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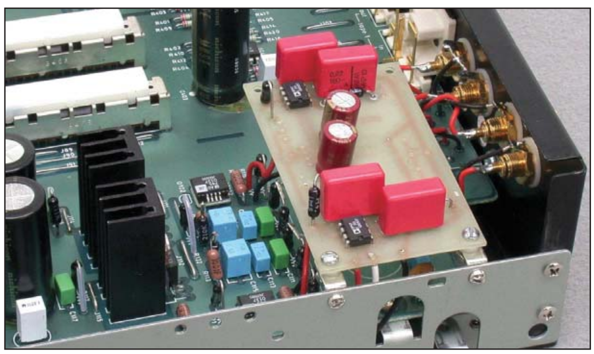

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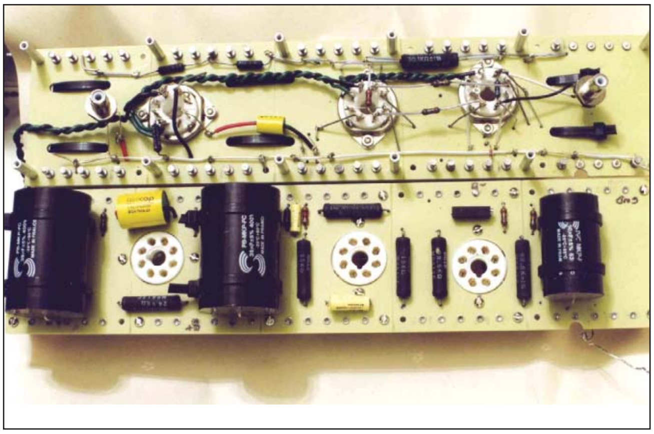

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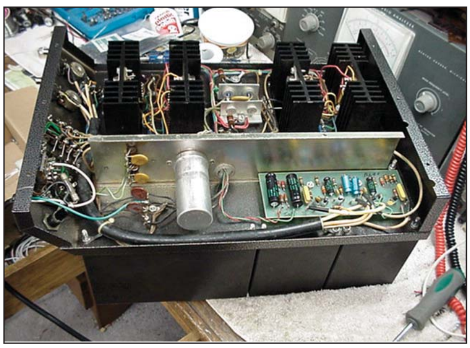

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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A Low-Mu Triode Preamp

This author describes how to achieve low distortion from power triodes. **By Pete Millett**


PHOTO 1: Top view of low-mu triode preamp.

Just about every stereo system uses a preamp—or line amplifier, if you prefer. This article describes a preamp (*Photo 1*) that's a little different from most, using low-mu power triodes to get good performance with only one amplifier stage, and no negative feedback.

WHY DO WE NEED A PREAMP?

The most visible functions of a preamp are to provide source signal selection (e.g., CD player, phono preamp, and so forth) and volume control for your audio system. A preamp doesn't really need very much voltage gain; in fact, some designs, called "passive preamps," don't have an amplifier stage at all. There is much to be said for these designs—after all, with no gain stage and only switches and passive attenuators in the signal path, the noise and distortion performance is hard to beat. So why would we need an amplifier?

The biggest problem (for me) with a passive preamp is that its outputs have a relatively high source impedance, and one that tends to vary substantially depending on the position of the volume control and the output impedance of whatever is driving it. This works OK as long as the load (the amplifier) is well behaved, with a high load impedance, and there isn't too much capacitance in the amp or interconnect cable.

In my system, this isn't usually the case. I typically have several power amps connected, with only one pair in use at a time. Often I also have a headphone amp or two plugged in. Each of the loads is usually pretty benign, but when you parallel several—along with their interconnect cables—you wind up with a load that really needs to be driven with a low source impedance to sound good.

Many amplifiers also aren't all that well behaved. People assume that a class A1 load (such as the input of their power amp) is always a high impedance. At DC, this is usually the case; but the effective input capacitance can be quite high, especially in designs that use high-gain triodes in the input.

TUBE PREAMP DESIGNS

So, you want a preamp, and you're going to build it with tubes. You can choose from several common design topologies used in tube preamps, as well as some unconventional ones.

The oldest preamp designs are essentially small single-ended power amplifiers. They use a gain stage with an output transformer as the plate load of a tube, with the turns ratio set to provide a high load impedance to the plate, and a low output impedance. Many people believe that this is the best type of circuit, partly because it doesn't use any coupling capacitors in the signal path. The downside is that it's expensive to implement—a good line output transformer can cost several hundred dollars, and the cheap ones don't perform very well.

A "classic" design that's typical of 1960s hi-fi equipment uses a high gain triode such as a 12AX7 as a grounded cathode amplifier, coupled to another triode acting as a cathode follower. Without negative feedback, this type of circuit has very high gain (around 80) and distortion, and high output impedance. Lots of overall negative feedback is used to get the gain down to close to 10 or 20, which reduces distortion and output impedance. Though it measures well, I don't like the way this type of circuit sounds. I suspect that I'm hearing the large amount of feedback that increases higher-order harmonic distortion.

A similar design uses a medium-mu triode such as the 6SN7, and no feedback. This type of amplifier, if properly designed and not overloaded, sounds pretty good to me. It has low output impedance and distortion, but cannot deliver lots of current to the load (current delivery is limited by the amount of current flowing through the tube, regardless of output impedance).

Cathode followers are not without their problems (and critics), though. I've had problems with the heater-cathode interface inducing noise, and even insulation breakdown, since the cathode is operated at an elevated voltage.

Another common preamp stage uses two triodes in an SRPP circuit, with the triodes connected in series and the

upper tube acting as both a cathode follower and a plate load for the lower tube. These circuits can achieve very low distortion and output impedance as well, but suffer from the same issues as a cathode follower output stage.

This design takes a different path—a single gain stage using a low-mu triode with very low plate resistance (*Photo 2*). This achieves the modest gain required along with very low output impedance.

LOW-MU POWER TRIODES

The output impedance of a grounded-cathode amplifier is roughly equal to the plate resistance of the tube in parallel with the plate load resistor. You need to make the plate load resistor large relative to the plate resistance to get low distortion. So, to get low output impedance *and* low distortion, you need a tube with a low plate resistance.

A number of large power triodes have low plate resistance (Rp), such as the 6AS7 and 6080, 5998, 7236, WE421A, 6528A, and 6336A. Most were designed for voltage regulator service; the 7236 (I believe) was designed to drive motors for vacuum-tube computer tape drives!

Unfortunately, at first glance, these tubes don't look linear. In a normal resistor-loaded, grounded-cathode amplifier, they have high distortion, mostly even harmonics. This distortion decreases with increasing plate load resistance, but a larger resistor means that you need a higher supply voltage. Luckily, there is a way to present a very high load to the plate of the tube without using a big resistor and high voltage power supply—a constant-current source (CCS) load.

CONSTANT-CURRENT PLATE LOADS

A constant-current source (CCS) plate

PHOTO 2: Quarter view.

