflecting surfaces).

When I moved about the room, the reverberation became a massive 3D "chorus effect" of moving pulses and changing overtone patterns, yet the speaker's directly-radiated impulses maintained the precise focus and tonal neutrality that I praised the Swans M1 for in the review.

The spectrum of a single impulse (for practical purposes, a unipolar pulse of shorter duration than a quarter cycle of the highest frequency of interest) is continuous and flat, with constant bandwidth (BW) analysis; that is, there's equal power per Hz BW across the band.

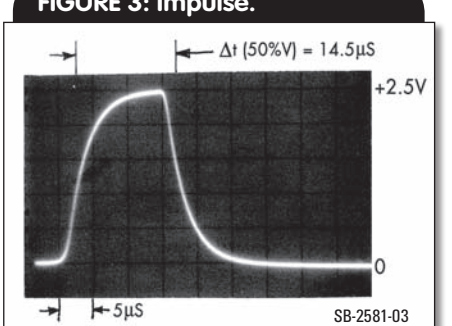
However, the ear analyzes on a frequency-proportional basis; that is, power per octave (or fraction thereof). Therefore, a flat *per-Hz* spectrum signal, such as an impulse or white noise, is perceived as having a 3dB/octave upward slope; that's why white noise sounds thin and "hissy." Conversely, pink noise sounds flat, spectrally balanced; its spectrum slopes downward (-3dB/octave) on a per-Hz analysis, but is flat on a per-octave basis. The spectra in **Fig. 1** are with per-octave analysis, the way we hear.

SPECIFIC WAVEFORM DESCRIPTIONS

1. Impulse

Figure 3 shows this on an expanded (5μs/div) time scale; pulse width at 50% Vpeak is 14.5μs; 10-90% rise and fall times are about 6μs. Compared with an ideal impulse, the spectrum is -1dB at 20kHz, and –3dB at 30kHz.

FIGURE 3: Impulse.



The sound (on a very neutral and coherent speaker) is very "snappy" and colorless like a small electric spark, but not as "sizzly bright"—that's because a small (<¼") spark radiates an acoustic doublet waveform: a differentiated impulse with spectrum sloped up 6dB/octave with reference to an impulse. (Lightning, a very

big spark, radiates nearly a step function sound wave, with bass to below 1Hz.)

The impulse, with its linear upward-sloping spectrum (to the ear), is best for hearing tweeter reflections, diffraction, and other anomalies of time coherence and HF tonal neutrality.

2. "Pink Pulse"

Figure 4 shows this on a range of time scales. This waveform has several interesting properties:

a) A flat spectrum on a per-octave basis (as the ear analyzes), like pink noise (hence the name).

b) The decay looks similar over a wide range of time scales.

FIGURE 4: "Pink pulse;" band limited, -3dB re ideal at 30kHz.

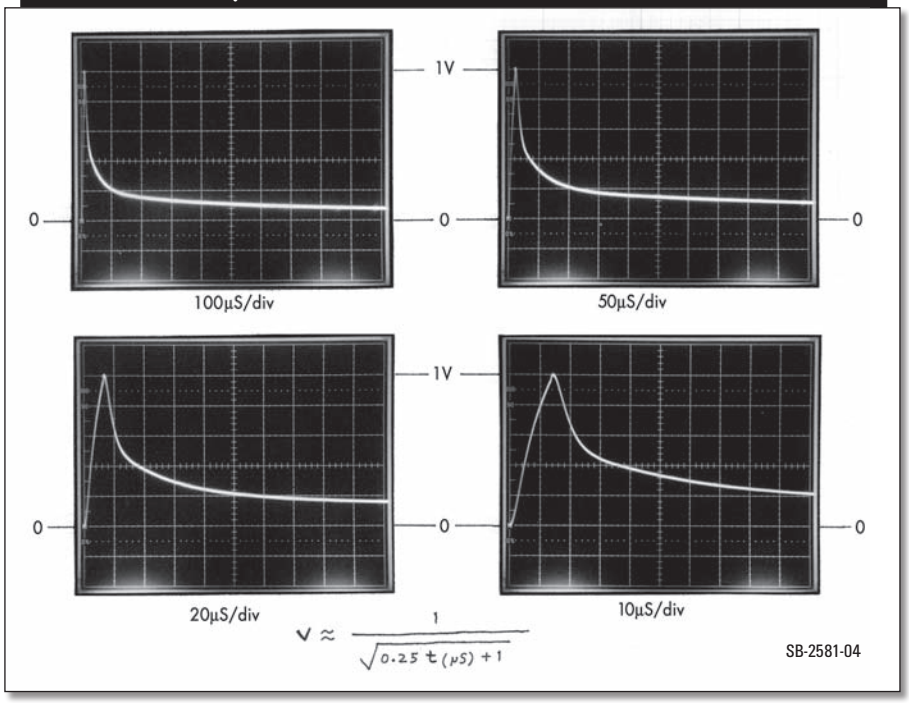
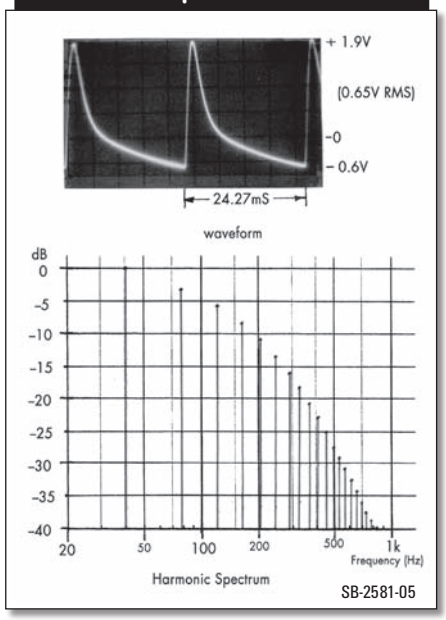


FIGURE 5: LF pulse at 41.2Hz PRF.



c) An ideal pink pulse (infinite BW) would initiate with a jump to infinity, and decay inversely proportional to the square root of time. The pulse shown has a -3dB rolloff at 30kHz.

d) With constant BW (e.g., per Hz) spectral analysis, such as is used in Laplace transforms, the (ideal) pink pulse is the only non-repeating waveform whose shape is identical to that of its spectrum: the spectral amplitude is proportional to 1/√frequency; that is, a slope of –3dB/octave with constant BW analysis (flat to the ear).

The sound has a sharp attack like the impulse, but has "full-bodied" midrange and bass, rather than the predominant HF "snap." It's basically a tonally-neutral full-band precision "click," extremely revealing of time and tonal response anomalies across the audio spectrum. Only with the very best speakers will this pulse maintain its pristine transient impact and lack of tonality.

3. Exponential pulse, 252µs time constant (TC)

As the sonically-perceived spectrum in **Fig. 1** shows, this pulse has a broad (non-resonant) peak at 632Hz, the logarithmic center of the audio band. This pulse doesn't sound "colored," but the midrange emphasis serves to best reveal midrange response anomalies, while maintaining enough perspective of lower and higher frequencies. The sound (on a good speaker, of course) could be described as a "fat click."

4. Exponential pulse, 830µs TC

This is similar to the previous pulse, but with the non-resonant peak at 192Hz. It's generated by 1st-order highpass filtering a step pulse, rolling it off below 192Hz (-3dB point). It's useful for hearing the step response of small speakers, without subjecting them to the excessive bass power of an unfiltered step pulse. It's also useful for evaluating mid-bass coherence and neutrality.

5. Exponential pulse, 25ms TC

This is basically a raw step pulse, the 25ms TC decay being the result of a 6Hz 1st-order LF rolloff. It's useful for evaluating full-band coherence and bass impact of large speakers. Caution: it's tempting to turn up the volume enough to feel the impact, rattle the walls, and so on, but with a high-powered amp, woofer damage is quite possible.

6. Low frequency pulse

As the spectrum in **Fig. 1** shows, this pulse doesn't have the rising deep bass energy of the previous pulse, but it's strongly filtered above 200Hz. The sound could be described as a non-resonant "thump," but sharper in precision than that word implies. The spectrum (again, to the ear) is flat from about 20-100Hz. This pulse is useful for hearing bass clarity, impact precision, and tonal neutrality.

Figure 5 shows the waveform and harmonic spectrum when this pulse is repeated at an audio rate, 41.2Hz (the low "E" on a 4-string bass). The rich but smoothly rolled-off harmonic content produces a pleasing, string-like bass tone that serves as a repeatable test of bass coherence and neutrality.

OSCILLATOR

U1A (latching comparator or "hysteretic switch") and U1B (integrator) form a linear-ramp sawtooth oscillator (**Fig. 6a**). Without D1 and R10, the output at U1B pin 7 would be a symmetrical triangle wave. But D1 conducts when U1A pin 1 is positive, producing a rapid fall time (13µs) at pin 7. Pot R32, R9, R11, and C1 determine the ramp charging time, and therefore the oscillator frequency.

The voltage at pin 1 pulses positive for about 13µs, and stays negative for the rest of the cycle. This waveform, through D2, R26, R27, and C16, produces the impulse signal selectable by S2. The pin 1 pulse also drives the circuitry from D3 to Q1, Q2, which flashes the green LED 1 (140mA for about 100µs, 1mA maximum average DC) in sync with the output pulses.

PINK PULSE SHAPER

The ramp reset step from U1 pin 7 has the step function's –6dB/octave spectrum with constant BW analysis, but –3dB/octave slope to the ear. So this is equalized with a +3dB/octave slope to achieve the desired flat auditory spectrum. Note: the impulse waveform could have been used with a –3dB/octave EQ, but for the same peak voltage limit, the impulse has much less energy.

R4 through R8 and C2 through C5 comprise a –3dB/octave impedance network, accurate within ±0.5dB from 10Hz–25kHz. Connected between the sawtooth step and the output op amp U2B inverting input pin 6 (when selector S2 is in position 5), the network's –3dB/octave impedance/frequency slope adds an upward +3dB/octave slope at the out-

PHOTO 3: Sound Strobe PCB.

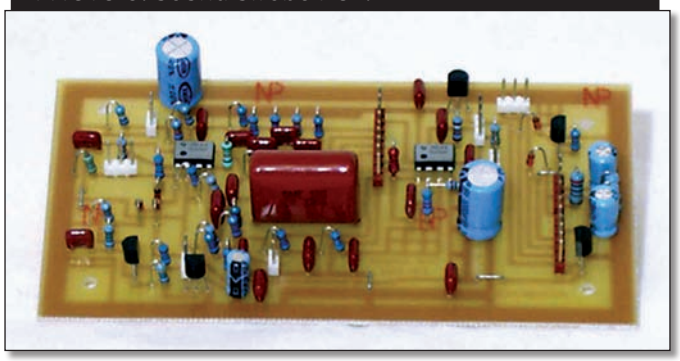
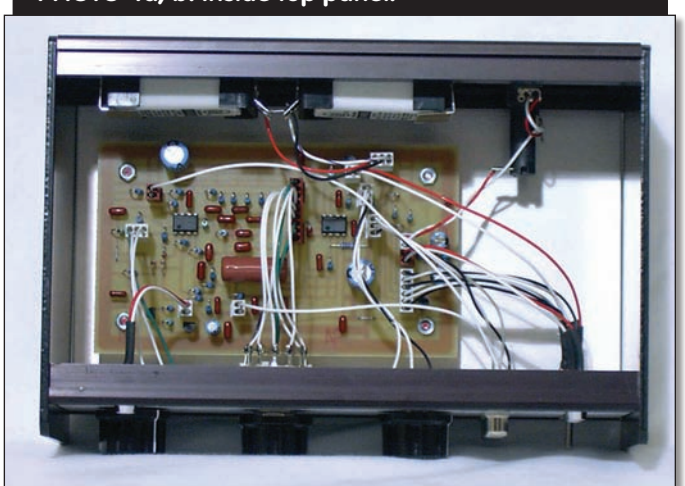


PHOTO 4a, b: Inside top panel.



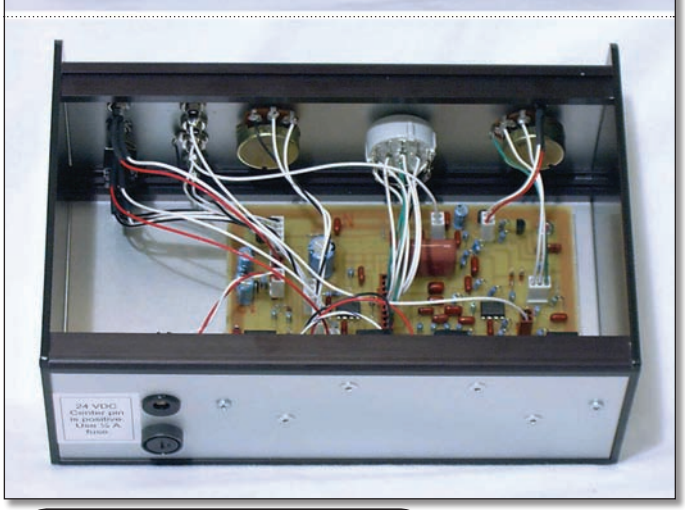


PHOTO 5: Battery power supply.

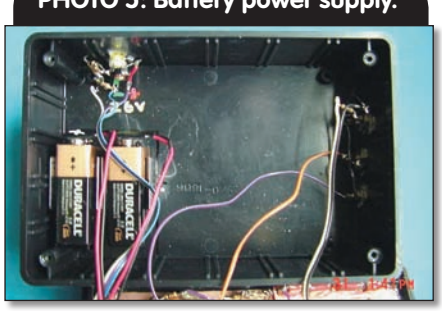


PHOTO 6: AC line "wall wart."

PHOTO 7: "Mad Katy" amp used for impulse power test.

put (U2 pin 7). **Figure 7** shows the 1kHz square wave (from an external generator) response of the shaping circuit.

EXPONENTIAL PULSES

C6, 7, and 8, with R14, 15, form first-order high-pass filters, determining the exponential decay time constants. While with constant BW analysis such pulses have a flat spectrum up to the HP filter corner frequency, then slope-down (-6dB/octave), to the ear the spectrum peaks at this frequency as shown in **Fig. 1**.

LF PULSE

The sawtooth wave is first sloped upward (+3dB/octave) by C9, 10 and R19, 20, 21, to produce a flat auditory spectrum; then it's low-pass-filtered above 200Hz by R22, 23, C11, 12, and C13, R18. The net LP filtered spectrum has a slope of about –15dB/octave. Consisting of cascaded first (and half)-order networks, the rolloff is nonresonant (sonically uncolored).

Note that in **Fig. 5** (harmonic spectrum of LF pulse at a PRF of 41.2Hz),

FIGURE 6a: Sound Strobe schematic.

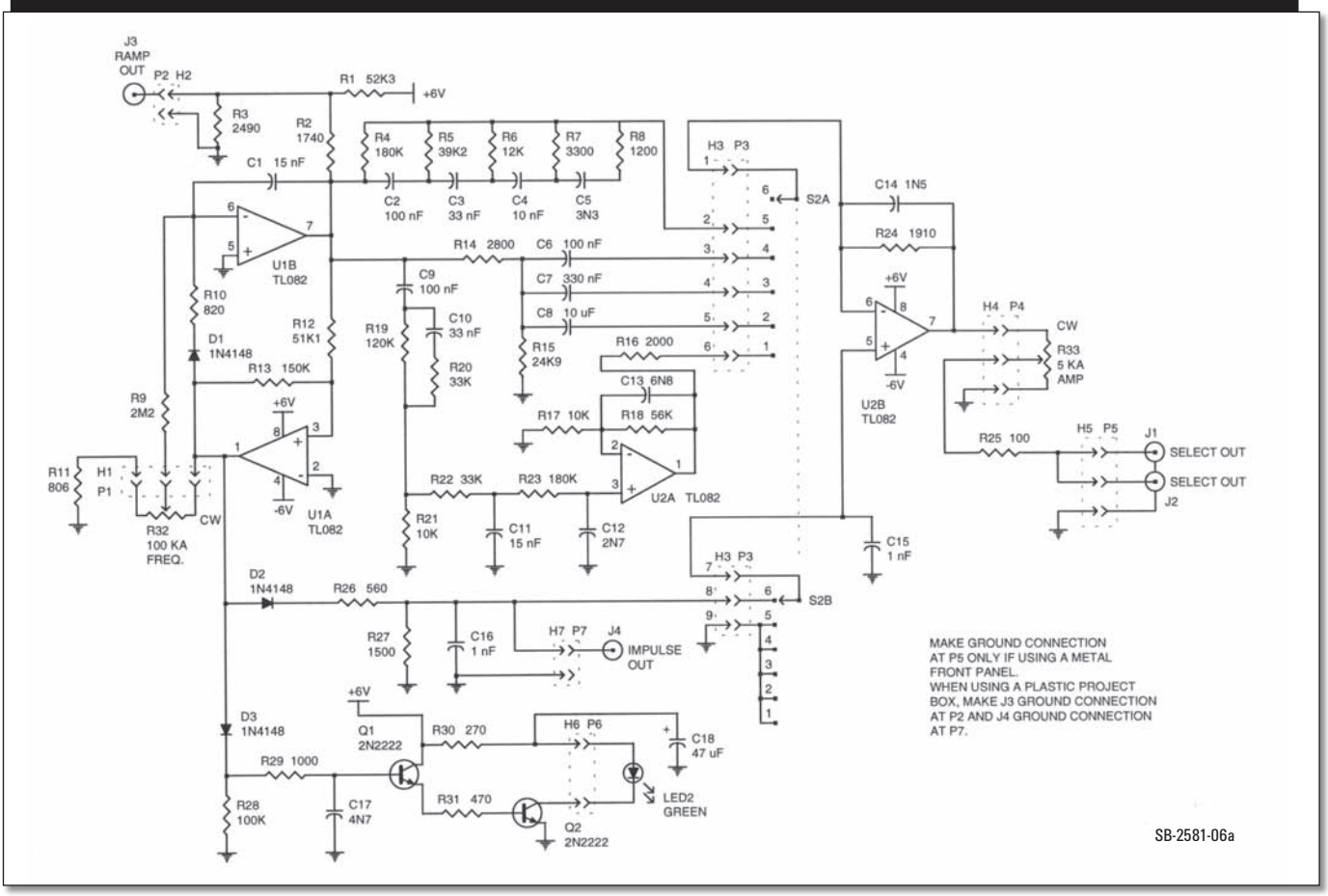
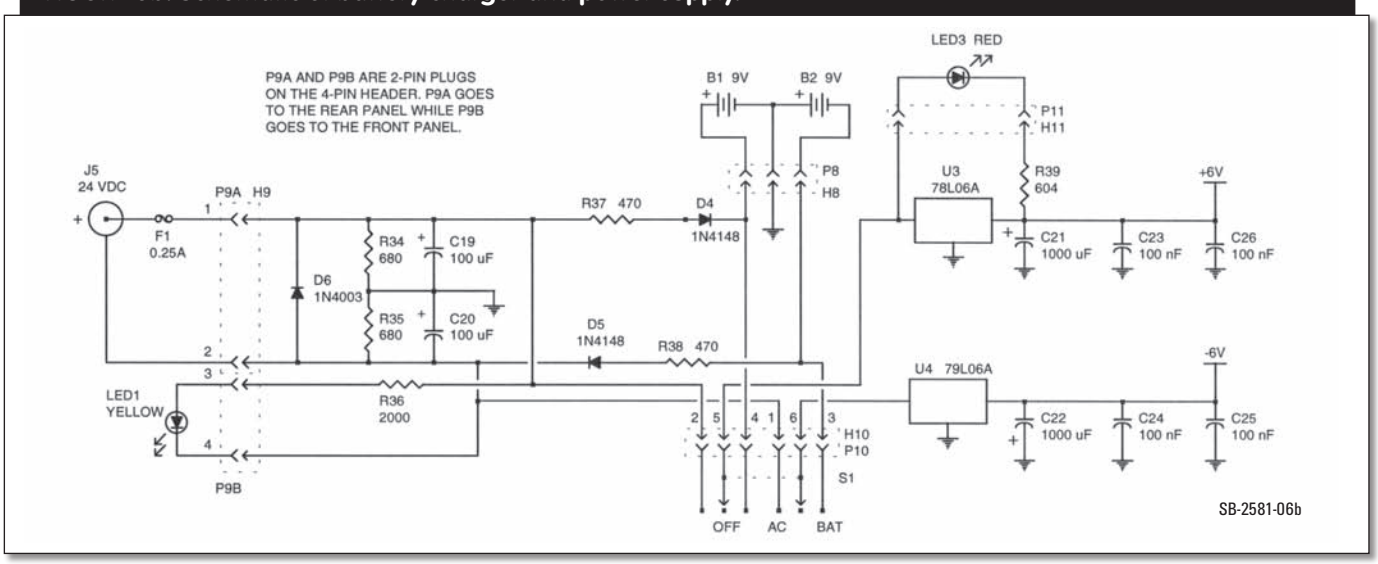


FIGURE 6b: Schematic of battery charger and power supply.



the first few harmonics decrease in amplitude by about 3dB/octave, while the single-pulse spectrum in **Fig. 1** is flat below 200Hz. This is because discrete spectral lines' amplitudes aren't subject to analysis bandwidth, unlike a spectral continuum such as noise or single events. But there's no contradiction: the harmonics (below the 200Hz filtering) *do* roll off at about 3dB/octave, but as frequency doubles, the *number* of harmonics *per octave* doubles, increasing the power per octave by a slope of 3dB/octave. So the net power/octave that the ear analyzes is approximately constant (below the 200Hz LP filtering).

SELECTOR SWITCH S2

In positions 1 through 5 (all waveforms except impulse), the output op amp's noninverting input is grounded by S2B, and the negative-going sawtooth step's shaped pulses are inverted for a positive-going output from U2B.

In S2 position 6 (impulse output), the impulse is fed to U2B noninverting input, while S2A opens

PARTS LIST

Reference	Value	Description	Manufacturer
R1	52K3	1%, ¼W, metal film	Mouser 271-52.3K
R2	1740	1%, ¼W, metal film	Mouser 271-1.74K
R3	2490	1%, ¼W, metal film	Mouser 271-2.49K
R4, R23	180K	1%, ¼W, metal film	Mouser 271-180K
R5	39K2	1%, ¼W, metal film	Mouser 271-39.2K
R6	12K	1%, ¼W, metal film	Mouser 271-12K
R7	3300	1%, ¼W, metal film	Mouser 271-3.3K
R8	1200	1%, ¼W, metal film	Mouser 271-1.2K
R9	2M2	1%, ¼W, metal film	Mouser 271-2.2M
R10	820	1%, ¼W, metal film	Mouser 271-820
R11	806	1%, ¼W, metal film	Mouser 271-806
R12	51K1	1%, ¼W, metal film	Mouser 271-51.1K
R13	150K	1%, ¼W, metal film	Mouser 271-150K
R14	2800	1%, ¼W, metal film	Mouser 271-2.8K
R15	24K9	1%, ¼W, metal film	Mouser 271-24.9K
R16	2000	1%, ¼W, metal film	Mouser 271-2K
R17, R21	10K	1%, ¼W, metal film	Mouser 271-10K
R18	56K	1%, ¼W, metal film	Mouser 271-56K
R19	120K	1%, ¼W, metal film	Mouser 271-120K
R20, R22	33K	1%, ¼W, metal film	Mouser 271-33K
R24	1910	1%, ¼W, metal film	Mouser 271-1.91K
R25	100	1%, ¼W, metal film	Mouser 271-100
R26	560	1%, ¼W, metal film	Mouser 271-560
R27	1500	1%, ¼W, metal film	Mouser 271-1.5K
R28	100K	1%, ¼W, metal film	Mouser 271-100K
R29	1000	1%, ¼W, metal film	Mouser 271-1K
R30	270	1%, ¼W, metal film	Mouser 271-270
R31	470	1%, ¼W, metal film	Mouser 271-470
R32	100K	Pot, audio taper	Mouser (Alpha) 31VJ501
R33	5000	Pot, audio taper	Mouser (Alpha) 31VJ305
R34, R35	680	1%, ½W, metal film	Mouser 273-680
R36	2000	1%, ½W, metal film	Mouser 273-2K
R37, R38	470	1%, ¼W, metal film	Mouser 271-470
R39	604	1%, ½W, metal film	Mouser 273-604
C1, C11	15nF	5%, 50V, polyester film	Digi-Key (Panasonic) P4584
C2, C6, C9, C23, C24, C25, C26	100nF	5%, 50V, polyester film	Digi-Key (Panasonic) P4525
C3, C10	33nF	5%, 50 V, polyester film	Digi-Key (Panasonic) P4569
C4	10nF	5%, 50V, polyester film	Digi-Key (Panasonic) P4582
C5	3n3	5%, 50V, polyester film	Digi-Key (Panasonic) P4557
C7	330nF	5%, 50V, polyester film	Digi-Key (Panasonic) P4549
C8	10µF	10%, 100V, polyester film	Digi-Key (Panasonic) EF1106
C12	2n7	5%, 50V, polyester film	Digi-Key (Panasonic) P4556
C13	6n8	5%, 50V, polyester film	Digi-Key (Panasonic) P4561
C14	1n5	5%, 50V, polyester film	Digi-Key (Panasonic) P4553
C15, C16	1nF	5%, 50V, polyester film	Digi-Key (Panasonic) P4551
C17	4n7	5%, 50V, polyester film	Digi-Key (Panasonic) P4559
C18	47µF	25V, radial electrolytic	Mouser (Xicon) 140-XRL25V47
C19, C20	100µF	25V, radial electrolytic	Mouser (Xicon) 140-XRL25V100
C21, C22	1000µF	25V, radial electrolytic	Mouser (Xicon) 140-XRL25V1000
U1, U2	TL082	Dual opamp, 8-pin DIP	Mouser (TI) 595-TL082ACP
U3	78L06A	+6V regulator, TO-92	Digi-Key (Panasonic) AN78L06
U4	79L06A	–6V regulator, TO-92	Digi-Key (Panasonic) AN79L06
D1, D2, D3, D4, D5	1N4148	Silicon diode	Mouser 1N4148MSCT
D6	1N4003	Silicon diode	Mouser 1N4003MSCT
LED1	Yellow LED Charge-on indicator		Lumax SSI-LXR1612YD (Digi-Key 67-1149)
LED2	Green LED Freq indicator		Lumax SSI-LXR1612GD (Digi-Key 67-1148)
LED3	Red LED Low-battery indicator		Lumax SSI-LXR1612ID (DigiKey 67-1147)
J1, J3, J4	Female, panel mount RCA, black		DGS (Mouser 161-1052)
J2	Female, panel mount RCA, red		DGS (Mouser 161-1053)
J5	2.5mm male, insulated, panel mount, power input connector		DGS (Mouser 163-4303)
P1, P4, P5, P8	3-pin shell with terminal pins		Molex WM2012
P2, P6, P7, P9A, P9B, P11	2-pin shell with terminal pins		Molex WM2011
P3	9-pin shell with terminal pins		Molex WM2018
P10	6-pin shell with terminal pins (39) terminal pins for the Molex shells		Molex WM2015 Molex WM2200
H1, H4, H5, H8	3-pin male header		Molex WM4001

(continues →)

the inverting input. Thus, U2B acts as a voltage follower for the (positive) impulse waveform.

C14 and C15 serve to attenuate ultrasonic pulse components (extending to above 100kHz).

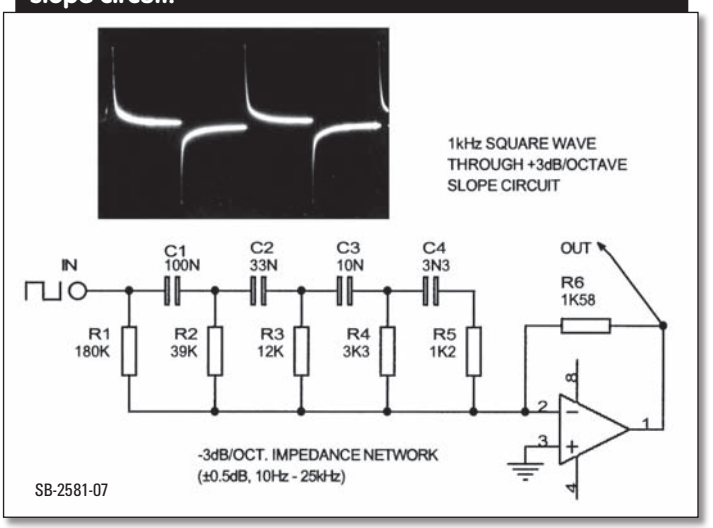
POWER SUPPLY (FIG. 6b)

S1 selects "AC," "off," or "BAT." The suppy selected is regulated to ±6V by U3, U4, and so on. Batteries B1, B2 (9V NiCd) are charged with a tapered current averaging 10mA.

LED1 (yellow) indicates AC power. LED3 (red), labled "Pwr on," indicates both that and sufficient charge of batteries is used. Note the connection of LED3 differentially across regulator U3, biased by R39 to glow only with sufficient positive battery charge; and because the positive battery (B1) has a slightly higher discharge rate than the negative one (B2)—about 12mA versus 10mA—than LED3 indicates satisfactory charge of both batteries.

Note that the current through LED3 contributes to the positive load current; therefore it is "free." Also, its brightness varies with charge level, serving as a "fuel gauge."

FIGURE 7: 1kHz square wave through +3dB/octave slope circuit.



PARTS LIST (cont.)

H2, H6, H7, H11	2-pin male header	Molex WM4000
H3	9-pin male header	Molex WM4007
H9	4-pin male header	Molex WM4002
H10	6-pin male header	Molex WM4004
	(Headers, shells and terminal pins are available from Mouser, Digi-Key, and others.)	
S1	DPDT miniature toggle switch	Digi-Key (Carling) 432-1149
S2	Rotary switch, 2-pole, 6-positions,	
	break-before-make	Mouser (Lorin) 105-14572
	Enclosure, 8 × 5 × 3″	Sescom MC-6A
	Circuit board	
	Front panel	Metalphoto of Cincinnati
	Stick-on label for rear panel	TDL M121R
	(3) knobs	Mouser (Eagle Plastics) 45KN017
	Panel mount 5mm fuse holder	Mouser (Littlefuse) 576-03455LS1H
	5mm fuse, 0.25A	Digi-Key (Wickmann) WK1035
	(2) battery holders	Mouser (Keystone) 534-1295
	(2) battery retaining clips	TDL M401BRC
	(4) aluminum pop rivets, 1/8″ dia × 1/8″ grip (to attach battery holders to rear panel)	
	(2) 4-40 × 5/16″ machine screws	Mouser H343
	(4) 4-40 × 5/8″ machine screws	Mouser H348
	(2) #4 lock washers	Mouser H236
	(6) 4-40 hex nuts	Mouser H216
	(4) 3/8″ long nylon spacers, tapped 4-40	Mouser (Eagle Plastics) 561-L4.375
	(4) Stick-on plastic feet for enclosure bottom	Mouser (3M) 517-SJ-5023BK
	(13′) #24 AWG stranded wire	
	(2) 9V rechargeable NiCd batteries	JDR NB9V or equal
	Wall DC power supply, 24V DC at 100mA	Mouser 412-124013

USING THE SOUND STROBE

The pulses provided can be very revealing of transient and tonal reproduction characteristics, but first you must know what the pulses *should* sound like. The short answer is "as sharp, spatially focused, and non-tonal (uncolored) as possible." Think of hearing an electric spark—there's no tonality, but an immediate "Snap!" that you can locate to within inches from across the room. It's so precisely focused that you instinctively jump to look for it.

The best full-range speakers, electrostatics, planar magnetics, or small cones, can come close to this precision. Many multi-way speakers fall short, and the Sound Strobe pulses are heard spread out in time and space. Admittedly this is unfair; we listen to music, not electronic pulses (except for synthesizers); and (at least acoustic) music doesn't have transients nearly as sharp as electronic impulses. But if you aren't completely satisfied with your speakers (present or contemplated), then these test pulses can help you diagnose and improve their sonic fidelity.

The longer answer to knowing what to listen for is this: I strongly recommend playing all six pulses on the very best speakers available, from friends, willing retailers, and so on. I also recommend good headphones, with their excellent time coherence. Of course, their lack of room acoustics makes for an unfair comparison with speakers; however, the pulses' sharp transient nature, coupled with the Haas (precedence) effect, allows you to distinguish a speaker's direct output from the room reverberation.

It's also informative to listen to the pulses on a wide variety of speakers, from the best to the worst available. Correlating the pulse sound with music will then build up an "experience base" of various transient, focus, and tonal anomalies. Then you can use the repeatable high-resolution pulses to quickly identify problem areas by direct listening (unlike trying to interpret electronic measurements), and without the immense variability of musical sounds.

Of course, the goal is to reproduce this musical variety with the utmost clarity and fidelity. With a modest "experience base," the Sound Strobe can be a useful tool for that goal. Plus, it's fun to experiment with—you'll hear a very intriguing array of speaker and room (distinguishable) pulse-response sounds that can directly relate to the reproduction of music.

I'd like to thank Ron Tipton (*info@tdl-tech.com*) for his excellent packaging, board layout, and finetuning for production. aX

The Sound Strobe is available in kit or assembled units from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 888-924-9465, e-mail: custserv@audioXpress.com

Ed Frias EFE Kit by Speaker City

Assembled and Reviewed by Duncan, Colin, and Nancy MacArthur

ED FRIAS EFE KIT
Speaker City, USA
115 South Victory Blvd.
Burbank, CA 91502
1-818-846-9921
Fax: 1-818-846-1009
www.speakercity.com

List price: $170 per speaker
System: two-way rear-tuned bass reflex
Woofer: imported 6.5″ polypropylene cone
Tweeter: imported 1″ textile soft dome
Impedance: 8Ω nominal
Power rating: 100W RMS continuous, 150W maximum.
Frequency response (±2dB), 45Hz-20kHz
Sensitivity: 89dB (2.83V/1m)
Dimensions: 13″ × 8.75″ × 9.75″ (H × W × D)
Available finishes: black oak, golden oak, and red oak

Ed Frias of EFE Technology designed this small two-way speaker for monitoring, music listening, and HT applications. The EFE design is available as a kit from several distributors; the Speaker City version is reviewed here. The Ed Frias EFE Kit (henceforward called the "EFE") uses a 1″ dome tweeter and a 6½″ mid-woofer and a redesigned (May 2003) crossover network; much information about the crossover is available at the GR research website (**http://www.gr-research.com/edspeaker.htm**).

The hardest aspect of building this kit was persuading the distributor to send it to us. The EFE arrived on our doorstep six weeks after the initial order, after three phone calls from Ed Dell to Speaker City.

The kit came in three large (but not huge) boxes (**Photo 1**). Two boxes contained the finished enclosures, double-boxed with the outer box separated from the inner box by a couple of inches of random foam "cutoffs." The foam was correctly distributed and provided good shipping protection, and the shipping boxes clearly are meant to serve as shipping containers for the finished systems.

All the remaining components were well packaged in the third box. (The crossovers were so well packed that we spent a fair bit of time getting into them.)