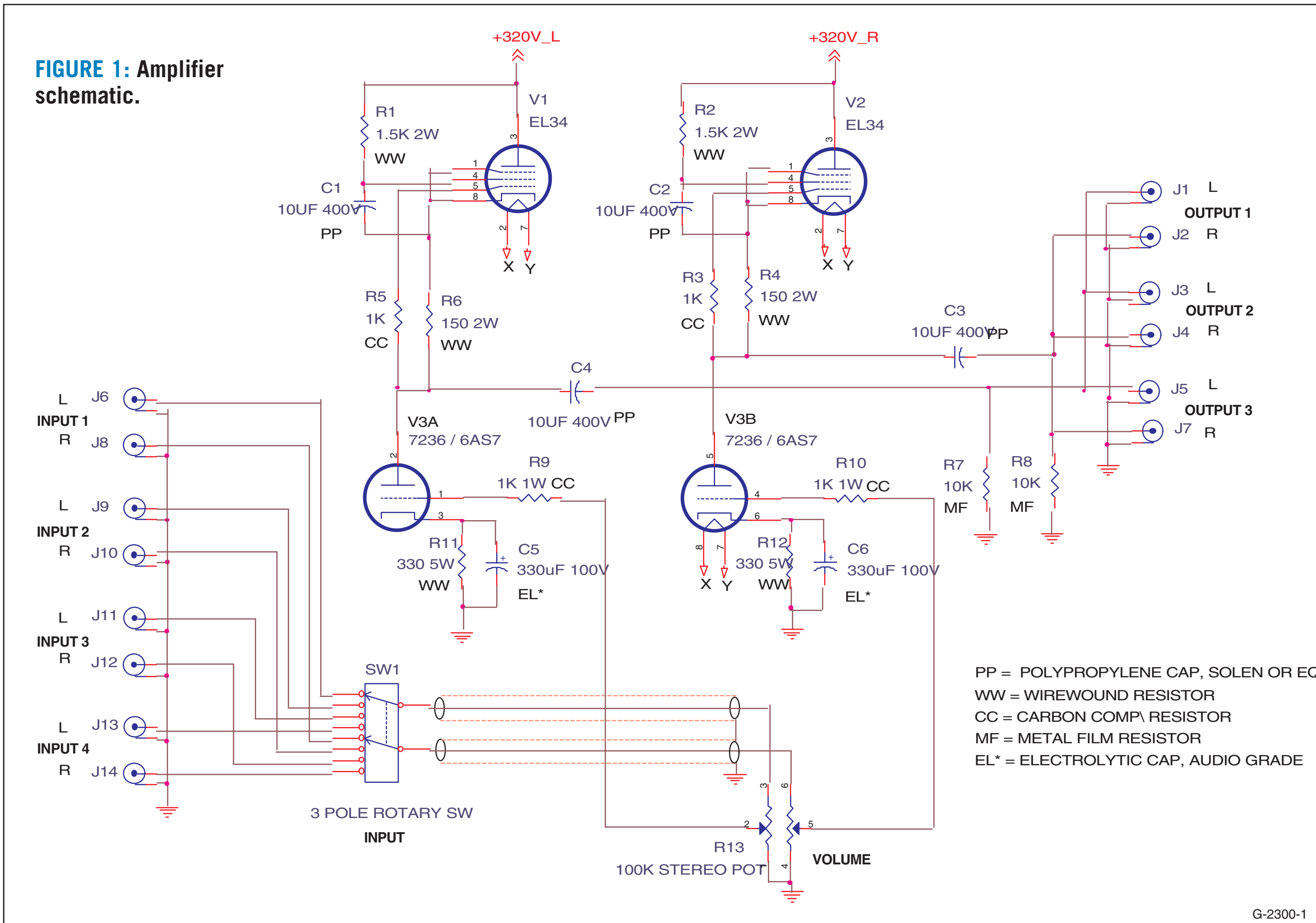


FIGURE 1: Amplifier schematic.

load uses an active device (usually a tube or FET) that's connected so that it always wants to supply a constant current to its load, which in this case is the plate of a triode. This provides a very high resistance load to the plate—essentially, a near-horizontal load line. With a CCS load, the load line that the tube sees is dominated by the current that's delivered to the load.

There are many possible designs for CCS plate loads—some very simple (just a depletion-mode FET and a resistor), and some very complex, using both tubes and solid-state devices. If you're interested in CCS plate load designs, there is a gold mine of info about them on the Internet (see "References" at the end of the article for some places to look).

I experimented with simple designs using an FET and a pentode. They both performed well. I opted for a pentode, partly because I had them—they were simple and performed well in this application—and partly because they don't need a heatsink. Using a tube lets you dissipate the power above the chassis!

One of the problems with using a CCS—especially a tube CCS—is that you need to have substantial voltage across them, especially if the output voltage swing is substantial. With a low-mu power triode preamp, this isn't too much of an issue; the output swing is only a few volts, and the triode can be happily run at low plate voltage, leaving more of the available power supply voltage to drop across the CCSs.

THE PREAMP DESIGN

The preamp design is straightforward, using EL34 pentodes wired as CCSs as the plate load for the low-mu triode. The triode tube has two sections, and one is used for each channel. The schematic for the amplifier is shown in *Fig. 1*.

Note that on the EL34 CCSs, the screen grid is bypassed by a large capacitor to the cathode of the tube. A screen resistor is used not so much to reduce the voltage on the screen, but to allow the capacitor to make the screen voltage constant with respect to the cathode.

The current through the CCS (and the triode) is set by the resistor in the cathode of the EL34. The 150Ω resistor shown sets the stage current at about 50mA. Note that the grid is connected

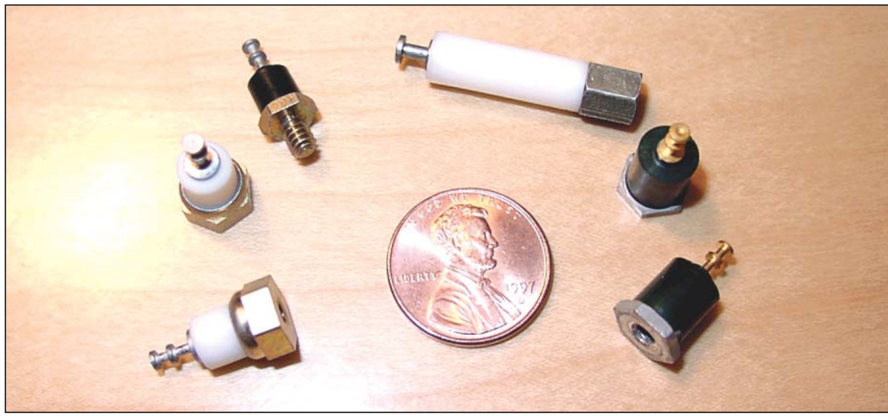


PHOTO 3: Single-point terminals.

to the lower end of this resistor, through a grid stopper resistor. This resistor is critical—without it, my amplifier oscillated at 115MHz, with 100V RMS on the triode plate!

The triodes are cathode biased with a 330Ω resistor. With a 7236 tube and a 50mA CCS plate load, this sets the plate voltage at about 125V. I experimented with different operating points using the 7236 tube, but found that this was about the optimum for lowest distortion.

I bypassed the cathode resistors with a good-quality audio electrolytic capacitor. At first glance it seems as though the cap is not needed, since the plate current is fixed by the CCS; however, the LOAD current still flows through the cathode resistor, and not bypassing it provides some local negative feedback, which reduces the gain somewhat. In this case (I'm not sure why), it *increased* the distortion.