COMPONENTS

The first instruction made sense: It told us to take an inventory of the supplied parts. Unfortunately, we had no parts list. Our kit consisted of the following (**Photo 2**):

- 1 page of assembly instructions
- 2 finished enclosures with grilles
- 2 woofers
- 2 tweeters
- 2 tweeter gaskets
- 1 bag fibrous stuffing
- 1 accessory bag containing:
  - 2 input terminal cups

PHOTO 1: The Speaker City EFE kits came well packed in three cardboard boxes (cats not included).

PHOTO 2: All components contained in the EFE speaker kits.

- black Phillips-head screws (10 long and 21 short)
- quick-connect terminals (17 wide, 3 medium, and 9 narrow)
- 2 lengths of 18-gauge "speaker wire"

The EFE cabinets (**Photo 3**) appear well constructed and finished. The port tubes are pre-mounted into the rear of the enclosure, and the grilles are pre-assembled. Our pair was finished in "red oak" veneer with square corners.

Although the red-stained oak doesn't imitate a more expensive wood veneer, it offers a welcome alternative to the usual light oak or black oak finish options. The overall level of "fit 'n' finish" is very good for a speaker in this price range. A curious aspect of these cabinets is that while the input terminal cups are recessed in the rear of the enclosure, the drivers aren't recessed into the front.

Peerless manufactures both drivers used in the EFE design: type 105DT2608 for the tweeters and X176WR3308 for the woofers. These drivers are of high quality and undoubtedly contribute to the good sound of the finished system. The woofers have integral gaskets, while the tweeters have separate gaskets. All four drivers were packed well to prevent damage during shipping.

The crossovers (**Photo 4**) supplied with the Speaker City kits are fully assembled and mounted on small boards (the boards are not printed circuit boards). All connections are made with solderless connectors. The crossover parts are good (but not superb) quality. All capacitors are film-type and the coils are air-core, although two of these seem to be mounted with metal bolts through the center.

These crossovers, described on the Speaker City website, appear to use a first-order filter with impedance compensation for the woofer and a second-order filter for the tweeter. The drivers are connected "in phase." The crossover schematic on the website shows multiple

PHOTO 3: Close-up of one of the EFE cabinets.

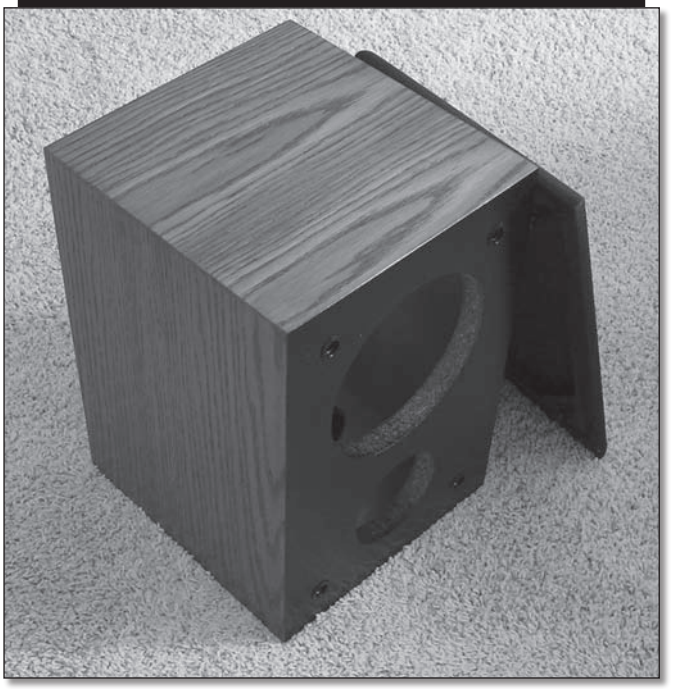
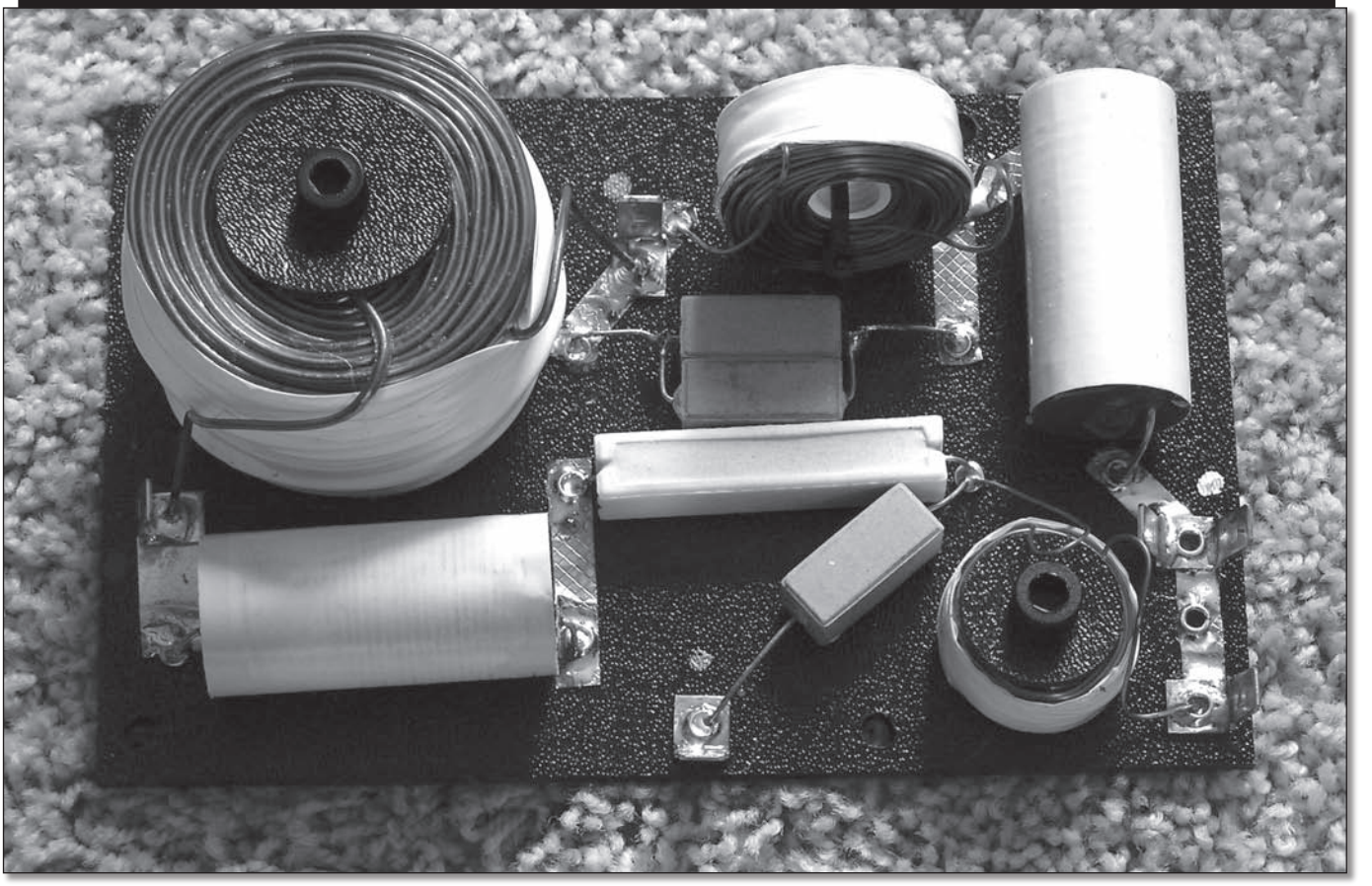


PHOTO 4: Close-up of the assembled crossover from the Speaker City EFE kit.



resistors to set the level of the tweeter, but these resistors were not included in the kit. Otherwise, the schematic seems to be representative of the assembled crossovers. However, the mechanical drawing from the website bears little resemblance to the crossovers in our kit.

This inconsistency in the mechanical drawing leads indirectly to one of our major issues with this kit. Although the input, woofer, and tweeter connections to the crossover are color-coded, the color-coding is explained only on the *bottom* of the crossover board, not in the instructions. Once you mount the crossovers into the cabinets, you no longer have access to this information. Because the website drawing is not correct, it cannot be used as a reference.

As noted above, the kit provides odd numbers of screws and connectors, and you must choose which are most appropriate for each application. (In our case the kit contained plenty of each type.) The internal wiring appears to be standard 18-gauge "speaker wire," which is paired and color-coded and probably is appropriate for speakers in this price range. The input terminal cups contain gold-plated "5-way" connectors, which are of a higher quality than the other small parts. In short, the EFE kit includes all parts (along with some extras) needed to assemble these speakers.

KIT ASSEMBLY

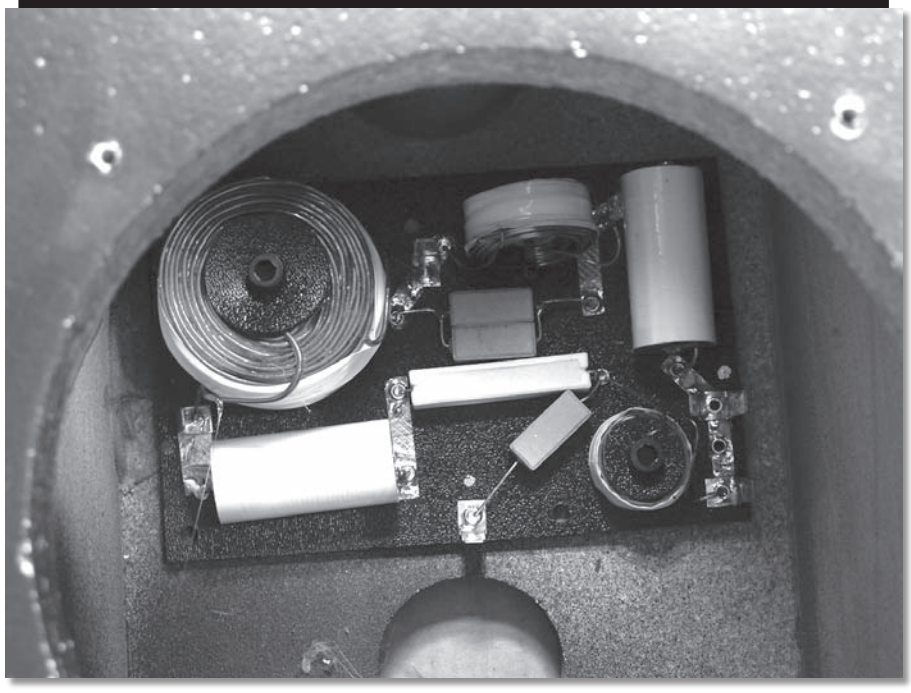
The assembly instructions provided by Speaker City consist of a single page that appears to be generic for all kits made by Speaker City. This instruction sheet has a list of tools and ten instructions that can best be described as an outline for building this speaker. Where appropriate, we have included the instruction numbers in our description of the assembly process. In several cases we suggest an alternative order for the steps.

The instructions list four tools required for assembling these kits: cordless drill, measuring tape, soldering pencil, and solder. Of these items, Colin and I needed only the first, but we also used a marking awl, hot melt glue gun, screwdriver, solderless crimping tool, and a wire stripper. None of these tools are particularly exotic, but a neophyte kit builder might not have them on hand.

Although the inclusion of a soldering pencil in the tools list suggests the intention to solder the wires onto the terminals, the supplied insulated terminals are not appropriate for solder connections. We used a crimping tool instead (also not a perfect solution).

As mentioned above, taking an inventory (Instruction 1) is always a good first step. In this case this step should include a little planning and a complete reading

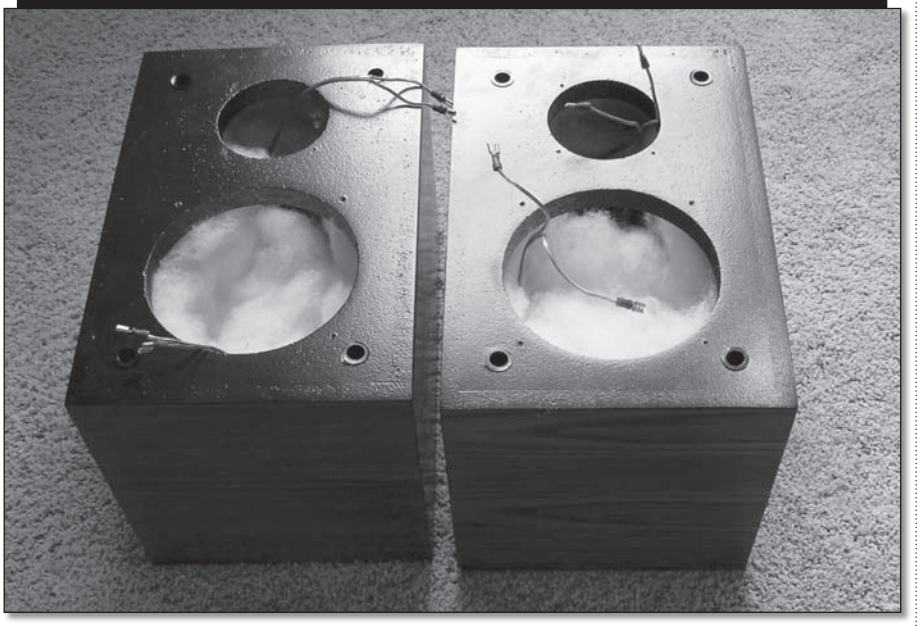
PHOTO 5: The crossover boards are mounted with hot melt glue on the real inner wall of the cabinet between the terminal cup and the port.



of the instructions to avoid any potential pitfalls. Any assembly of a pre-finished speaker requires a soft, clean, assembly location (Instruction 2) to avoid marring the finish of the enclosures. We found a carpeted floor to be ideal.

Because the supplied instructions do not specify which screws and connectors are appropriate for each use, you should plan how best to utilize the supplied hardware. In our case, we used the long screws to mount the woofer and the shorter screws to mount the tweeter and input terminal cup. Although we could have mounted the crossover board using the shorter screws, we chose to use hot melt glue instead. (Glue is easier to use in a confined space.) We chose the wide connectors for the crossover and woofer, the medium connector (plus one wide one) for the input terminals, and the narrow connectors for the tweeter.

PHOTO 6: The EFE cabinets following stuffing but before the drivers are attached. Note the driver connection wires sticking out of the stuffing.



We drilled all the pilot holes (Instruction 5) for both drivers and the input cup before mounting any electronic components in the enclosures. We used a ⅛″ drill bit for the woofer and 3⁄32″ for the tweeter and terminal cup. Rather than blowing off the excess wood particles as Instruction 5 suggests, we used a vacuum cleaner to avoid distributing the particles around the workspace.

We mounted the crossover (Instruction 3) using hot melt glue on the back wall of the enclosure (**Photo 5**). This location lets you keep all of the wire lengths reasonable, although, given that the wire is not precut, other locations were possible.

After physically mounting the crossover board, we attached the input terminals to the crossover using a technique that worked well for all connections. Simply strip one end of a long piece of speaker wire and crimp two wide terminals onto the end of the wire. Then thread the wire through the input cup hole and attach the terminals to the appropriate lugs on the crossover board. At this point we cut the cable long enough to conveniently reach the cup resting outside the box. Then we terminated the free end of the cable, attached it to the input terminals, and screwed the terminal cup into place.

This termination technique allows you to remove the terminal cup if required, ensures that the minimum reasonable length of wire is used, and avoids the need to measure wire lengths. By cutting all cables using this procedure, we ended up using less than half the supplied wire. At this point we attached the woofer and tweeter wires to the crossover (first half of Instruction 8), cut the wires to the appropriate length, and left them hanging out of the correct openings in the front panel.

Next we stuffed the speaker enclosures (Instruction 6). The supplied insulation material was in the form of a 1½″ thick blanket about 2′ wide. We cut 1′ by 1′ squares, punched holes for the woofer and tweeter wires, and pushed the mate-

rial into the back of the enclosures over the crossover. A second 1′ by 2′ piece was wrapped around the inside of the enclosure to cover the sides. We stuffed fist-sized pieces of the remaining material into each enclosure until a firm, but not packed, density was reached (**Photo 6**).

This procedure used only about two-thirds of the supplied insulation material. It's possible that the boxes are incorrectly tuned; further experimentation probably would be helpful. The only hitch with Instruction 7 involved placing gaskets around the driver mounting holes: only the tweeters have separate gaskets.

PHOTO 7: The completed EFE loudspeakers ready for listening.

Once the box is stuffed, the other end to the wires coming from the crossover to the drivers can be terminated and attached to the correct driver (second part of Instruction 8). We terminated all four wires, then attached one driver at a time and screwed them into the holes in the cabinets (Instruction 9). Finally we replaced the grilles and connected the finished speaker systems to an amplifier (Instruction 10). As always, we started with the volume turned all the way down and turned it up slowly, listening for problems. Step-by-step instructions from the distributor would be helpful, but until those instructions appear, we'd recommend the following assembly order: plan, 1, 2, 5, 3, 4, 8a, 6, 7, 8b, 9, 10, enjoy.

OPERATIONAL ISSUES

Once complete (**Photo 7**), the EFEs didn't resemble "kit" speakers, aside from their lack of a manufacturer's logo. Both speakers performed flawlessly immediately after construction (Speaker City recommends a 10- to 16-hour break-in

period prior to any serious listening).

The only difficulty in using the EFEs was the small size of the input terminal cups. Getting fingers into the cup to tighten or remove speaker cables wasn't easy. Fat speaker cables exacerbated the problem, but it recurred to some degree with every type of connector other than banana plugs.

We listened to the EFEs as part of three different setups. System 1 was a music-only system using a Manley StingRay integrated tube amplifier. System 2 was a home theater powered by an Onkyo TX SR601 receiver; the EFE speakers were used in the front channels. System 3 was a computer–based system with an AudioSource AMP-1 amplifier. We used System 1 to judge the ultimate ability of the EFEs, System 2 to judge the applicability to HT, and System 3 to judge the effectiveness of the EFEs as a step up from an entry-level speaker (Madisound Sledgeling, in this case).