The triode grids are connected directly to the wiper of the volume control potentiometer. There is a compromise in selecting the volume control resistance: You want a large resistance to achieve a high input impedance, but a high resistance, in combination with the Miller capacitance of the tube, causes a high-frequency rolloff. I used a 250k pot, and as you can see from the measurements, obtained reasonable high frequency response. You will get better response from a 100k pot, and even better still with a 50k pot.

I wanted several inputs to be selectable with a switch, so I used a good-quality rotary switch to select one of four inputs to feed the volume control. The output is taken from the triode plate through a polypropylene coupling capacitor. I connected three sets of output jacks in parallel, so I could drive several loads at the same time.

POWER SUPPLY

The power supply is conventional, using a tube rectifier and capacitor-input filter. I used a 5U4GB rectifier

PHOTO 4: PCB, top view.

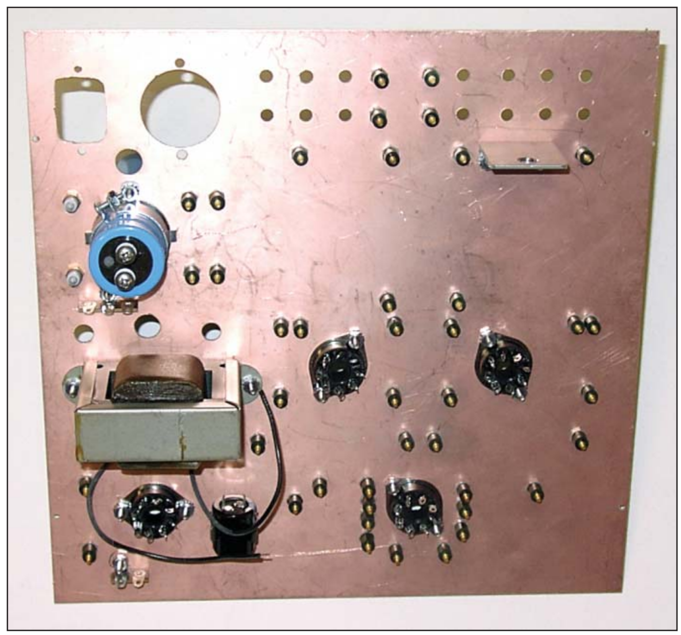
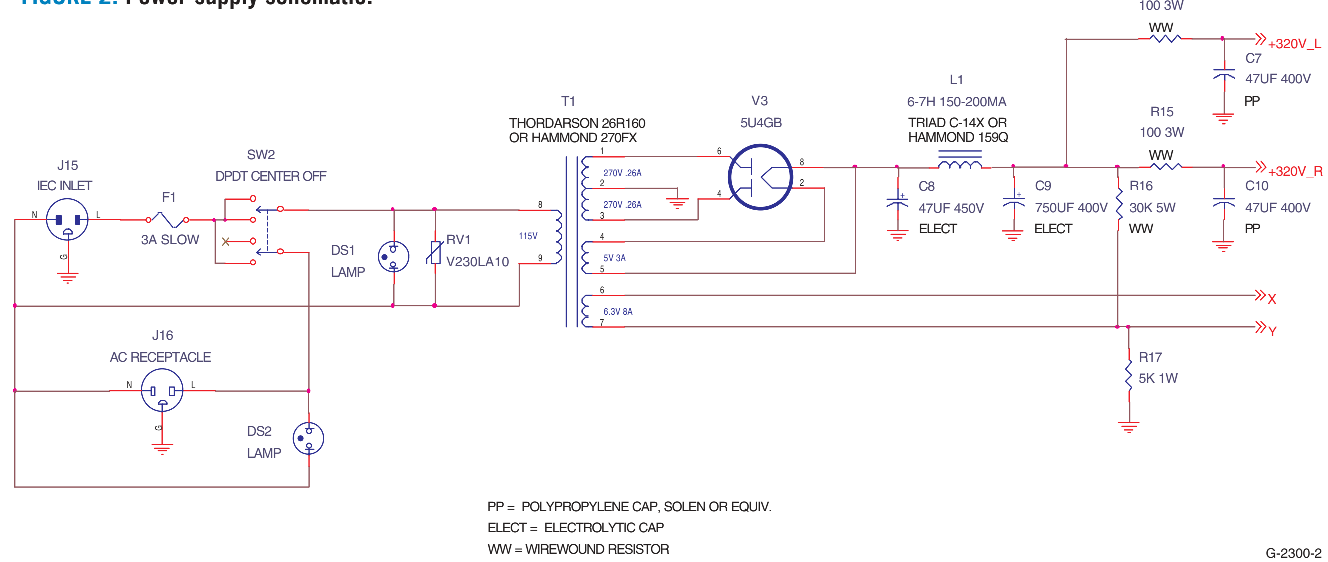


PHOTO 5: PCB, bottom view.

FIGURE 2: Power-supply schematic.



tube, but others would work as well, as the total current draw is only around 100mA. I used a large second capacitor in the filter, 750µF, to provide low ripple. The B+ voltage is between 300V and 325V, and in bench tests appears to be not very critical.

The audio circuitry is de-coupled from the power-supply filter by using 100Ω resistors and 47µF polypropylene capacitors. I believe that this scheme, which keeps almost all of the audio current out of the power-supply filter and instead sends it through the higher-quality polypropylene caps, sounds better than relying on the large electrolytic cap in the filter.

Filaments are powered with 6.3V AC, which is biased up to about 45V DC, derived from the power-supply bleeder resistor. This bias is needed to stay within the 100V maximum heater-cathode rating of the EL34 tubes, whose cathodes are operating at around 140V. It also helps reduce any noise coupling from the triode tube's filament, by reverse-biasing the intrinsic diode formed between the heater and cathode.

PHOTO 6: **Wood for base, cut and ready for assembly.**

I use the preamp power switch to control the power to my amplifiers, but there are times (such as when using a headphone amp) that I don't want to power them on but still use the preamp. To accomplish this, I used a center-off DPDT power switch. In one position, power is applied to just the preamp; in the other, power is applied to both the preamp and to an AC receptacle which is used to turn on the power amps. In the center "off" position, both the preamp and power amps are off. Neon pilot lamps are used to show what's turned on.

In the schematic, you'll see a metal-oxide varistor placed across the power transformer primary. This device is a transient suppressor, and helps to clamp both the line voltage spikes as well as the inductive spike that occurs when you turn the power off. I also used a filtered IEC power input connector to help keep noise out.

THE CHASSIS AND BASE

I built the preamp using a little different construction method than usual:

point-to-point wiring and single-point solder terminals. The chassis top is made up of a sandwich of PCB material and polished stainless steel sheet. The use of single-point terminals, as opposed to terminal strips, allows you to

FIGURE 3: Measured specifications.