LISTENING CRITIQUE

As with previous listening critiques, we auditioned the EFE using tracks from the *Hi-Fi News and Record Review Test Disk III* (track 2: Parry's "Jerusalem," track 4: Vivaldi's trumpet concerto, tracks 5 and 6: excerpts from Prokofiev's "Peter and the Wolf," track 7: Purcell's "Welcome, Welcome Glorious Morn," track 10: a Corkhill percussion piece, and track 14: Rio Napo RSS demo), as well as a wide variety of other music.[1] We judged imaging (between the speakers), naturalness with human voice, and bass extension (for a 6½″ speaker) to be strengths of the EFE, and height of image, congestion with massed instruments, and occasional treble harshness to be its weaknesses.

By far the greatest strength of the EFE lies in the lateral imaging it produces between the speakers. The voices and instruments on every track auditioned were placed with pinpoint accuracy; the words "strong imaging" appear and reappear in our listening notes. Although making a general comparison is inappropriate, lateral imaging is the single area where the EFEs surpassed the far more expensive Genesis 400 speakers normally resident in System 1. (Note: the EFE's ability in this area doesn't extend much beyond the physical location of the speakers.)

Another significant virtue of the EFEs was the naturalness they brought to vocal reproduction. This characteristic was particularly evident with voices that we know well in a live context (the MacArthur Road and Stabat Mater recordings), but also with voices that we know less well (the Comedian Harmonists, Cowboy Junkies, and Flatt & Scruggs disks). The accuracy of midrange reproduction was also evidenced by the "you are there" naturalness of the vibraharp on the Modern Jazz Quartet pieces.

The amount of bass produced, particularly on the kick drums of the BROTHER recording, is impressive for a single 6½″ driver. Of course, it's no match for dual 8″or 12″drivers, but it easily beats the single 6½″ in the Sledgeling. The mid-bass seemed a little loose on certain recordings such as *MacArthur Road*. (This characteristic may be due to the recording or to the stuffing issue mentioned earlier.)

Unfortunately, for $170 you're not going to get it all. Along with the sterling lateral imaging mentioned before, you get extremely restricted vertical imaging and ambiance. Although the EFEs were placed

on relatively tall speaker stands (28"), many performers sounded like munchkins playing under a low ceiling. This effect was less pronounced on certain recordings (piano music, for example) but was obvious with vocal groups.

Although EFEs performed well with unaccompanied or lightly accompanied voices and single instruments, they became slightly muddied with large ensembles. The Charlie Daniels recording, for example, sounded slightly smeared. To be fair, this CD is a "difficult" recording that sounds terrible on many systems. Although the EFE's performance in this area wasn't perfect, it was acceptable.

The EFE loudspeakers lacked transient detail when compared with live music or even with their more expensive electromechanical brethren. With the Beethoven piano sonatas, the tone of Brendel's piano is spot on, but the lack of crisp transients lets you know that you're listening to a recording.

The high-frequency response of the EFEs is variable. Some instruments, such as the brushed cymbals in *Blues on Bach*, sound marvelous, while others, such as the Vivaldi trumpets, sound downright painful.

FINAL THOUGHTS

DM: This Speaker City kit strikes me as more a complete collection of parts than a finished kit. With one trivial exception (a solderless connector), all components required to assemble a pair of Ed Frias EFE speakers are included. If the builder has sufficient experience and a reasonable collection of DIY hand tools, I would certainly recommend the Ed Frias EFE design as being musical and well worth the money. This kit represents a very significant step up from entry-level speaker systems as well as being well suited to home theater applications. I would be curious as to how the Speaker City version stacks up against the same design supplied by other manufacturers.

NM: I'd like to emphasize that this kit is appropriate for experienced speaker builders only. The EFEs weren't Colin's first kit, but he would have had trouble building them without help from his father. These speakers were musical and non-fatiguing, and we all enjoyed listening to them.

CM: The EFEs are a significant step up from my current speakers. I wouldn't hesitate to buy them.

1. RECORDINGS USED FOR LISTENING

Beethoven, Ludwig van, *Beethoven Piano Sonatas: "Pathetique," "Moonlight," and "Appassionata,"* Alfred Brendel, pianist, Philips compact disk 411 470-2.

BROTHER, *This Way Up*, Rhubarb Records compact disk RR07CD.

The Charlie Daniels Band, *Fiddle Fire*, Blue Hat Records compact disk BLH-9703-2.

The Comedian Harmonists, *The Comedian Harmonists*, Hannibal compact disk HNCD 1445.

Cowboy Junkies, *The Trinity Session*, BMG Canada compact disk 8568-2-R.

Dvorak, Antonin, *Stabat Mater*, Jiri Belohlavek, Supraphon compact disk SU 3311-2 231.

Dylan, Bob, *Highway 61 Revisited*, DCC Compact Classics compact disk GZS-1021.

Flatt & Scruggs, *On Foggy Mountain*, Sony compact disk A 18071.

Hi-Fi News and Record Review, *Test Disk III*, compact disk HFN 020

Jethro Tull, *Thick as a Brick*, Mobile Fidelity compact disk UDCD 510.

MacArthur, Dan, Margaret, Megan, and Gary, *MacArthur Road*, Front Hall compact disk FHR 040CD.

The Modern Jazz Quartet, *Blues on Bach*, Atlantic compact disk 1652-2.

Strauss, Richard, *Also Sprach Zarathustra*, Andre Previn, Telarc CD-80167.

Tchaikovsky, Peter Ilyich, *Tchaikovsky 1812,* Erich Kunzel., Telarc CD-80041.

Watson, Doc and Merle, *Pickin' the Blues*, Analogue Productions compact disk CAPFG 026.

This review marks the first audio article for Colin, whom we first met as a four-year-old in "Building a Toy Soldering Iron" (*Speaker Builder*, 3/97, p. 20). His parents, Duncan and Nancy, have written for *audioXpress, Speaker Builder, Glass Audio, Audio Electronics, Audio Amateur,* and *Hi-Fi News and Record Review*. The MacArthurs live in northern New Mexico. —*Eds.* aX

Peerless/Morel Mini-Monitors

By John and Sandra Schubel

The Peerless/Morel Mini-Monitors proved to be a musically superior monitor speaker system that can be purchased in kit form and easily assembled (**Photo 1**).

The kit was delivered in two boxes, the first containing the two speaker enclosures, and the second the drivers and components for building the crossover networks. The speaker enclosures were securely packed in Styrofoam. The second box, containing the components, was actually a box within a box, again providing protection from shipping damage. **Photo 2** shows the packed speaker enclosures.

ASSEMBLY

Photo 3 shows the speaker enclosures before the addition of any components. **Photo 4** shows the components that were included in the second box.

The instructions for assembling the kit were provided in the form of a stapled set of 8½ × 11″ sheets. It is advisable to read these instructions before assembling the kit.

The first steps are to install a port tube and mount the speaker terminals to the rear of the enclosures. This requires driving the terminals into pre-drilled holes in the enclosure. This is an interference fit, and the terminals must be set with a hammer (I used a rubber mallet). **Photo 5** shows the rear of a speaker enclosure with these components installed. The inside ends of the terminals are threaded, and you install a series of nuts, washers, and wiring tabs on the terminals from the inside of the cabinet.

I did this, and all work on the enclosures, on a soft surface to prevent scratching the finish. Later, during listening tests, the finish proved to be soft, because scratches appeared where I had placed the speakers on wood speaker stands. As a point of reference, I have used these stands with my NHT speakers without experiencing any scratches. So special care is required!

The next step is to cut the foam provided in the kit to size, and then glue

PHOTO 1: Assembled mini-monitors.

PHOTO 2: Packed speakers.

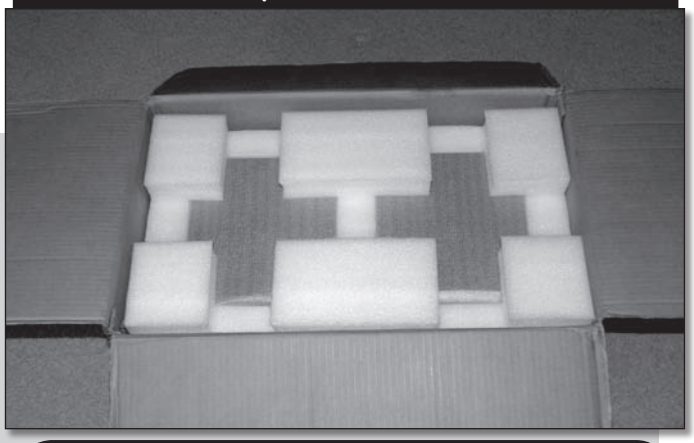


PHOTO 3: Before assembly.

the foam to the inside of the speaker cabinet. Dimensions for cutting the foam are not provided in the instructions, so you should measure the interior of the speaker enclosures before cutting. I chose a carpenter's square and a box knife to cut the material to size. The instructions offer the option of using the adhesive provided (Loctite clear household silicone adhesive/sealant, part no 30808) or hot glue. I chose the adhesive, which gave me more working time to carefully position the foam.

DRIVER INSTALLATION

The next step involves installing the drivers in the baffles. The Peerless 850108 5½″ woofer was provided with a gasket, so mounting this speaker was no more difficult than driving four screws into the baffle. The Morel MDT-20 1-1/8″ tweeter required the installation of gasket material, which was provided in the form of a roll of caulk. The instructions require you to roll the caulk into a bead ¼″ thick. The illustration in the instructions showed this bead molded to the driver frame very close to the magnet.

I placed the tweeter in the mounting hole provided in the baffle and found that there was a fair amount of clearance between the mounting hole and the magnet; if I placed the bead where the instructions suggested, it would not contact the mounting surface. Also, the caulk provided was a flat bead, already ¼″ thick. I therefore chose to apply the caulk as provided to the tweeter frame at a radius just inside that of the driver's screwholes.

The first time I tried this approach, the caulk still did not seal against the mounting surface in the baffle in some spots. I moved the caulk further out on the driver frame to be sure contact was made all the way around. **Photo 6** shows the caulk applied to the driver frame. I installed the tweeter and tightened the screws lightly until the face of the speaker was just flush with the surface of the baffle.

The assembly instructions suggest that it is usual for the drivers to be positioned so that the electrical connections are pointing downward. As I proceeded with the installation, it became obvious that the electrical cables to connect the drivers to the crossover networks were sufficiently short that the assembly was much easier if the driver's electrical connections were positioned toward the center of the baffle. I assembled one of the speakers in this manner.

PHOTO 4: Kit parts.

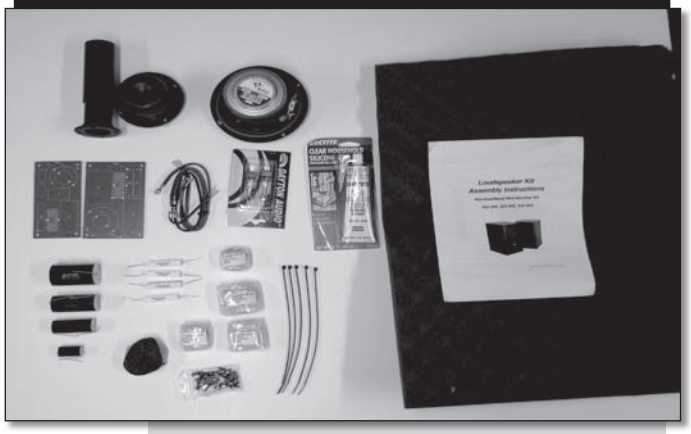


PHOTO 5: Rear of enclosure with components installed.

CROSSOVERS

The next step was to assemble the crossover networks. There are two circuit boards for each speaker—one each for the woofer and for the tweeter. It is best to lay out the components for each board prior to assembly.

Each printed circuit board is marked with the component identifier (R1, C1, L1, and so on), but the components themselves are marked with their electrical characteristic (10Ω, and so on). You must refer to either the schematic or to the photograph that appears later in the instructions in order to associate the components with their respective positions on the circuit boards. **Photos 7** and **8** show the two assembled crossover networks.

The instructions suggest that you glue the electrical components to the circuit boards using either the adhesive provided or hot glue. I chose hot glue in this case in order to speed the assembly process.

Once the boards are assembled, the instructions wisely suggest that you connect the drivers and the crossover networks together outside of the enclosure—using the cables provided—to assure that the speaker systems work. I found a missing solder connection, and making the repair after the speaker had been assembled would have been very difficult with the circuit boards glued into place.

The next step in the assembly is to glue the two circuit boards into the rear compartment of the speaker enclosure. **Photo 9** shows the crossover networks installed in the enclosure. I recommend that you practice getting the circuit boards into the rear of the enclosure, because the port tube makes this awkward. The instructions again offer the alternative of using hot glue. I chose to use the adhesive provided, as it takes time to get the circuit boards into the rear of the enclosure. I would have preferred a different installation method, so that future maintenance of the boards could be performed.

Once you have glued the crossover networks into place, you can attach the cables from the crossover networks to the speaker terminals and from the crossover networks to the drivers, and screw the baffle into place. **Photo 10** shows the speakers connected to the crossover networks and ready for final assembly.

PHOTO 6: Caulk applied to driver frame.

Assembly is completed by carefully setting the baffle into the speaker enclosure and fastening it with the screws provided. I recommend testing the speakers again before installing the speaker grilles, which are provided with four sockets that you must press into the screw holes in the baffle. Once the sockets are in place, it will be difficult to remove them.

LISTENING IMPRESSIONS

We played the speakers for a while using Kenny Daverne's *One Hour Tonight*, Music Masters 5003-2-C, on a HHB Compact Disk Recorder model CDR800, feeding a NAD Model 214 power amplifier via a passive control amplifier. We noted during this break-in period that the clarinet seemed more strident than that experienced with our NHT speakers, and the bass seemed subdued. Our impressions of the Peerless/Morel Mini-Monitors would soon change, perhaps indicating the importance of a break-in period. As is our custom, we listened to all program material on both the Peerless/Morel Mini-Monitors and on a pair of NHT Model 1.3s.

Our music source was *Hi-Fi News and Record Review*'s *CD Test Disk III*, in addition to selections from our private collection. Our initial observations, following the break-in period, were that the Peerless/Morels are very transparent; that is, there is not a sense that the sound is coming from the direction of the speakers. We also noted that treble response did not diminish when listening to these speakers off-axis. The soundstage did appear more distant than we would prefer, and the bass response was diminished both in level and in low-frequency response, compared to that perceived with the NHT speakers.

Because this can easily be compensated for through the use of a control amplifier or pairing with a subwoofer, we consider this a minor observation. The speakers reward you instead with very smooth and clear mid- and high-frequency response and the ability to handle loud and complex passages with great clarity. We never thought these speakers were harsh or strained in their reproduction of sound.

We listened to *Peter and the Wolf*. The voice of the narrator was clear, without the coloration to which we are accustomed. The oboe was clean and expressive, with great variety of tone. We found the flute to be more "edgy" than we were accustomed to with our NHTs. Similarly, the clarinet was less "sweet."

Horns were more exciting on the NHTs, but we concluded that it was probably coloration by the NHTs, and the Peerless/Morels were likely more accurate. Tympani and drums are not exciting, but probably more accurate with the Peerless/

PHOTO 7: Woofer crossover.

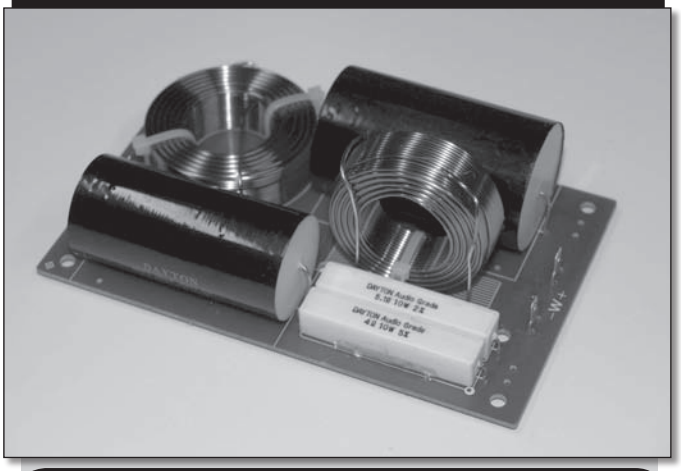
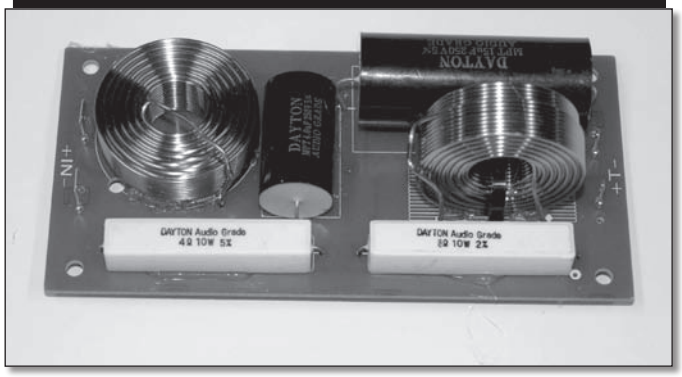


PHOTO 8: Tweeter crossover.



Morels. This impression again may be due to the perceived weaker response of the Peerless/Morels at lower frequencies. In general the soundstage seemed distant. The Peerless/Morels seem to emphasize percussion instruments.

We were divided on the assessment of the Peerless/Morels regarding the musicality of the brass. John thought that the brass stood out more clearly on the Peerless/Morels and were well balanced with the other instruments, as compared to the NHTs. Sandra preferred the balance obtained with the NHTs.