7236 tube:

THD + noise (800Hz–80kHz bandpass):

1kHz, 1V RMS out, 100k load	0.05%
20kHz, 1V RMS out, 100k load	0.06%
1kHz, 1V RMS out, 10k load	0.07%
20kHz, 1V RMS out, 10k load	0.07%
1kHz, 2V RMS out, 100k load	0.10%
20kHz, 2V RMS out, 100k load	0.10%
1kHz, 2V RMS out, 10k load	0.14%
20kHz, 2V RMS out, 10k load	0.13%

Frequency response (using 250k pot–see text):

2V RMS in, 1V RMS out, 100k load			
	20Hz:-0.09dB	20kHz:-0.45dB	-3dB @ 65kHz
2V RMS in, 1V RMS out, 10k load			
	20Hz:-0.13dB	20kHz:-0.41dB	-3dB @ 70kHz
2V RMS in, 2V RMS out, 100k load			
	20Hz:-0.06dB	20kHz:-0.95dB	-3dB @ 40kHz
2V RMS in, 2V RMS out, 10k load			
	20Hz:-0.11dB	20kHz:-0.96dB	-3dB @ 40kHz

Output impedance: 450Ω

Noise: -55 dBV (mostly 60Hz)

Voltage gain: 3.96

6AS7G tube:

THD + noise (800Hz–80kHz bandpass):

1kHz, 1V RMS out, 100k load	0.15%
20kHz, 1V RMS out, 100k load	0.12%
1kHz, 1V RMS out, 10k load	0.16%
20kHz, 1V RMS out, 10k load	0.13%
1kHz, 2V RMS out, 100k load	0.18%
20kHz, 2V RMS out, 100k load	0.24%
1kHz, 2V RMS out, 10k load	0.33%
20kHz, 2V RMS out, 10k load	0.27%

Frequency response (using 250k pot–see text):

2V RMS in, 1V RMS out, 100k load			
	20Hz:-0.05dB	20kHz:-0.40dB	-3dB @ 65kHz
2V RMS in, 1V RMS out, 10k load			
	20Hz:-0.11dB	20kHz:-0.43dB	-3dB @ 65kHz
2V RMS in, 2V RMS out, 100k load			
	20Hz:-0.08dB	20kHz:-0.60dB	-3dB @ 55kHz
2V RMS in, 2V RMS out, 10k load			
	20Hz:-0.12dB	20kHz:-0.56dB	-3dB @ 55kHz

Output impedance: 324Ω

Noise: -55dBv (mostly 60Hz)

Voltage gain: 1.92

G-2300-3

place each end of each component exactly where you want them.

Single-point terminals are used in some mil-spec point-to-point wiring, and are available in a number of shapes, sizes, and styles. *Photo 3* shows an assortment of types. I used terminals with female threads, such as the ones at the right side of the picture (refer to the "Parts Sources" section at the end of the article for information on where to get these terminals).

The terminals are fastened to a piece of bare PCB material, using flathead screws that are countersunk flush with the surface. *Photo 4* shows the PCB from the top, with the flush screw heads. Other components inside the chassis are mounted to the PCB material the same way.

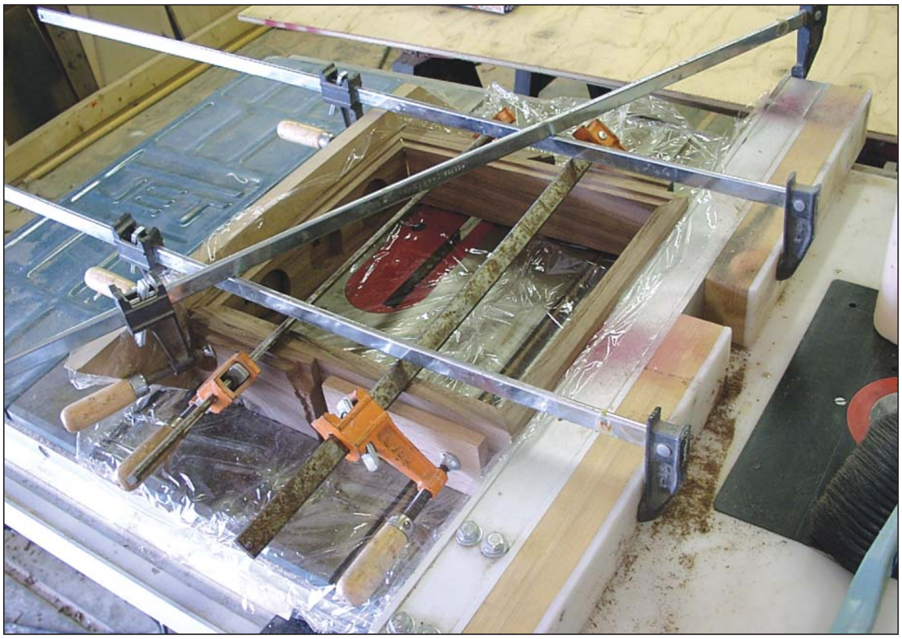


PHOTO 7: Gluing and clamping the base.

After I mounted all the terminals, tube sockets, and other components to the underside of the PC board, I attached a thin piece of polished stainless steel over the top, using an adhesive called "3M high-performance transfer tape." This stuff is no ordinary double-stick tape—it's an acrylic adhesive that forms a permanent bond. The "squishiness" of this material also helps to make the completed chassis acoustically dead.

The stainless steel covers all of the countersunk screws, giving a neat appearance on the outside. The only holes drilled or punched in the stainless are for components that mount on top of the chassis, such as the tubes and transformer. I purchased the polished stainless steel pre-cut in a 12″ × 12″ square from McMaster-Carr.

The copper foil on the inside of the PCB is used as a ground plane. You could argue that a well-thought-out single-point ground scheme is better than a ground plane, but I've found that a solid copper ground plane works very well, and is much easier to wire; you can just drop a component lead down and solder it to the ground plane. *Photo 5* shows the copper side of the PCB, with the terminals and components mounted. Note the bracket for the input selector switch, clamp for the electrolytic cap, and filter choke are mounted to the PCB.

The PCB and stainless-steel sandwich is mounted into a wooden base that I made from walnut. *Photo 6* shows the individual pieces of wood cut and ready to be assembled. Note the counter bores on the inside of the front piece to allow mounting the power switch and volume control. I glued the pieces together with a number of clamps as shown in *Photo 7*. The clamp going diagonally across the base pulls the base into square—you need to make sure the base is square, or your chassis won't fit together!

You can see the finished wooden base in *Photo 8*. The chassis top sits into a step that I routed into the wood pieces before gluing them together. Similarly, an aluminum plate attaches to the underside to form a bottom.

PHOTO 8: Completed wooden base.

WIRING AND COMPONENT INSTALLATION

After mounting the chassis "sandwich" into the base, you then install the remaining components (power transformer, RCA jacks, and so on). I used some copper foil tape to provide a ground connection to the volume control from the copper ground plane (*Photo 9*).

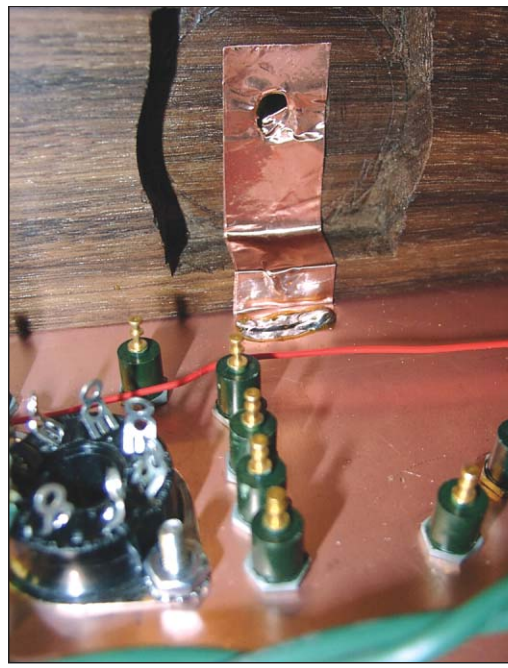


PHOTO 9: Inside chassis, showing volume control grounding.

I finished all of the wiring before installing any of the parts, using Teflon-insulated solid wire. This type of wire, though a little hard to strip, is nice to work with, because the solid conductor is easy to wrap around the terminals, and the insulation doesn't melt from the heat of a soldering iron. It's also easy to wrap around the single-point terminals and tube sockets without fraying as stranded wire would.

I mounted the input selector switch right above the input connectors on a bracket, and operated it with an extended shaft, to minimize the length of the signal wiring. After wiring, install the leaded components, wrapping their leads around the terminals and soldering. Adding the parts after the wiring makes it easier to change them after the fact, if you want to try different component values. A view of the inside of the completed chassis is shown in *Photo 10*.

MEASUREMENTS

I made a number of measurements (*Fig. 3*), using both a Cetron 7236 tube and a Russian 6AS7G tube. The measurements included THD, noise, frequency response, output impedance, and maximum voltage gain. In general, there were only small differences between tubes, other than the expected gain difference due to the tubes' differing amplification factors.

I also performed spectral analysis (FFTs) of the output distortion to check the harmonic content. I didn't include

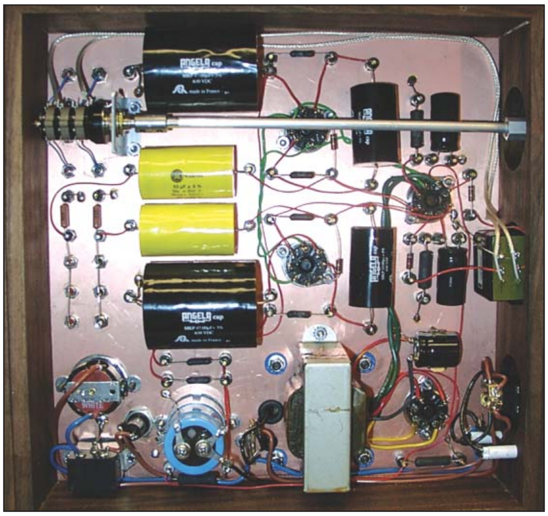


PHOTO 10: Inside of the chassis.

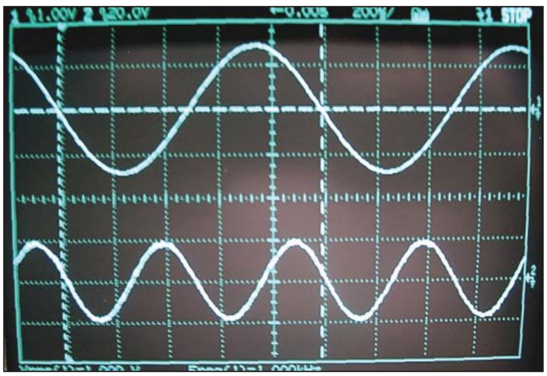


PHOTO 11: Output (top) and distortion residual (bottom, not to scale).

any pictures of them because they're not very interesting—the only harmonic that was above the noise was the 2nd harmonic. This was true for all of the conditions tested, and for both tubes.

Photo 11 shows an oscilloscope picture of the output (top trace) and the distortion residual (bottom trace). The distortion residual, shown at a much higher scale than the output signal, was filtered and averaged to remove all the noise components. You can see that it's pretty much a pure 2kHz sine wave.

As discussed, I used a 250k volume control pot, which is the limiting factor for high-frequency response. Using a lower resistance control will extend the high-frequency response considerably.

LISTENING IMPRESSIONS

I never thought much about what sonic impact my old preamp (a PAS-type 12AX7 circuit, with lots of negative feedback) was having on my system until I replaced it with this preamp. The biggest difference was in the clarity of high frequency transients. For example, the strike of a bell sounded *real*, not recorded.

I often don't like live recordings, because they tend to sound too muddy—there's so much low-level stuff in the background (such as the audience). This preamp seemed to resolve more of that detail, and live recordings I didn't like much before came alive. I was pleasantly surprised.

Of course, nothing is perfect. My only complaint is that the tubes are microphonic and a little noisy—especially while warming up. It's understandable—after all, these are big power tubes, and they weren't designed as low-noise, low-level amplifiers. After they've heated up for a while the noise diminishes, and I suspect some vibration dampers—maybe just a silicone O-ring—would help, especially on the tall, skinny EL34s. ❖

REFERENCES

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

www.bottlehead.com – The folks at Bottlehead were one of the first to popularize CCS plate loads. Check out their kits.

home.pacifier.com/~gpimm/ – The personal page of Gary Pimm, who has done an impressive amount of work perfecting CCS plate loads. Gary also sells PC boards of some of his designs.

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

PARTS SOURCES

McMaster-Carr
www.mcmaster.com
(630) 833-0300
Stainless steel and aluminum sheets, hardware, 3M high-performance transfer tape adhesive

Cambion
www.cambion.co.uk
(800) 947-1256
Single-point terminals
(part number 572-4827-01-05-16)

Concord Electronic Hardware
www.concord-elex.com
(800) 847-4129
Single-point terminals (part number 1127-30-0516)

USECO Fastners
www.staffall.com
(401) 461-5554
Single-point terminals (part number 1417)

The J-Low Project

Fortified with an ample supply of MDF, provisions, and perhaps too much vacation time on their hands, our intrepid designers reach new lows as they brave the minimalist path to undistorted bass.

By Nelson Pass and Dana Kruse

The aesthetic appeal of a single wide-bandwidth loudspeaker driver is obvious in its simplicity. There are no crossover networks, little or no phase shift versus frequency, and a single acoustic source location. As a concept, what's not to like? What could be more perfect than a nice little cone that could do it all, from the deep bottom end to tinkle beyond human hearing?

Unfortunately, there are good reasons why such speakers are uncommon—they are very difficult to make, relying more on hard work and good taste than your average loudspeaker part. They need to be a lot more than a stiff piece of paper, plastic, or metal.

IN THEORY

The "rigid piston" model of a mass-controlled loudspeaker diaphragm works remarkably well over a specific range of frequencies. For a given physical displacement, the acoustic output of a loudspeaker cone increases with the square of the frequency as long as the wavelength of the sound is large compared to the cone size. At approximately the frequency where the wavelength equals the circumference of the cone, this output levels off.

What does this mean? That for a given excursion distance a rigid circular type of loudspeaker will have a frequency response which rises 12dB per octave until the wavelength is three times the diameter of the circumference. For a 12″ speaker cone, this frequency is about 400Hz, above which the response should be relatively flat—for a given excursion distance.