As we moved on to the last part of *Peter and the Wolf* (track 6), we both agreed that the Peerless/Morels were less strident and handled loud passages beautifully. We did find, however, that there was an emphasis on higher frequencies, such as those produced by triangles. Sandra believed that there were more "layers" in the sound on the Peerless/Morels than we had experienced with other speakers, and we both agreed that the instrumentation seemed well balanced with these speakers.

Purcell's *Welcome Welcome* was reproduced beautifully. The harpsichord was well reproduced, and the bass and tenor voices were clear and in perfect balance. Perhaps the bass voices are reproduced at a slightly lower level than the tenors. The female voices seemed closer to the

PHOTO 9: Crossovers installed in enclosure.

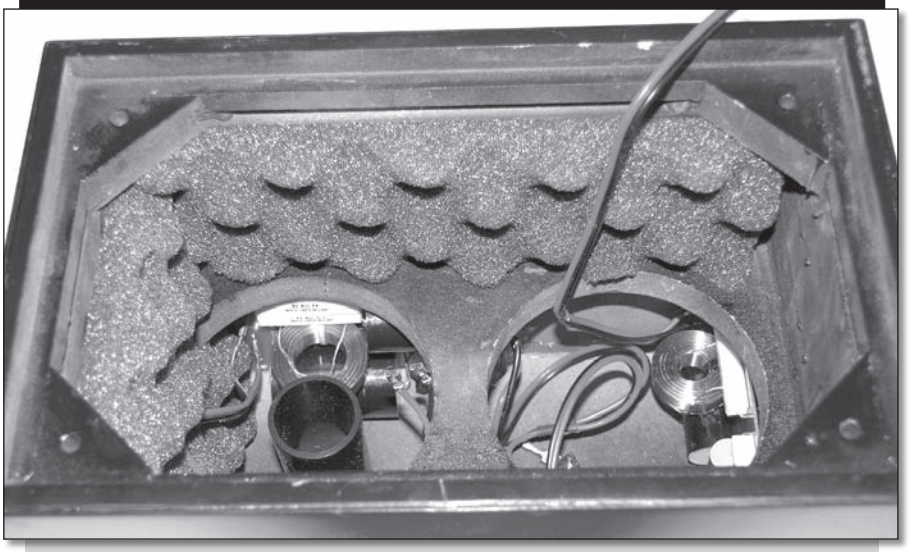
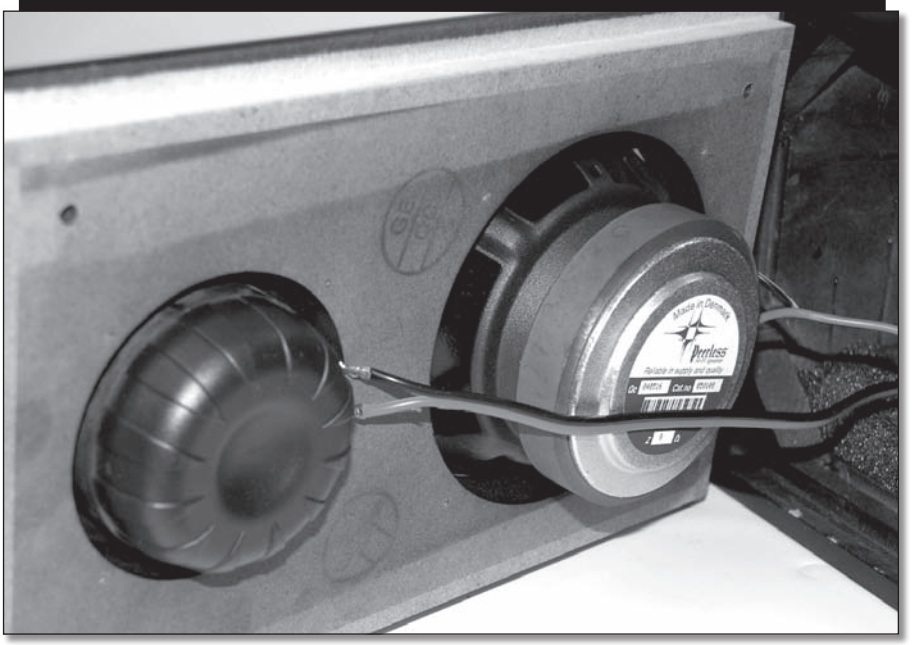


PHOTO 10: Cables connected.



soundstage, as might be expected with the typical arrangement of voices in a choir. John believed they were a little too bright, but Sandra was happy with the balance.

We both appreciated that we could hear each vocal part clearly. Sandra, who is not a fan of the harpsichord, commented that the instrument had a light sound that was less incessant than on other speakers.

Our final selection from the *CD Test Disk III* was Korkhill's *Five Short Pieces/ Prince Consort Percussion*. Sandra found that there was an ethereal quality to these pieces on the Peerless/Morels, with more separation between the silences and the very unusual percussion sounds. John particularly liked the clarity of the cymbal crashes.

We moved on to our private collection. We returned to Kenny Daverne's *One Hour Tonight*, and played the second track, *Pretty Baby*. Now that the drivers were well broken in, we preferred the sound of the Peerless/Morels. Cymbals were clearer, with less coloration of the guitar. The sound of the clarinet was sweet.

We continued to the next track, *Comes Love*, and found the sound of the Peerless/Morels to be more sterile, but we have come to perceive that the sound is actually more accurate. When Kenny does his runs on the clarinet, the Peerless/Morels handle them cleanly.

We moved on to an Empire Brass album, *Braggin' in Brass*, Telarc CD-80249. We selected *Bugle Boy from Company D* for this audition. This album is rich in low frequencies, and a subwoofer is essential for maximum enjoyment. The Peerless/Morels seemed weak in bass compared to our NHT 1.3s, but the Peerless/Morels sounded more natural and controlled.

The bass did not satisfy John, who wondered whether a graphic equalizer with a mild boost below 250Hz might not improve the experience. So, in what is clearly not an apple-to-apples test, John took the CD to a SONY DVP-S7700 DVD player connected to a SONY STR-DA555-ES receiver, reset the level using a Realistic sound pressure level meter, and kicked up the bass by 5dB. He was pleased with the result.

Returning to the original test configuration, we played Erik Satie's *Gymnopedie No.1* as recorded by Leonard Slatkin and the Saint Louis Symphony Orchestra, Telarc CD-80059. We found the oboe and flute to be delightfully reproduced, and the bass natural, although diminished. We continued on to *Gymnopedie No. 3*, and were again pleased. The sound of the harp is clear and natural, the violins well reproduced, and the oboe and gong pleasing.

Finally we played Bach's *Brandenburg Concerto No. 1* as recorded by the Academy of St. Martin-in-the Fields, conducted by Neville Marriner, Philips 400-076-2. We found the sound to be transparent, although we thought the soundstage was distant. The violins and harp are cleanly reproduced, and appear to be in the foreground of the orchestra. There is less emphasis on the wind instruments than with the NHT 1.3s, but that may be the nature of the recording, given the common placement of wind instruments in an orchestra. The more delicate passages of the *Concerto* were particularly pleasing to listen to. The violins soared on the high passages.

CONCLUSION

The Peerless/Morel Mini-Monitor Kit 302-900 is easily assembled from the high-quality components provided. Care is required in its assembly not to damage the tweeter's soft dome, which extends forward of the frame. John also found the gloss black coating on the speaker enclosure to be very soft and subject to scratching when placed on a wooden speaker stand. Our only reservation is with the method for mounting the crossover networks.

The sound quality of the speaker is outstanding, with excellent imaging, transparent sound, and good off-axis performance. The soundstage seemed to be a little farther back than with other speakers, but the placement of instruments on the soundstage was excellent. We would love to hear these speakers coupled with a good subwoofer.

The Peerless/Morel Mini-Monitor kit (which comes in black, cherry, or beech) is available from Parts Express (725 Pleasant Valley Drive, Springboro, OH 45066, 513-743-3000, Fax 513-743-1677, e-mail *sales@partsexpress.com*) aX

Dayton Euro Series Monitor

By Ken and Julie Ketler

PARTS EXPRESS
725 Pleasant Valley Drive
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Part # 302-920 (glossy black), 302-922 (cherry), and 302-924 (beech)
Frequency Response: 45–25,000Hz
Impedance: 6Ω
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Power Handling: 75W RMS
Dimensions: 14″H × 8.5″W × 12.5″D
Weight: 60 lbs.
Price: $446.39

Available from Parts Express is a sleek and sexy two-way speaker kit, which is available in cherry, beech, or glossy black finishes (**Photo 1**). When the glossy black review model showed up at my door, I immediately spotted a potential image problem. You see, upon opening the three shipping containers (two each with a speaker enclosure and one with all remaining components), I realized that many guys might be forced to fight off the temptation to get a new tattoo, or perhaps an earring, or maybe trade in their partner for a younger version. These Euros are like sports cars for speaker builders: small, shiny, and fast.

The Dayton Euro Series two-way reflex-loaded system utilizes an excellent and well-matched set of loudspeaker drivers including a 7″ paper composite woofer and "slightly horn-loaded" textile dome tweeter (**Photo 1**). This kit also allows you to construct a 24dB/octave crossover operating at 2kHz. The enclosures are each a compact 0.5ft^3 and built like solid, well-braced ¾″ medium-density fiberboard battleships!

Each driver is recessed into its enclosure, and all edges of the front baffles are rounded to help smooth out high-frequency diffraction effects, and lend the Dayton Euro Series every possible advantage for great imaging characteristics. The rather compact size of this system makes it perfect for front surround speakers, main stereo music speakers, or even multimedia use, although this may be pushing it a bit.

PHOTO 1: Assembled units shown with optional grille.

CONTENTS AND ASSEMBLY

The enclosures were packed very well in separate boxes with form-fitted foam pads. Because I witnessed the delivery *gentle*man actually drop them down approximately 4′ from his truck (*thwack!*) right onto my driveway, I can say with a high level of assurance that this packaging is effective; there were no signs of even a scratch when I opened the boxes.

All other parts including drivers, crossover boards and components, bind-

PHOTO 2: Euro Series textile tweeter and 7″ woofer. Note the random "grain" of the woofer cone in order to break up vibration modes and keep the midrange frequencies controlled.

ing posts, "egg crate" sound damping foam, adjustable reflex ports, silicone adhesive, cable ties, screws, speaker sealing caulk, and an 11-page assembly manual were also housed in their own shipping container. All you need to supply is solder and your own basic tools.

The assembly manual is very complete including a nice parts inventory list and suggestions for workspace considerations to get you started. The length of each 2″ port tube is adjustable, so you need to measure and glue them to length before you screw them into the cabinet. The binding posts are ready to be gently hammered into pre-drilled holes and are very clearly outlined in the manual. Any newcomers to speaker building will want to break any family traditions of *not* reading directions, or else!

The damping foam is packed as two separate sheets, and you will need to measure and slice it into appropriately sized pieces using scissors or a sharp utility knife. To avoid over-damping the speakers with too much foam, Parts Express suggests alternating it between walls, or leaving opposite walls uncovered rather than lining the entire enclosure. A new kit builder may not have different types of adhesives around, but you can sparingly use some of the supplied silicone adhesive to mount the foam.

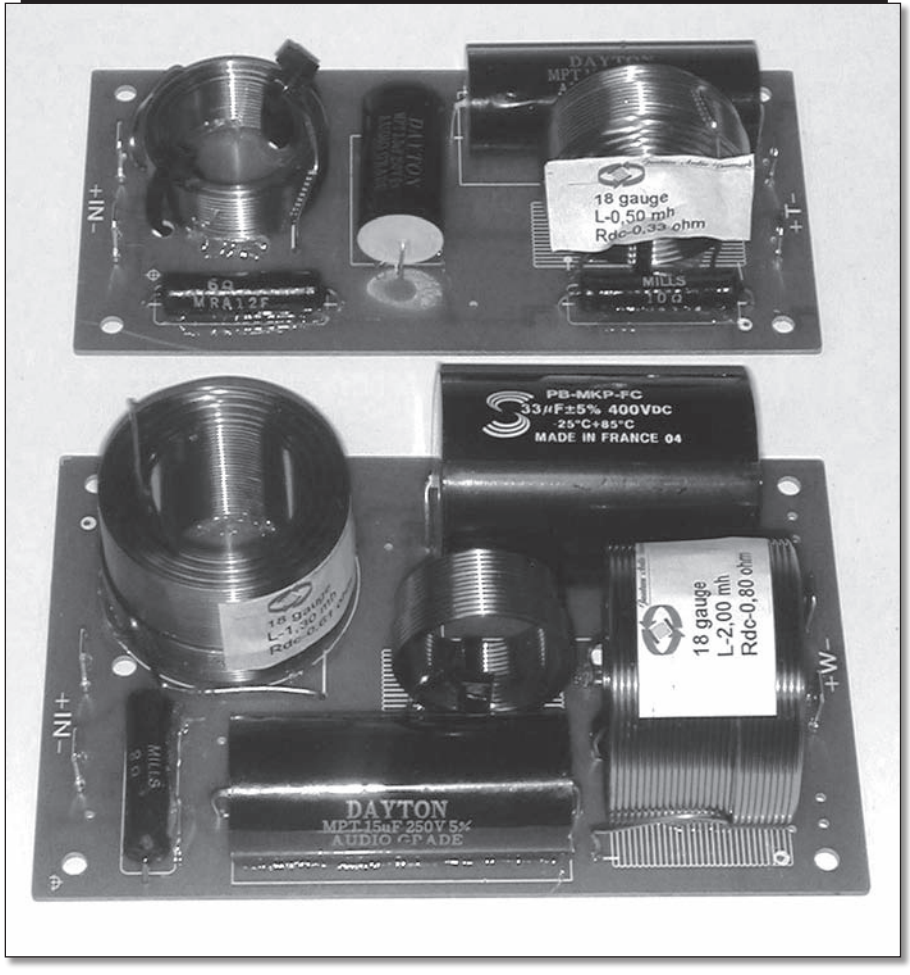
Crossover assembly is very straightforward. The boards are neatly laid out and easy to work with (**Photo 3**). All components are well marked and shown in a schematic; however, the photos in the manual are "generic" and are not representative of this particular model. For an absolute novice, this could cause some frustration, but with a little patience, most should be able to decode the schematic diagram quite readily.

Builders are not only given a brief lesson in soldering, but also instructed to bind down all components using silicone and also cable ties for all inductors—solid! After the crossover assembly section, the manual helps you through a quick check of your work before you permanently install them inside the cabinet. This is a particularly useful lesson for new builders.

The next step is to install the crossovers. Given the 14″ × 8.5″ × 12.5″ dimensions of the box, including a vertical brace in the middle of the enclosure, this wasn't the easiest task. An easy detail to miss is that the front baffle must be removed before mounting anything to the inside of the enclosure. Without doing so, I imagine it might be nearly impossible to reach parts of the internal structure. Although the builder *is* briefly instructed in the manual to remove the baffle, I recommend that this detail be highlighted in the manual or a small insert card included inside the enclosure shipping boxes. For an excited new owner of the Dayton Euro Series Monitor, it's just too easy to miss this important instruction—making life somewhat difficult for the project builder.

All in all, the assembly was very organized and simple to follow. With the

PHOTO 3: A completed highpass board (top) with its lowpass counterpart (bottom). These boards leave plenty of room for you to work comfortably.



exception of the enclosure's somewhat tight quarters, this was a painless assembly.

LISTENING

Jack DeJohnette, *Free Above the Sea*, from *Oneness*

This jazz album contains a great deal of very delicate, yet blindingly quick ride cymbal work. The resolution of the textile dome tweeters seems very good on this track, as every individual tap of the drumsticks is very focused and clear. Speakers with less definition seem to mush these together to sound more like a "rhythmic hiss."

The stereo image of the piano in this track is fairly stable, and I found it easy to locate the position of every string in the left-to-right panorama.

PHOTO 4: The enclosures arrive preassembled, including all driver recesses, screw holes, and wood finish. You add the foam, crossover filters, reflex port, and all wire.

Wayne Shorter, *Atlantis*, from *Footprints Live*

This track begins with John Patitucci playing a bowed acoustic bass in its upper register. The Dayton Euro Series revealed the subtle scratching of the bow across each string. This is a detail that I do not hear as distinctly with many other speakers I have sampled.

As Patitucci moves into the lower register, I find the bass to be fairly full; however, the best results occurred when I mounted the speakers 12″ from the back wall. This system would certainly be at its best if aided with a subwoofer. Wayne Shorter's saxophone sounds very smooth with no signs of being honky or midrangy.

Ravi Shankar, *Get in Dhamar*, from *Full Circle—Carnegie Hall 2000*

Ravi and his young daughter Anoushka play together in this live recording from 2000. Two sitars may risk sounding "pingy" together, but are quite smooth both on quiet sections and hard string plucks. The drone strings of the sitars have beautiful midrange fundamental tones accompanied by upper frequency harmonics that come through very well. As with Patitucci's bowed bass, the sitars show how nicely focused the Dayton Euro Series tweeter performs.

John Fogerty, *Bad Bad Boy*, from *Blue Moon Swamp*

Fogerty's screaming lead guitar à la Creedence Clearwater sounds smooth, although it is somewhat reserved. This made me want to grab the volume knob to bring out the madness of this foot-stomping rocker a bit more, as it didn't have quite the right "feel." The bass guitar sounded fairly powerful, but again, the low end could benefit from a subtle subwoofer.

Counting Crows, *Mr. Jones*, from *August and Everything After*

A great rock & roll track from the early 1990s, "Mr. Jones" is represented with great authority. Bass guitar and bass drum together sound quick, thick, and punchy. Any weaknesses I may have noted in the previous track were *not* present here. After listening to this track for evaluation, I played it again, just to howl along like an "American Idol" reject; humiliating if anyone witnessed me. This song felt good!