That's very nice, except that the excursion of the cone is not independent of frequency. The electromotive force provided by current traveling through a voice coil in a magnetic field might be constant, but this force produces acceleration that must be translated to velocity, which is not the same thing, and then excursion, which is twice not the same thing. At the risk of mentioning calculus, we would say that the velocity of the moving assembly of a driver results from the integration of the force applied, and further, that the excursion is the integration of the velocity.

Well, what does that mean? Assuming that your loudspeaker voice coil and cone are much heavier than air (and they are), the excursion falls off at a rate of 12dB per octave as the frequency increases.

I would like to have been there on that happy day (probably at Bell Labs) when somebody noticed that the acoustic output rising at 12dB/octave nicely cancels the excursion falling off at 12dB/octave. After he ran out of champagne, this same scientist probably saw the flaw—this only works up to the frequency where the wavelength is larger than the circumference. Of course, perhaps back then they weren't quite as critical as we imagine ourselves today, and were pretty happy with response up to a couple thousand hertz.

There was another fly in the ointment in that the excursion of the cone needed to increase by a factor of four for every lower octave of output. At

PHOTO 1: Dana Kruse standing in front of the right channel loudspeaker.

some arbitrarily low frequency this becomes a problem, somewhere less than a couple inches.

So we end up with a piston model of the loudspeaker driver that for a given size is bracketed on the bottom frequencies by its excursion limits and on the top by a leveling off of the acoustic resistance. As we make the cone bigger, we can play louder at low frequencies, but we limit the high frequencies by the same proportion. As we make the cone smaller to get the high frequencies, we find that the cone can't travel far enough to give us undistorted bass. Alas.

ANOTHER WAY

If we choose to stick with something resembling the mass-controlled piston model, we have no choice—we must divide the job into multiple drivers. A 1″ diameter cone that would do a nice job with the bells on "Dark Side of the Moon" (hint: Pink Floyd) is not going to deliver on the bottom end.

Well, that's OK, and usually that's how it's done. But it's not the only way. If we ignore the rigid piston model, we can imagine cone materials that are flexible in such a way that the center of the cone is free to vibrate at a higher acceleration than the outer edge of the cone. Actually, I believe the Bell Labs guys thought of this the next day after the party.

Of course, the Egyptians—bless them—thought of paper first, but it so happens that carefully wrought paper materials have close to the right properties of not only being very lightweight, but also having the right balance between stiffness and flexibility. Combined with a certain amount of intrinsic damping, paper cones do a remarkable job of overcoming these problems.

They aren't perfect, but the audiophile attraction is there. Of the products on the market, most make some sort of effort at decoupling the low frequencies from the high. In my opinion, the ones that do the best job are the smaller ones, which is no surprise. This means they need some sort of help on the bottom end.

A DRIVER SOLUTION

Kent English at Pass Labs has a job description that includes acquisition of interesting drivers (he finds them, I sign the checks). He bought a pair of Jordan JX92Ss, which are full-range cone speakers with about 3.5″ diameter cones and some sort of metallic coating on the cones. Well aren't they cute, sez I, and they don't cost much, so one fine Saturday we put them into boxes and started playing with them.

We were most surprised. I would call them flat to 20kHz, and remarkably, they made it down below 50Hz in a modest box. *Figure 1* shows the wideband response curve at 1m, and *Fig. 2* shows low-frequency detail with the driver in a 3ft^3 box.

They sounded so good that we started cranking them up, and immediately ran into the distortion from the high excursion in the bass. Alright, so they did not play that loud, but they were still very pleasant to listen to over the long term.

And so it remained, listening at low levels in the dead of night, until Dana Kruse showed up. Now Dana is not your average audiophile. Supposedly a successful architect in Seattle, he has spent many of his vacations in

Foresthill sitting on the production line building amplifiers, presumably for therapy. Dana Kruse is also dangerously witty in an abstract way, so that almost none of the remarks or scatological diagram titles he has contributed will appear in this article.

It happens that he also owns a pair of Jordan JX92Ss, so we decided that for this vacation we would do something with them. Designing at the table saw is a particular specialty of Dana's, and so we went straight out to the woodshop.

We needed more output level on the bottom end for these speakers, and horns looked like the best way to get it. Not front-loaded, which would obscure the high frequencies, but some nice rear-loaded boxes—something which would give us intimate access to the front wave but back it up with a little authority on the bass.

Horn loading is a well-understood science. As Leo Beranek points out in his classic book on acoustics, a horn is an acoustic transformer, turning a small diaphragm into a big one without cone resonance. The most common shape for a horn is an exponential curve, where the surface area down the length of the horn is given by the equation:

$$S = T * e^{(M * D)}$$

where S is the cross-sectional area at the distance from the throat D whose cross-sectional area is T. M is the flare constant given by:

$$M = 4 * pi * F/C$$

where F is the cutoff frequency of the horn, and C is the speed of sound.

As a practical matter, the cutoff frequency is where you have no resistive output for the horn, so a practical horn operates at –3dB at about 1.4 times the value of F. Another consideration for the horn is that the mouth of the horn should have a circumference greater than the wavelength of the lowest frequency to be amplified or have the equivalent area.

It's pretty straightforward—the lower the frequency you want to take your horn, the bigger it will need to be. The following table illustrates the incremental distance from the throat it takes to double the cross-sectional area for various cutoff frequencies:

60Hz	12″
50Hz	14″
40Hz	18″
30Hz	24″
20Hz	36″
15Hz	48″

Of course, a cutoff of 15Hz means that the horn would be usable down to about 20Hz. The mouth area required follows a similar pattern:

60Hz	37ft²
50Hz	54ft²

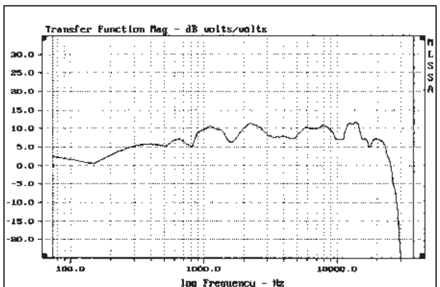


FIGURE 1: Jordan JX92S full-range response at 1m. B-2337-1

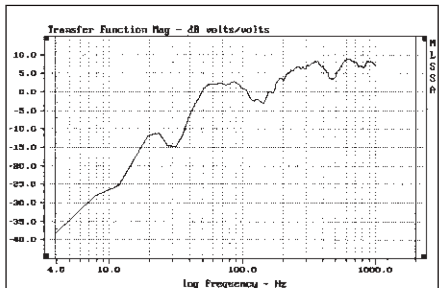


FIGURE 2: Jordan JX92S near-field bass response. B-2337-2

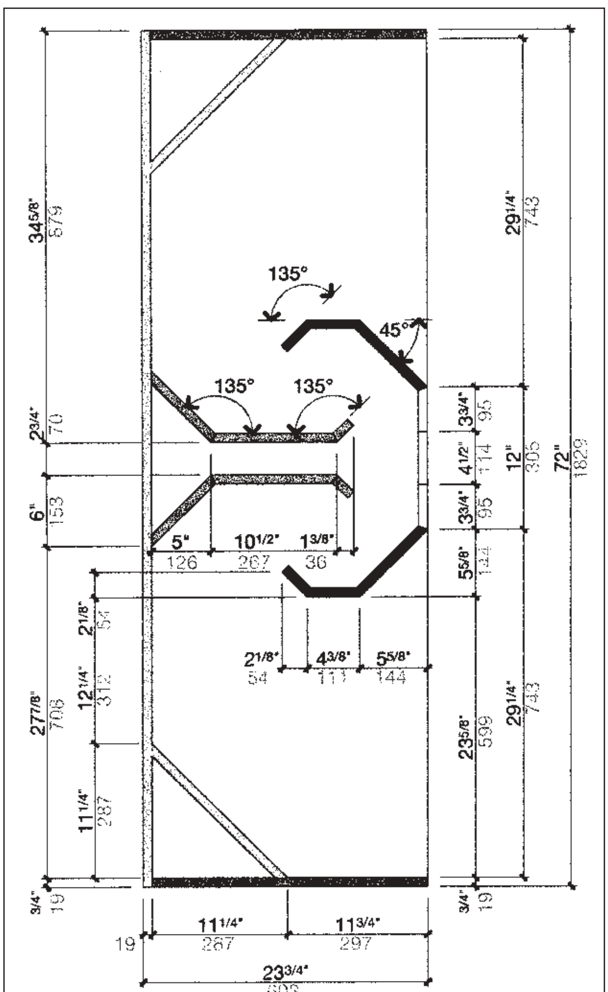


FIGURE 3: Side view. B-2337-3

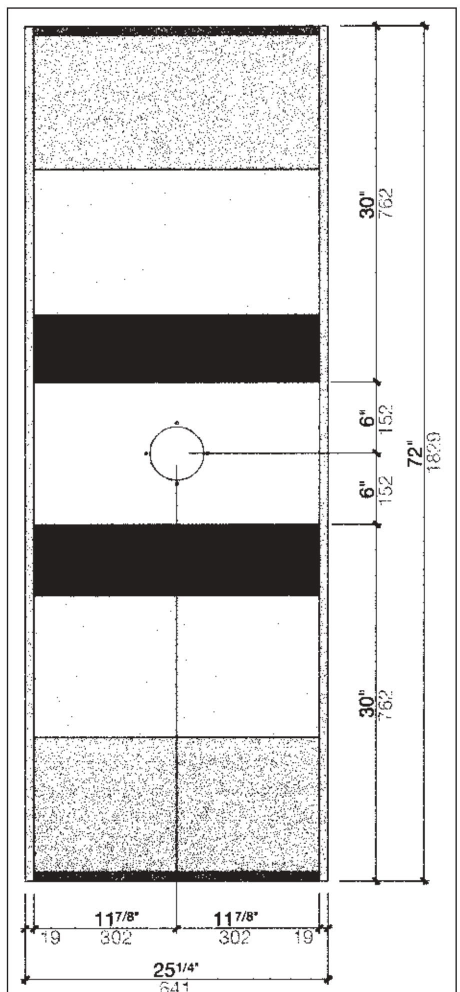


FIGURE 4: Front view. B-2337-4

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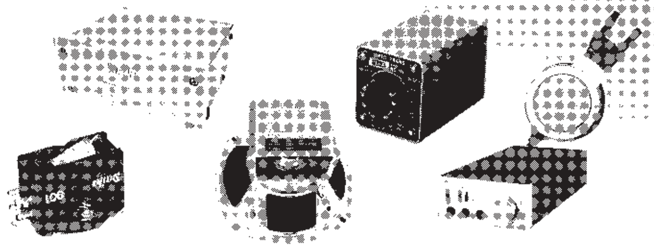

■Cartridge

Model	Price (US$)	Postage (Air Economy)	
Denon DL-102 (MONO)	150	Area I $18	China, Korea, Hong Kong, Taiwan
Denon DL-103 (STEREO)	200	Area II $22	Singapore, Malaysia, Indonesia
Denon DL-103R (STEREO)	250	Area III $27	North America, Oceania, Europe
Denon DL-103 PRO (STEREO)	350	Area IV $34	Africa, South America
Shelter Model 501 II (CROWN JEWEL REFERENCE)	750	These Area I ~ IV are for all products except book.	
Shelter Model 901 (CROWN JEWEL SE)	1,400		

■Japanease Audio Book

Postage $15

Title		Price (US$)
Attractive Tube Amps Vol. 1&2	(Isamu Asano)	30 each
The Joy of Vintage Tube Amps 1&2	(Tadaatsu Atarashi)	30 each NEW
Direct & Indirect Tube Amps	(Kiyokazu Matsunami)	40 NEW
SE Amps by Transmitting Tubes	(Kouichi Shishido)	50
The Remembrance of Sound Post	(Susumu Sakuma)	30
Classic Valve	(Hisashi Ohtsuka)	40
MJ Selected 300B Amps	(MJ)	30
Top-Sounding Vintage Power Tubes	(Stereo Sound)	30
Output Trans of The World	(Stereo Sound)	30
20TH CENTURY OF AUDIO	(Stereo Sound)	30
Vintage Speaker Units	(Stereo Sound)	30 NEW
Tube Amp Craft Guide	(MJ)	30

■MC STEP UP TRANS

Model	Specifications			Price (US$)	Postage**
	Pri.Imp(Ω)	Sec.Imp(kΩ)	Response		
Shelter Model 411	3~15	47	20Hz~50kHz	980	Area I $25
Jensen JE-34K-DX	3	47	20Hz~20kHz	550	Area II $30 Area III $40
Peerless 4722	38	50	20Hz~20kHz	300	Area IV $50

**Air Economy

■STAX

Model	Price (US$)
OMEGA II System (SR-007+SRM-007t)	Ask
SRS-5050 System W MK II	Ask
SRS-4040 Signature System II	Ask
SRS-3030 Classic System II	Ask
SRS-2020 Basic System II	Ask
SR-001 MK2 (S-001 MK II+SRM-001)	Ask

■Speaker

Model	Specifications					Price* (US$)	Postage** (US$)			
	D (cm)	Ω	Response	db	w		I	II	III	IV
Fostex FE208Σ	20	8	45Hz~20kHz	96.5	100	296	62	74	120	156
Fostex FE168Σ	16	8	60Hz~20kHz	94	80	236	42	50	73	98

*Price is for a pair **Air Economy

■TANGO TRANS (ISO) 〔40models are available now〕

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US$)	I	II	III	IV
XE-20S (SE OPT)	20	2.5, 3.5, 5	20Hz~90kHz	300B,50,2A3	396	47	56	84	113
U-808 (SE OPT)	25	2, 2.5, 3.5, 5	20Hz~65kHz	6L6,50,2A3	242	42	50	73	98
XE-60-5 (PP OPT)	60	5	4Hz~80kHz	300B,KT-88,EL34	620	62	74	115	156
FX-40-5 (PP OPT)	40	5	4Hz~80kHz	2A3,EL34,6L6	320	47	56	84	113
FC-30-3.5S (SE OPT) 〔XE-60-3.5S〕	30	3.5	20Hz~100kHz	300B,50,PX-25	620	62	74	115	156
FC-30-10S (SE OPT) 〔XE-60-10SNF〕	30	10	30Hz~50kHz	211,845	620	62	74	115	156
X-10SF〔X-10S〕	40	10W/SG Tap	20Hz~55kHz	211,845	1160	90	110	180	251
NC-14 (Interstage)	—	[1+1:1+1]5	25Hz~40kHz	[30mA] 6V6(T)	264	30	40	50	70
NC-16 (Interstage)	—	[1+1:2+2]7	25Hz~20kHz	[15mA] 6SN7	264	30	40	50	70
NC-20F [NC-20] (Interstage)	—	[1:1]5	18Hz~80kHz	[30mA] 6V6(T)	640	42	50	73	98
NP-126 (Pre Out)	—	20,10	20Hz~30kHz	[10mA] 6SN7	264	30	40	50	70