Chris Norman, *Sleepy Waltz* and *Thousand Pipers*, from *Man With the Wooden Flute*

Sleepy Waltz is a quiet lullaby-like piece in which Chris Norman's wooden flute sounds very nice and liquidy in lower registers, while intimate and breathy in the high range. Robin Bullock accompanies on mellow-sounding acoustic guitar. This piece is beautiful and sounds like one performance (as it should); not like two separate players. In *Thousand Pipers*, Robin Bullock is on the citern, a Celtic guitar-like instrument, sounding quite reserved, yet the attack of each picked note is finely detailed and uncompressed.

Orpheus Chamber Orchestra, *Allegro—Suite in D Major, HWV 349*, from *Handel: Water Music*

The oboes have a very sweet (not "honky") tone and sound slightly brighter than their French horn counterparts, as they should be. The strings are very well detailed and present a sharp, defined stereo image. Everything seems to be in the right place.

Overall, the Dayton Euro Series Monitor 302-920 (also the 302-922 and 302-924 versions) is a wonderful project for nearly anyone who is interested in speaker building. Because the crossover boards require assembly, this appears to be a great learning project which may be a bit more involved than a new hobbyist may care to tackle, but it's perfect for your next conquest after getting your feet wet. More important, the Dayton Euro Series Monitor is a very well-made system, which I believe you could call "high end" in terms of its high/mid frequency characteristics, not to mention its looks! The system would sound complete if mated with a subwoofer for the lowest frequencies, but this isn't a deal breaker by any means.

MANUFACTURER'S RESPONSE

While the speaker cabinets are delivered with the baffles installed, this is for shipping purposes only. The baffles should be removed by taking out the screws located in the grille socket holes. If this is done first, it will eliminate any difficulty installing the crossovers or damping materials.

Darren Kuzma
Product Manager– Speaker Building
Parts Express *aX*

Testing the Parts Express Dayton Euro Loudspeaker Kit

By Joe D'Appolito

I ran a series of impedance, frequency response, polar response, and distortion tests on the Parts Express Dayton Euro loudspeaker kit built by the Ketlers.

IMPEDANCE

It is my habit in testing these kits to arbitrarily pick one unit of a pair for all tests and then check the second unit simply to match against the first. **Figure 1** is a plot of system impedance magnitude (A) and phase (B). At low frequencies the plot displays the double peaked curve typical of vented systems.

The impedance minimum between the peaks at 40Hz indicates the vented-box tuning frequency (F_b). There is an absolute impedance minimum of 5.4Ω at 375Hz. Over most of the frequency range impedance phase angles range between ±36°. A rather large negative angle of –60° occurs at 85Hz, but the impedance magnitude at this point is 17Ω. I would rate this an 8Ω speaker.

FREQUENCY RESPONSE

Figure 2 shows the full-range frequency response of one of the pair of Dayton Euro speakers. This response is obtained as a combination of the far-field quasi-anechoic response in the free standing position and a set of appropriately combined near-field woofer and port measurements[1]. I placed the microphone along the tweeter axis at a distance of 1.06m to produce the far-field free-standing response. Near-field and far-field responses were then spliced together at 200Hz to produce the full range response[1].

The response data is 1/6 octave smoothed. In the two octaves between 500Hz and 2kHz, sensitivity averages 85.3dB SPL/2.83V/1m. Response varies by ±3dB over the range of 100Hz to 20kHz. The response would be flatter except for the "S" shaped peak-dip of 6dB between 1.5kHz and 2.5kHz. (As you will later see, the second speaker has a much flatter response.) The low-frequency –3dB point is at 58Hz.

The Dayton Euro is not configured for bi-wiring so I could not determine responses for the individual drivers, but the assembly instructions state the crossover frequency is 2kHz.[2]

WOOFER/TWEETER TIMING

The Dayton Euro step response is plotted in **Fig. 3**, which shows 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 later. The woofer peak is delayed relative to the tweeter peak by approximately 300µs. This is a rough estimate of the tweeter/woofer delay.

Although both drivers are connected with positive polarity, the system is not time-coherent. Notice the "wiggles" in the step response both at the negative-going peak and the woofer peak. This suggests that there is some oscillatory behavior in the speaker response.

CUMULATIVE SPECTRAL DECAY

This is confirmed by the cumulative spectral decay (CSD) response in **Fig. 4**. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. Frequency increases from left to right and time moves forward from the rear.

FIGURE 1: Dayton Euro impedance magnitude (A) and phase (B).

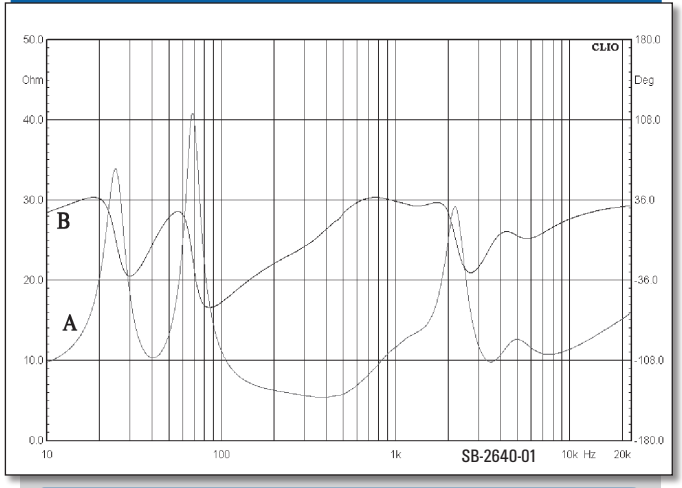
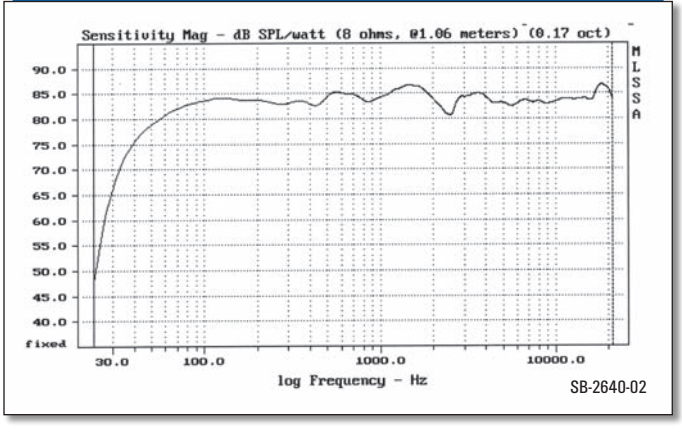


FIGURE 2: Dayton Euro full-range response.



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. Real speakers, however, involve inertia and stored energy that take a finite amount of time to die away. 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 between 1kHz and 9kHz decaying slowly and extending out to 5ms. Considering the frequency range of this decay, it appears to be coming from both the woofer and tweeter. Compared to other loudspeakers I have tested for *audioXpress*, this is rather poor decay performance.

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° off-axis the curves are indeed fairly smooth replicas of the on-axis response. The –3dB coverage at 15kHz is ±25°. At 10kHz the -3dB coverage extends to ±35°. This is typical of 25mm dome tweeters. The 60° curve shows the response transition from the woofer to the tweeter starting at about 2kHz.

Within the woofer's active range the worst-case –3dB coverage occurs at 1400Hz and ±35°. This is due to the woofer directivity at that frequency. At ±60° off-axis 1426Hz response is down 7.2dB. By 3kHz the system output has fully transitioned from the woofer to the tweeter. At 3kHz and ±60° response is down only 3.1dB. 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 with the on-axis response (dotted line). The average horizontal response is within 1.1dB 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 20° below (-20°) the tweeter axis to 20° above it. Moving below the tweeter (negative angles), the response changes little from the on-axis curve out to –15°. Recall that the woofer is behind the tweeter acoustically, so as you drop below the tweeter axis you move closer to the woofer, bringing the drivers closer into phase. The opposite happens as you move above the tweeter axis. As shown in **Fig. 7** a deep notch develops just above the crossover frequency, reaching -13.5dB at +20°. Clearly you should be listening at or below the tweeter axis for best response.

INTERMODULATION DISTORTION

In past reports I have listed results for both harmonic and intermodulation distortion (IMD) tests. However, I believe that IMD tests are more revealing of loudspeaker performance. So in this and future reports, I will list only IMD results, which do not directly predict how

FIGURE 3: Dayton Euro step response.

FIGURE 4: Dayton Euro cumulative spectral decay.

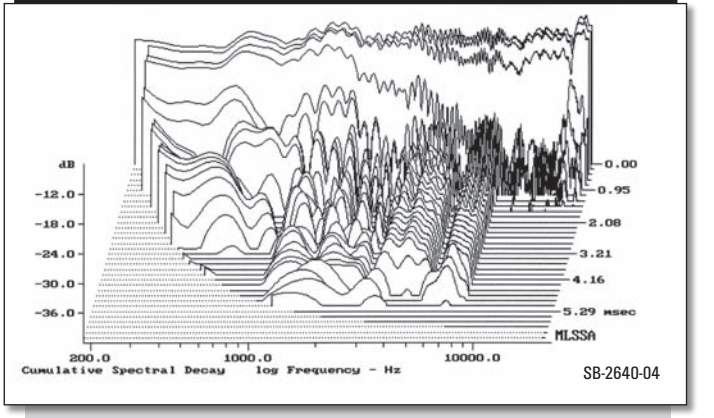
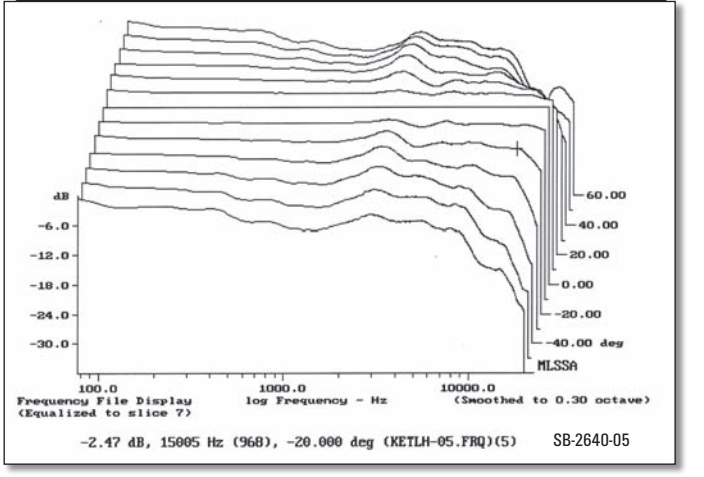


FIGURE 5: Dayton Euro horizontal polar response waterfall plot.



a speaker will sound, but help us judge driver linearity and by implication driver quality. In the next section I'll introduce a new criterion for judging driver quality.

In IMD tests two frequencies are input to the speaker. IMD produces output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion. Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a 2nd-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A 3rd-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 700, 950, and 1500Hz. However, the overall level is only 0.06%. This is an excellent result.

Next, I measured tweeter intermods with a 9kHz and 10kHz input pair adjusted to produce a 90dB SPL at 1m. Results are plotted in **Fig. 9**. The major IM products occurred at 8, 11, and 12kHz. Total IM distortion was 0.068%, 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 10kHz, again at a 90dB level. Ideally, the crossover should prevent

FIGURE 6: Dayton Euro horizontal response averaged over ±30°.

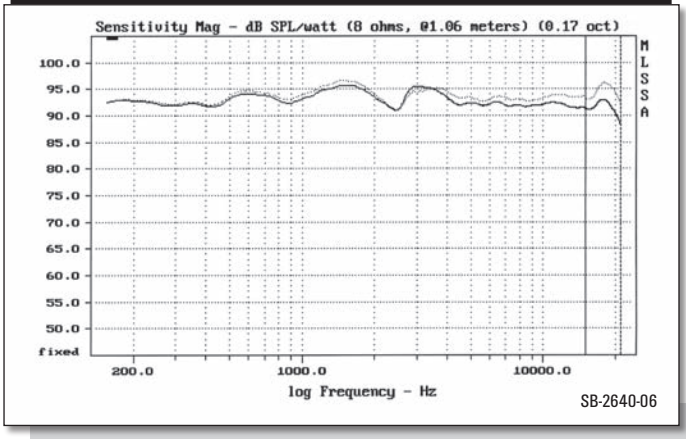


FIGURE 7: Dayton Euro vertical polar response waterfall plot.

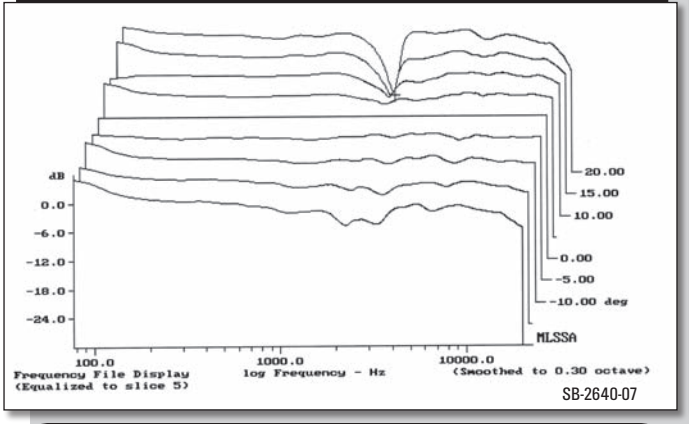


FIGURE 8: Dayton Euro woofer IMD.

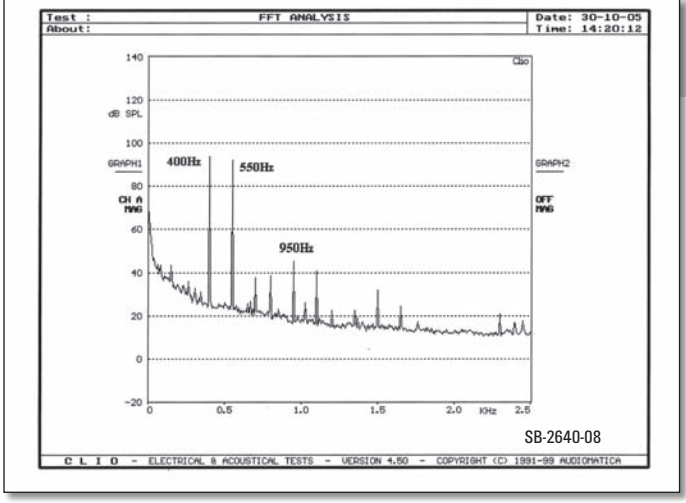


FIGURE 9: Dayton Euro tweeter IMD.

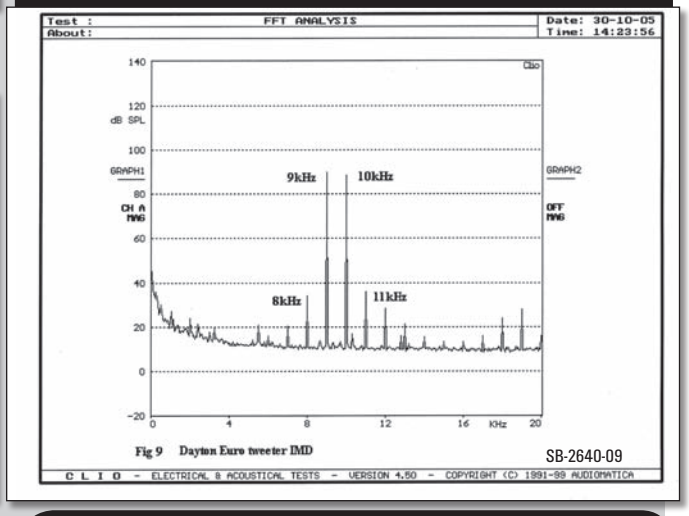
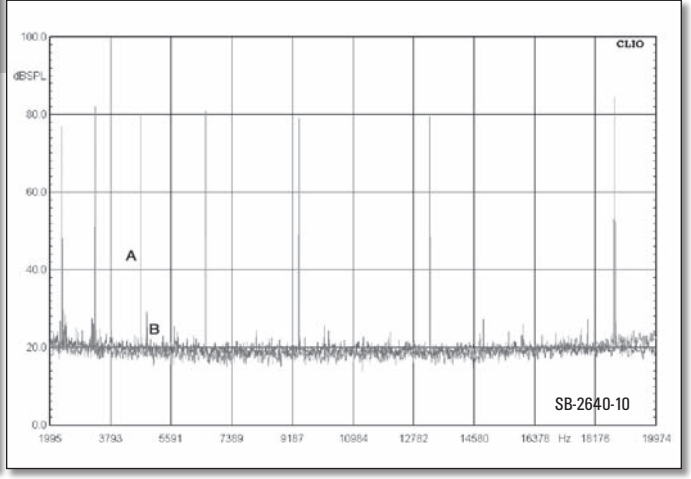


FIGURE 10: Dayton Euro spectral contamination—Euro spectrum (A), background noise (B).



high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. Tweeter sidebands appeared at 9450 and 10,550Hz at an overall level of 0.056%. This is a very good result and indicates good inter-driver isolation by the crossover.

SPECTRAL CONTAMINATION

This is a new and interesting test of driver performance. When excited by a broad range of frequencies in program material, drivers may produce responses unrelated to the program material. The causes of these spurious responses may be turbulent air velocity in the gap, cavity resonances in the back volume, and mechanical hysteresis, among others. This response is called spectral contamination. You can test for it by exciting the loudspeaker with a spectrum of widely spaced sine waves. Then, analyzing the output spectrum, you look *between* the sine waves to see whether there is any significant acoustic output.

I drove the Dayton Euro with a multi-tone signal consisting of 17 sine waves between 50Hz and 18kHz spaced one-half octave apart. The resulting output spectrum is marked A in **Fig. 10**. The scale has been expanded to show the spectrum between 2kHz and 20kHz. The output sine waves are seen as a series of narrow vertical lines averaging 80dB SPL each. Between the sine waves you see a low-level noise-like spectrum averaging 20dB SPL. Is this noise-like spectrum coming from the drivers, or is it simply the background noise level in my lab?

To answer this question I show the measured background level on **Fig. 10** marked B. Now it is clear that most of the acoustic energy between the sine waves is background noise, and thus spectral contamination is at least 60dB below the program material. This is a good result. Defective or poorly designed drivers often show spectral contamination levels 40dB or less below the sine waves.

FIGURE 11: Matching between two samples of the Dayton Euro.

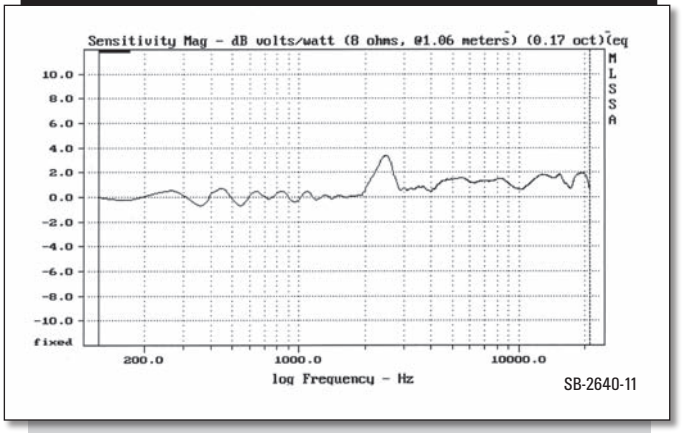


FIGURE 12: Comparing responses of Samples 1 & 2.

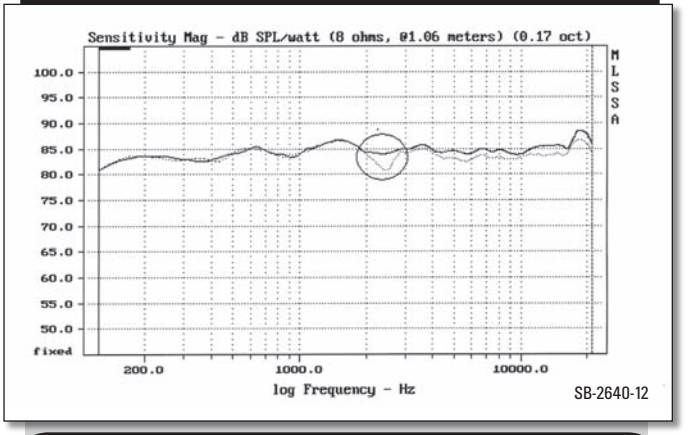
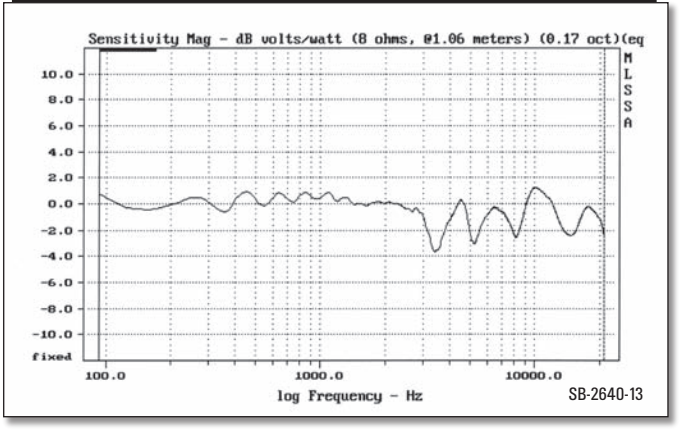


FIGURE 13: Effect of grille on Dayton Euro response.



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 Dayton Euro samples match quite well below 2kHz. Above 4kHz the second system slowly departs from the first reaching a +2dB mismatch at 20kHz. Between 2 and 3kHz you see the mismatch peaking at 3.6dB. Looking at the circled area in **Fig. 12**, you see that the second system is actually flatter than the first. Ignoring the area between 2 and 3kHz, matching is fairly good, especially for a kit.

GRILLE EFFECTS

I conducted all of these tests with the grille off. **Figure 13** shows the Dayton Euro response with the grille on, but referenced to the response with the 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 3kHz. Overall response variations of +1.6dB to –3.8dB are seen. As usual, the grille is there only for cosmetic effect and to protect the drivers from prying fingers. *aX*

REFERENCES

1. J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Inc., Peterborough, NH, 1998.

2. Loudspeaker Kit Assembly Instructions: Dayton Euro Series Monitor Kit, Parts Express document.

A note on testing: The Dayton Euro 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 was 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 the Old Colony Division of Audio Amateur Inc.

Listening Review: NAD L53 Receiver

By Gary Galo, Regular Contributor

NAD L53 DVD Receiver. NAD Electronics International, a division of Lenbrook Industries, Ltd., 6 Merchant St., Sharon, MA 02067, 781-784-8586. www.nadelectronics.com. List Price: $599. Source for review sample: Factory-refurbished unit purchased by *audioXpress*.

This review was originally intended as a companion to Charles Hansen's extremely thorough test and measurement report on the NAD L53 DVD Receiver published in the Dec. 2005 issue of *aX*. My portion of the review was interrupted by our move to a bigger house just as the fall 2005 semester at SUNY Potsdam got under way. Between moving some 12,000 recordings (78s, LPs, and CDs), taking care of the many things a new house entails including setting up a new listening room, and keeping up with my day job, many projects fell behind.

OPERATION

The NAD L53 comes with a remote control that I had to learn, in part, by trial and error. The manual has six pages of instructions on operating the remote, but many things could use more detailed explanation. There is a hierarchy to the remote's logic that is not at all intuitive.

You first select a device via the Device Selector keypad, and then use the various other buttons associated with that device. Some buttons have multiple functions, depending on which device (AMP, DVD, TUNER, and so on) you selected. These functions are confusing at first, and should have been better explained in the manual.

The L53 uses a Vacuum Fluorescent (VFL) Display as its main display. Unfortunately, the manual does not reproduce the display or explain the various indications. There are three indicators that are illuminated all the time: "L," "R," and "LFE." "L" and "R" are self-explanatory, and "LFE," which stands for "Low-Frequency Effects," refers to the L53's summed, mono subwoofer output, a line output intended to feed a powered subwoofer ("LFE" is standard Dolby terminology for the "point one" subwoofer output on 5.1 surround systems).

Because the left, right, and LFE outputs are always active, and can't be changed by the user, it may seem strange that the display illuminates these indications. The left and right main outputs always operate full-range. The manual should at least mention all of this, letting you know that you haven't missed something in the setup (I was left wondering whether the "LFE" somehow needed to be turned off in my installation, but NAD makes no mention of "LFE" in the manual). NAD uses the same VFL Display in several other products, including some with surround capabilities. In these cases, there are additional indicators that display which outputs are operational.

The L53 manual contains one misleading error. On page 6, the manual states: "High-performance components used throughout the receiver's analog audio circuits maximize quality from all sources, including multichannel analog sources such as DVD-Audio and SACD." The problem here is that the L53 does not play DVD-Audio discs or SACDs.

If you put a DVD-Audio disc in the player, after attempting to load the disc the L53 will display a message on your TV screen that says "DVD Audio Disc," and will then eject the disc (there are some DVD-Audio discs that are DVD-Video compatible, which will play on the L53). If you try to load a single-layer SACD, the player displays a "No Disc" message (the L53 will play the CD layers of dual-layer discs).

NAD's product literature does not make any claims for DVD-Audio or SACD compatibility, but the manual is available online, so potential purchasers may be misled into believing that the L53 is compatible with these formats. Also, the NAD L53 is not a multi-channel device—it is a two-channel stereo receiver. NAD should correct the online and printed manuals ASAP.

THE SOUND

The L53 receiver is likely to be used with fairly modest loudspeakers, but I decided to evaluate the unit with my main loudspeaker system, a pair of ACI (Audio Concepts, Inc.) Sapphire III satellites along with their Sub-1 subwoofers. Each Sapphire III was bi-wired using a pair of D.H. Labs Q10 loudspeaker cables, and the Sub-1s were connected with double runs of D.H. Labs T14. The Sapphire

III/Sub-1 combination is nominally a 4Ω system, but it is a fairly easy load to drive, and the L53 had no difficulties with it. Because my Sub-1s are passive, and I have one for each channel, I did not use the subwoofer output.

I normally shy away from "all-in-one" pieces of equipment, because there are invariably performance trade-offs that must be made when several devices must be incorporated into a single chassis. Adding a DVD player to the three components already found in a receiver (preamp, power amp, and tuner) can only complicate matters further. But, there are plenty of instances where integration is desirable, for reasons of space and/or budget. As Chuck Hansen noted in his review, NAD seems to have made the right compromises in the L53, rejecting cheap, switching power supplies in favor of conventional transformers and regulation, and using high-performance Burr-Brown/TI OPA2134 op amps in the main signal path.

NAD's design choices have paid off in the sonic performance. The L53 is sonically much better than I would have expected, with at least one foot solidly in the entry-level, high-end camp. The L53 provides musically satisfying, nonfatiguing listening over the long haul. It produces a credible stereo image from left to right, and gives at least a hint of depth perspective.

Bass performance is surprisingly good for so modest a product. Bass drums and low strings have decent weight and impact. In the *Rachmaninoff Symphonic Dances* DVD and *Respighi's The Birds* (Mercury CD), the low strings were warm and rich, and even the high strings have a pleasingly musical sound. The L53 offered quite smooth reproduction of choral music when played at reasonable levels, particularly on the 96kHz/24-bit DVD transfer of *Abravanel's Berlioz Requiem*.

Any product in this price range must have weaknesses, but it is almost unfair to compare a product as affordable as the L53 against my reference system. Nonetheless, readers are entitled to know where it falls short. The midrange has a rather dry quality; instrumental timbres are tonally quite accurate, but there's not much liquidity in the sonic presentation. The soundstage is somewhat narrow from left to right, and depth perspective is rather shallow and not particularly precise.

Though the bass weight is pretty impressive, definition is somewhat lacking. Low string articulation can be a bit mushy. At large orchestral climaxes, the L53 will become a bit congested, and high string tone can become a bit coarse or wiry. If really pushed, the treble can become a bit sizzly (this is, after all, a modestly-powered device). The L53 does not resolve every last ounce of inner detail, but at reasonable levels its errors are mostly things left out, rather than the addition of unmusical things. Bear in mind that I have evaluated the L53 on an extremely revealing, high-end loudspeaker system.

I also found that the L53 is good enough sonically to make differences between inexpensive and high-end loudspeakers readily apparent. I found that the Madisound *Sledgling* (more on this system follows) performed very well with the L53, but the small loud-

speaker's weaknesses were also obvious, compared to my reference ACI system. These weaknesses include a tendency toward brightness in the treble, a less precise stereo image, and a loss of inner detail. Of course, it goes without saying that such a small loudspeaker system will lack the bass of my Sub-1s. If space, rather than cost, is the main consideration, the L53 becomes better with more refined speakers.

Video performance of the L53 leaves nothing to be desired, and delivers the superb picture quality I have come to expect from DVDs with current-generation video circuitry. The FM tuner did a fine job on our local public radio station, and offered slightly lower stereo noise than my reference tuner (an Adcom GTP-400) on the CBC station from Ottawa, some 85 miles away. The L53 does not have an FM signal strength indicator, but NAD clearly could not offer every desirable feature at this price. The FM tuner display supports the Radio Data System (RDS), which allows the L53 to display the station name (PS mode) and radio text (RT mode), if the stations encode that information in their carrier (our local public radio station does).

SRS TECHNOLOGY

The L53 is basically a two-channel device, but it incorporates the Sound Retrieval System developed by SRS Labs which simulates surround sound from two-channel sources. The L53 manual notes that "SRS, the leading 3D Sound technology in the world, is the result of many years of research based on the psychoacoustics of sound and the dynamics of the human hearing system. SRS retrieves the spatial information that is lost during the recording and playback process and restores the original three-dimensional sound-field. As a result, the reproduced sound is much closer to what the artist originally intended and to that of a live performance."

On classical music recordings, I found the SRS system to be very unnatural. Orchestral recordings are reproduced with a strangely hollow quality. SRS destroys the original two-channel image, making it impossible to localize instruments within the original soundstage. The orchestra seems awash in a sea of confusion.

While listening to SRS, I can feel the pressure differences on my eardrums produced by the extreme phase changes rendered by the system, similar to listening to opposite polarity (incorrectly, "out-of-phase") loudspeakers. I found this an annoying effect, one that did not make me wish to continue listening. For classical music collectors who hear live music on a regular basis, SRS is a gimmick. I hear live classical music every day, and SRS doesn't sound like any live performance I have ever heard. Collectors of more popular fare may find the effect more pleasing.

I did try SRS on a couple of DVD movies with stereo soundtracks, and found the effect a bit more satisfying. My reaction to the SRS feature has nothing to do with the rest of the L53's performance. If you don't like SRS, you don't have to use it.

A CUT ABOVE THE ALTERNATIVES

The L53 is likely to be used in situations where space is at a premium and/or cost is a deciding factor, including a college dormitory, office, apartment, or as a second home system. For many applications, the NAD L53 provides an attractive alternative to products like the Bose Wave Music System with integrated CD player, which is currently selling for $499. This one-piece unit incorporates a pair of less-than-overwhelming 3″ speakers and does not play DVDs.

There are a number of loudspeakers priced around $200/pair or less that would make fine companions to the NAD L53. Audio Advisor (**www.audioadvisor.com**) often has incredible sales on high-quality, small loudspeaker systems. Currently, they are selling the Athena Audition AS-B1 for $99/pair (list is $180), and last spring they offered the Wharfedale Diamond 8.1 with maple finish for the same price (half of list). Check their website for current specials like these. The PSB Alpha B loudspeakers are also getting favorable press these days, and normally sell for less than their list price of $249/pair (**www.**

psbspeakers.com). PSB is a division of Lenbrook, NAD's parent company.

DIY audiophiles should consider the Dayton BR-1 kit from Parts Express (**www.partsexpress.com**) at $140/pair, or Madisound Speaker Components' fine *Sledgling* kit priced at $216/pair (**www.madisound.com**). Both come with assembled enclosures, with the *Sledgling* featuring particularly robust construction. The point here is that for $200 to $300 more than the cost of a Bose Wave Music System, you can get performance several orders of magnitude better by mating an L53 with any of these loudspeakers, and you get a DVD player to boot.

In short, the L53 is a very good performer at a very fair price, continuing the NAD tradition of real value for the money. Anyone in need of a cost-effective, space-efficient audio system should consider this fine product.

RECORDINGS USED FOR EVALUATION

Rachmaninoff: Symphonic Dances, Op. 45. Dallas Symphony Orchestra, Donald Johanos, conductor. Classic Records DAD 1004. (96kHz/24-bit PCM transfer of 30-ips analog tape engineered by David Hancock).

Berlioz: Grande Messe des Morts (Requiem), Op. 5. Charles Bressler, tenor; Utah Symphony Orchestra, University of Utah Civic Chorale and University of Utah A Capella Choir conducted by Maurice Abravanel. Classic Records DVD HDAD 2012. (This set contains two double-sided DVDs, one with 96kHz/24-bit DVD-Audio Surround and Dolby Surround programs, and another with 192kHz/24-bit stereo DVD-Audio and 96kHz/24-bit stereo PCM programs; the 96kHz/24-bit stereo PCM program was used for this evaluation).

Rimsky-Korsakov: Scheherazade, Op. 35. Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo "Gold" CD 68568-2.

Mussorgsky, arr. Ravel: Pictures at an Exhibition. Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo "Gold" CD 68571-2.

Dukas: The Sorcerer's Apprentice. Boston Symphony Orchestra conducted by Charles Munch. RCA Victor Living Stereo CD 68978-2.

Respighi: The Birds. London Symphony Orchestra conducted by Antal Dorati. Mercury Living Presence CD 432 007-2.

Schoenberg: Five Pieces for Orchestra, Op. 16 (1949 Revision). London Symphony Orchestra conducted by Antal Dorati. Mercury Living Presence CD 432 006-2.

Stravinsky: The Firebird (Suite, 1919 Version). Atlanta Symphony Orchestra conducted by Robert Shaw. Telarc SACD-60039. (Direct Stream Digital transfer of the original Soundstream digital recording; the CD layer was used for this evaluation). *aX*

MANUFACTURER'S RESPONSE:

Gary points out that the manual contains an error on page 6. This will be reworded in future issues of the manual and on the website.

—Mark Stone
NAD Electronics International

VENDORS

Pro audio kits: mic pre, mixers, buffers, crossovers, power amps.
Also custom designs.
www.borbelyaudio.com
Selected Borbely Audio kits in Japan:
http://homepage3.nifty.com/sk-audio

Amplifier building is easy with ezChassis®.
Pre-punched holes and slots.
www.designbuildlisten.com

Revised Audio Restoration, new Audio Help, new products models 465 and 912.
www.tdl-tech.com (505-382-3173)

WANTED

stereo gear old/new/quality, speakers, amps, preamps, turntables,
reel to reels, vacuum tubes, tube testers, guitars, parts, etc. **850-314-0321**
Email: **sonnysound@aol.com**

Ad Index

Yard Sale

FOR SALE

Gorgeous prototype cabinets in piano-laquered cherry veneer for cancelled high-end speaker manufacturer's project. 3 types: 2-way bookshelf, 3-way 8″ and 10″ tower, 1 pair each. Absolutely rock-solid cabinets. Also many sample and prototype drivers from Audax, Seas, Scan-Speak, and many others. E-mail for details and pictures of cabinets.

dpierce@cartchunk.org

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

LINE ARRAYS

Line arrays have been a very popular topic among DIY enthusiasts for the last half-decade or so, partially due to the availability of reasonably priced electrodynamic planar speakers. Mr. Russell's article ("A Unique Stereo Column System," Oct. '05 *aX*) was interesting, covering many of the advantages of line arrays, although he did not touch on the problem of comb filtering, which will limit the fidelity of an array in the high-frequency range. Mr. Russell's array will suffer from this somewhere in the region of 3.2kHz and above (based on driver spacing of 4.25″). Admittedly, some DIYers report that the problem is not as serious sonically as theory would predict, but still—with that spacing—it will have phase problems in the top two octaves at least.

Today, Dr. Griffin's line array paper (2003) is the leading set of guidelines used by those building line arrays http://www.audiodiycentral.com/resource/pdf/nflawp.pdf. Also of interest is Paul Taylor's 1964 paper http://homepage.mac.com/tlinespeakers/TLS/downloads/taylor-line-array.pdf. Although primarily focused on far-field use of line arrays, this one goes deeper into the math and has some very illustrative graphs of the comb-filtering issue. There is also good coverage of power tapering.

Mr. Russell's patent #4,267,405 is an interesting read. The tweeter array described with 3.1″ spacing will comb above 4.4k, meaning it will provide not much more than two octaves before the phase problems associated with combing come into play.

David Dlugos
Victoria BC
Canada

Roger Russell responds:

The information that Mr. Dlugos is looking for is contained in my second article to be published in *audioXpress* sometime this year. This is about an updated system having improved drivers plus information concerning comb filtering, driver spacing, and driver directionality.

Many years ago, I read an article titled "Performance of Series-Parallel Speaker Arrays" by James Novak that was published in *Audio Magazine*, September 1961. It explains the expected losses from driver spacing, and so on. The effect of reduced output at the higher frequencies has been taken into account in both article designs.

I don't know whether Mr. Dlugos has ever heard a properly designed column, but despite all the misgivings about comb filtering, phase and theoretical predictions, stereo imaging and coherence in music are more accurate than ever. Response is very smooth and extended and is confirmed by both listening and measurement.

Power tapering or response tapering is not necessary for long column designs, and it defeats some of the advantages of the long column, such as increased floor reflections for the mids and highs and reduced power handling at the higher frequencies. Worse than that, it could introduce those dreaded parts used in crossover networks!

NOS COMPONENTS

I've come across a source for new-old-stock of hard-to-find semiconductors. You're probably familiar with the PMD17K100, PMD16K100 power Darlington transistors from Lambda that were used in the Pass A-40 power amp (*TAA* 4/78). I used these as an example, and I can get 300 of each just from one particular lot dated 1990. Price is $3.50 each in quantities of 100, Lambda metal can parts from a trustworthy source.

I believe that I can get the rest of the transistors (the small signal parts should be much less) in case you want to bring back the Pass A-40 kit. This might be a way to have much more affordable kits. An air-cooled monoblock version in a nice retro chassis would be sleek: http://www.hammondmfg.com/jpeg/1441Grp_B.jpg.

Pete B.

Professional page:
http://members.aol.com/basconsultants/bas_cons.htm

Informal audio page:
http://members.aol.com/basconsultants/bas_cons.htm

Some of my classic speaker-related posts:
http://www.classicspeakerpages.net/dc/dcboard.php?az=show_topic&forum=3&topic_id=5243&mesg_id=5243&page=
http://www.classicspeakerpages.net/dc/dcboard.php?az=show_topic&forum=3&topic_id=5576&mesg_id=5576&page=9
My amplifier-related posts:
http://www.diyaudio.com/forums/showthread.php?s=8d5fe7f215d9f5330e08752b5d939f87&threadid=48873

CD MODS

In the February 2003 *aX* ("Upgrade Rotel's 970BX CD Player," p. 26), Charles Hansen outlined a simple and effective mod for the Rotel 970BX CD player. Using his template, I modified my player, albeit with slightly different parts. I have the itch to fiddle a bit more with this player and I would like to bounce a few questions off Mr. Hansen.

As noted in the article, the player's output is higher than the standard 2V. What resistor values might you suggest trying at the outputs to attenuate this monster's gain? I'd like my ARC SP6B preamp's gain pot (Alps Black Beauty 100K) to be more usable! As it is, I can use only the first half-dozen clicks, or about to "9 o'clock." With other sources, the pot's usable range goes to "3 o'clock."

I have not bypassed the 100µF output 'lytics yet, but I did replace the op amps. I used new quality sockets and fitted them with Burr-Brown's OPA627s. Sounds terrific. Thanks for the hints!

I know that circuits like this usually comprise low-noise metal film resistors, but I've had such fantastic results using both Riken and Kiwame carbon resistors in my amplifier's signal path that I'm tempted to try these out in the CD player. I know it's atypical—and perhaps good reasons exist for this—but I think the player's sound could stand a little "warming up" anyhow. I must say, too, that the new op amps warmed the sound considerably. The player immediately played with a chestier, harmonically richer sound after this surgery.

What do you think of this modern carbon resistor idea? The Rikens, in particular, sound lovely, and although I don't have the equipment or skill to measure such things, they claim to be very quiet with tight tolerances. I've also been itching for a reason to try a couple of the nearly disappeared Shinkoh tantulums. Might be a good spot.

Also, have you ever messed about with biasing the op amps into class A operation? If so, how'd you implement it and what did you think of the results?

Thanks so much for your articles. Like many of those in *audioXpress*, I enjoy them quite a bit.

Mike Genett
Los Angeles

Charles Hansen responds:

Thanks for your kind letter concerning my Rotel 970BX modification article. I'm glad you found the article helpful. Let me try to answer your questions.

1. According to the Philips data sheet, the TDA1305 has a full-scale output (Vfs) of 1.5V RMS into 3k. In order to obtain a 2V RMS Red Book full-scale output from the 970BX, you will need a gain (Av) of 1.333 at your OPA627. Change the feedback resistor (pin 6 to pin 2) from 4k70 to 4k02 and you should have it.
2. Many people think highly of the OPA627.
3. The use of different resistor types is a subjective choice, so I must leave that up to you and other readers. There are two types of carbon resistors—carbon composition and carbon film. Riken seems to call the latter carbon composite film.

Carbon composition resistors are quite noisy, but guitar amplifier manufacturers seem to like them for their warmth. They do not have tight tolerances because of the generally accepted temperature coefficient (tempco) of –1200 ppm/°C. The exact tempco is hard to pin down because it is not only nonlinear but also varies with resistance value. In addition, in the presence of high temperature or humidity and as they age, they change resistance, usually downward as much as –15%.

Carbon film resistors are only slightly more noisy than metal film types. The tightest tolerance is usually 5%, but 1% tolerance is possible over a narrow temperature range for selected resistance and wattage ratings. Long-term resistance drift may still occur due to time and temperature.

Tantalum film resistors seem to have come and gone, as you stated. Nichrome is usually used for metal film resistors.

4. Biasing op amps into Class-A operation is a well-established technique. You can find information in *TAA* 2/90 pages 48-49, *TAA* 1/91 page 48, *TAA* 1/94 page 37, *TAA* 1/96 page 25, and *AE* 3/97 page 39.

JUNK BOX AMPS

Bruce Brown's article on junk box amps was excellent (Dec. '05 *aX*; Larry DeRuyter co-author). This is valuable to all of us who are working on tube amps and keeping this great culture alive. I have a few questions.

- Can I use a Heath A9 as one of the monoblocks for this project?
- Since I am unable to find another A9, is there another Heath that could serve as donor for the mate?
- Do you have a schematic and bill of materials for the modification to the A9?

Bernie Hyman
NMC Americas
Managed Network Services

Bruce Brown responds:
You can use virtually any output transformer designed for a plate-to-plate push-pull amp. Transformers used with 6L6s, 6BQ5s, EL34s, 5881s, 7591s from Heath, Sherwood, Fisher, as well as many others. Any of the A, B, Cs, or W4s will work, and you can find these online most of the time.

You can contact me directly if I can be of further assistance: Tuninfork@aol.com.

I was quite impressed with the amp project in Dec. '05 *aX*, which was contributed by Bruce W. Brown and Larry DeRuyter ("Junk Box Tube Amp Jubilee," p. 24). There is, however, an error in Table 2 (p. 26) which lists tube rectifiers and their ratings. The filament current for the 5V3 is not 3A as listed—it is 3.8A. It is the 5V3A which has the 3A requirement. Also, the stated 1200mA plate current rating should not be confused with the DC output current given under the voltage drop heading, which is correct.

In Table 1 (also p. 26) the heater current for the 6SN7 and 7N7 is .6 amps—not .5A and .8A (respectively) as listed.

Neal A. Haight
Castro Valley, Calif.

Bruce Brown responds:
You are quite correct on the 5V3. I apparently didn't get the A to stay in the revisions. You are also right on the 6SN7 and 7N7 filament currents. Thanks for your sharp eyes.

PEAK AUDIO LEVELS

I thoroughly enjoyed Dennis Colin's article on musical waveform peak power in the November '05 *aX* (p. 48). I would like to see more articles like this. The subject of peak audio levels and crest factor—and what it means for required amplifier power—is something I have been researching for the past year or so. In my case, I built a special meter that can measure average RMS voltage and peak voltages at the same time, outputting each to its own DVM display. The peak detector includes two stages of precision peak-hold circuits so that even the briefest of peaks is held long enough to be displayed accurately. A couple of friends and I demonstrated this at the recent DIY

speaker gathering in Washington, DC.

My experience was similar to Dennis', with surprisingly large peaks noted on well-recorded CDs with good dynamic range. I took measurements on the following tracks:

1. George Strait—"Ocean Front Property,"—*All My Ex's Live in Texas*—track 1.
2. Rickie Lee Jones—"Flying Cowboys,"—*Ghetto of My Mind*—track 3.
3. "Alison Kraus + Union Station Live,"—*New Favorite*—track 9.
4. Annie Lennox—"Bare,"—*A Thousand Beautiful Things*—track 1.
5. Tracy Chapman—"Crossroads,"—*Crossroads*—track 1.

The results are summarized here. All of the tracks were reproduced at the same volume control setting. The voltages were measured at the terminals of the left channel speaker. The power amplifier used was rated at 250W per channel. The SPL readings were taken at the listening position with both channels driven.

Artist	Vpeak	V RMS	avg SPL	max watts	avg watts	crest factor
Strait	43	4.1	82	116	2.1	21
Jones	46	2.1	77	132	0.6	27
Kraus	30	4.1	80	56	2.1	17
Lennox	54	12.5	90	182	19.5	13
Chapman	41	4.7	83	105	2.8	19

The speakers are 8Ω three-way Morel-based bookshelf speakers with an efficiency of about 83dB 1W/m. The listening position was about 11′ from the speakers. The first number in the watts column is the minimum power rating, per channel, of the power amplifier that would be required to reproduce the peak voltage without clipping. The second number is the actual average power corresponding to the RMS voltage. Both power levels are referred to an 8Ω load.

Obtaining the V RMS and average SPL numbers required some judgment, because they tend to jump around with the program material. In both cases here, it is the number that the meter hovers around during the louder sustained vocal passages over a period of more than one second. I believe this is likely to correlate reasonably well with the perceived loudness of the playback level. The SPL numbers were obtained from a Radio Shack SPL meter set to A weighting and slow response, and they reflect the contribution from both stereo channels.

There are several interesting things to note from the data. First, the crest factor of the Rickie Lee Jones track is substantially higher than even that of the George Strait track measured by Mr. Colin. Also note that the Jones track is recorded at a lower average level than the others, presumably to make room for the high peaks, which occur on snare drum hits. This is a well-recorded album and does not seem to have artificially boosted highs.

Although a 132W per channel amplifier is required to reproduce the Jones peaks, the average SPL at the listening position is nowhere near loud. In fact, a "realistic" level for this would probably be at least 3dB higher, at 80dB SPL, implying the need for a 264W per channel amplifier to reproduce it without clipping at that level with these speakers. To play the Jones cut at the 90dB listening level Dennis used as an example would require a 2640W per channel amplifier. Use of more efficient speakers rated at 86dB 1W/m would only cut this number in half.

While it is true that the Jones track has an unusually large dynamic range, this data still suggests that many listeners may be clipping their amplifiers more often than they think. This may especially be the case for those with tube amplifiers who are not using extraordinarily efficient speakers. The amount of clipping, and the way in which amplifiers handle clipping, may account for more of the perceived differences in amplifier sound than we realize.

It would be nice if all amplifiers had accurate and fast clipping indicators. It might be a real eye-opener. In my view, if your amplifier is clipping, you have left the realm of high fidelity.

Once again, many thanks to Dennis for a thought-provoking article.

Robert R. Cordell
Holmdel, N.J.

Dennis Colin responds:

Thank you for your interest in my article. I especially appreciate your work and the results you published here, which rein-

force the need to allow sufficient amplifier headroom.

I like popular music, including hard-driving rock, but I'm frustrated with the lack of well-recorded (i.e., noncompressed, etc.) material in this genre. I realize this compression (and attenuation of natural deep bass) is perpetrated upon us in the interest of the industry's "loudness wars" (sales are proportional to dB SPL/dollar). Rickie Lee Jones is an exceptional testament to the importance (to perceived loudness) of maintaining the source's dynamic range, even of the short duration (but high amplitude) peaks.

I agree with your belief that the amount and nature of amplifier clipping account for much (if not most?) of perceived amp differences.

My "Mad Katy" tube amp, a 125W per channel, has fast and accurate clipping LEDs, sensitive enough to show a single cycle of overdriven 20kHz. In a fast A/B test using Sennheiser HD 450 phones to compare the amp's input/output signals (amp loaded by a reactive speaker simulator), no difference could be heard until the amp was overdriven about 3–5dB on fast peaks. But just listening normally (through speakers and without referencing to the input), more than 10dB overdrive was often needed (depending upon source) to notice a difference. This amp clips very smoothly with no recovery transient problems. But many amps, tube and SS, produce nasty overdrive waveforms; these would be expected to produce noticeable degradation with even slight overdrive levels.

Thanks very much, Robert, for sharing your collaborative research.

BATTERY REPLACEMENT

I started reading with interest the article by Charles Hansen called "Rescuing Fluke's Top Digital Meter Cheaply," (April '05 *aX*, p. 36). About five years ago I went to the Dayton Hamvention with a Fluke 8050 or 8060 on my shopping list. I found an 8050 for $75 (as I recall). Its batteries were also dead. I hoped to pick up some pointers on improving or repairing my 8050.

As I started reading the article, I realized that all he was doing was replacing the batteries. I recalled that I had ordered some battery packs from Digi-

key that had dropped into the factory holders. I was able to unearth a record of that purchase, and found that I had ordered two P163-F021-ND battery packs from Digi-key. My memory says they totaled about $35 or $40, but I do not have the figure in front of me, and it has been several years.

I was actually disappointed that the whole article was about the battery replacement, something which I had done in about an hour (not counting the time waiting for the batteries to come), and with no more fuss than opening the case, measuring the batteries, ordering them, soldering on the four wires, and reassembling the meter. I don't recall how much work it took to trace down a source for the correct size of battery, but I am sure it was less than Mr. Hansen spent retrofitting his meter.

Anyone else having an 8050 may put the batteries in using either method. I recall that I studied the circuitry and found that it should have worked fine off AC power if you replace the batteries with 2.4V to 2.8V zener diodes. However, the batteries are sometimes very helpful, because measurements are then free of both a ground connection and any AC hum from the transformer and power supply.

I have found the meter very helpful. It is very satisfying (and quick) to send a 1kHz tone through equipment under test, adjust the level for test conditions, and press the "RELATIVE" button and watch the meter show you 0dB. Finding the -3dB points is just a matter of going up or down until it says +3dB or –3dB, and recording the frequency. The portable version (8060, I believe) made it even easier, because it included an audio frequency counter. I miss that feature, but the 8060 stayed with my former employer.

Ted Miller
Elkhart, IN

NOISE METER

I read with interest Charles Hansen's article on the Solid State Noise Meter Amp (Jan. '05 *aX*) and have some comments. You seem to be under the impression that a low input impedance is desirable. While the noise is lower with no source attached, low input impedance degrades the s/n with any source. For example, with a 600Ω source and 600Ω input resistance, the noise level is degraded 3dB over theoretical minimum. The input impedance should be at least 100kΩ.

Also, I question the use of the AD745 with its 3nV/root Hz noise. You can do 14dB better with a transistor front end. In general, low voltage noise sources have low impedance, so current noise is not an issue.

You use additional compensation because the 745 has a minimum stable gain of 5, yet in the circuit it has a gain of over 5.

I assume you use two ICs to get the bandwidth. A three-transistor circuit would provide the bandwidth with more simplicity (single power supply, though you would need some large electolytics).

I wonder why the noise is higher with the AC adapter?

David Hadaway
dbsys@att.biz

USHER REVIEW

I recently read a review on the Usher CP8871 written by Mr. Moriyasu in Feb. '04 *aX* (p. 54). I had a few questions for him about his findings regarding the crossover design/shortcomings of this particular model. Basically, are the design and quality issues correctable without needing to build separate outboard crossovers?

David E. Hawk
david.hawk@sbcglobal.net

James Moriyasu responds:

Yes, because the cabinets are relatively large and roomy, it would be easy to remove the crossover boards and make the necessary modifications without needing to build outboard crossovers.

The main issue would be the interaction of the inductors. Because there were three of them, the first two could have their poles oriented at 90° to each other. However, the third must be separated by some 10–12″ or more to reduce coupling. Of course, the farther the better, and you could mount individual inductors on separate boards and move them as far away from each other as possible. *aX*