Postage** (US$): columns I–IV. Price is for a Pair

**Air Economy

■TAMURA TRANS (All models are available)

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US$)	I	II	III	IV
F-7002 (Permalloy)	10	3.5	15Hz~50kHz	300B,50	836	60	70	110	145
F-7003 (Permalloy)	10	5	15Hz~50kHz	300B,50	836	60	70	110	145
F-2013	40	10	20Hz~50kHz	211,242	786	70	84	133	181
F-5002 (Amorphous)	8	3	10Hz~100kHz	300B,2A3	1276	65	80	120	160

Price is for a Pair

** Air Economy

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E-mail:info@eifl.co.jp
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Card Charge 6%

40Hz	$84ft^2$
30Hz	$149ft^2$
20Hz	$336ft^2$

DUST AND GLUE

Designing at the table saw is also a well-understood science. You look around and see what kind of pieces of wood you have. Fortunately I had quite a few sheets of MDF with a nice oak veneer, and so we sketched up a nice rear-loaded design, checking the expansion through trial and error with ruler and compass.

Dana comments:
"After cutting the pieces on the table saw, we introduced some beer into the equation, resulting in an alternate geometry as described below:

$$S = T * e^{(M* G* D) + :) + :)}$$

With the new-found efficiency this formula afforded us, we were able to take a little time away from the dust and glue to prepare an impromptu meal of Tostitos, bread, wine, bread, DiGiorno, bread, beer, bread, steak, bread and parmesan, chased with bread, champagne, wine, beer, and ice cream. Reflecting upon what we had accomplished the next morning at our 8 AM Philip Kaufman screening, it seemed that the flurry of snappy banter and the complete absence of flesh-meets-whirling-steel, added up to a pretty fair day in the shop. With coffee, peanut butter, bread, honey, bread, and Equal in our glue-encrusted and splinter-pricked hands, we toasted to a job well done and looked forward to what revelation this day would bring."

The design is pretty simple, as you can see from *Figs. 3* and *4*. We were tempted to go with a full-size MDF sheet, but decided to limit ourselves to something that would actually fit through the listening room door. This placed a serious restriction on how low a frequency we thought we were going to get away with, and we decided on about a 35Hz cutoff taper (which would take us down to about 50Hz in reality) and hope that the $20ft^2$ mouth areas of the combined speakers would get some help from the floor and back wall in a $30 \times 30'$ room.

Dana later generated a full-boat set of diagrams in glorious color, with cutting and assembly instructions which is too large to reproduce here. You can download the pdf file from www.passdiy.com.

HOW IT SOUNDS

Photo 1 is a picture of Dana standing in front of the right channel loudspeaker. What appears to be a cute little on/off button at the front of the speaker box is the Jordan JX92S. We conducted our listening tests with a 40W balanced version of the Zen Lite, which is the box you see with the light bulbs on top.

Going back to *Fig. 3*, you see a chamber behind the driver that opens up into upper and lower horn throat areas. This chamber before the horn throat helps to form a low-pass filter that reduces the amount of high frequencies that will pass through the horn. This is an important item, as you don't want to be listening to the rear wave above 100Hz, which will interfere with the front wave. This acoustically capacitive chamber should also be filled with absorbent material, such as Dacron, wool, or fiberglass. We chose Dacron.

You can easily tune this acoustic low-pass filter by altering the volume of the chamber behind the driver, and in our case we played with the volume by altering the density of the Dacron.

The result was simple and dumb, but very effective. The JX92Ss didn't go any lower in frequency than before, but they picked up about 10dB of gain centered in the 70Hz region, and in fact became bass heavy. This was what we were looking for.

Figure 5 shows the near-field response of the result. *Figure 6* shows the smoothed response full range as taken from the listening position. Between measurement and listening we concluded that the bottom end was about 6dB too much at about 70Hz.

This put us in a fine position to apply a high-pass filter to the system. We played with single-pole high-pass filters at 50, 100, 130, and 180Hz, and settled on 100Hz as the most satisfying. *Figure 7* shows a more detailed response of the system as seen from the listening position with the four different filters applied.

Previously the speakers delivered decent bottom end, but they couldn't play loud. Now with rear-loaded horns and a high-pass filter we get a similar response curve, but can play about 10dB louder. In addition, the bass response also picked up a qualitative improvement in its dynamic quality. I have remained happy listening to them for months now.

Dana had to go back home to design skyscrapers for Microsoft, but Kent is continuing to acquire full-range drivers, and we have started construction on a larger set of horns based loosely on this design, but which must be assembled on site, as they will not fit through any ordinary doors. We will be testing these drivers shortly, starting with the Mangers, and a follow-up article will appear. ❖

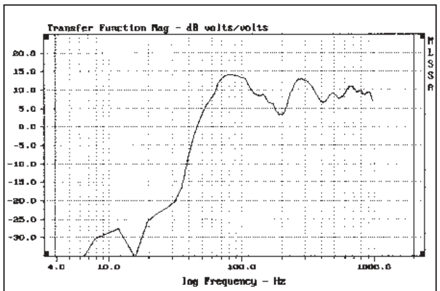


FIGURE 5: Near-field bass response in horn. B-2337-5

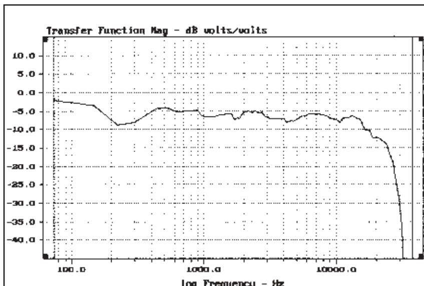


FIGURE 6: Listening position full-range response. B-2337-6

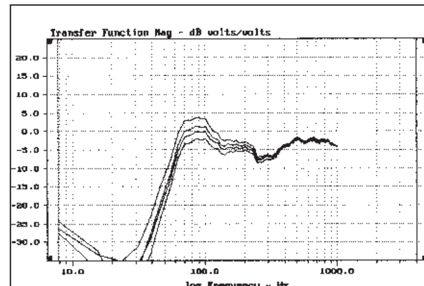


FIGURE 7: Listening position bass response with high-pass filters. B-2337-7

Rebuilding a Classic: McIntosh MC2100

This enthusiast brings new life to a classic piece of amp equipment.

By Bruce Brown

PHOTO 1: The refurbished McIntosh 2100 amp.

I have been an avid audio enthusiast for over 30 years. I started with tubes, went on to transistors (first germanium, then silicon, finally MOSFETs), then back to tubes. I have a mixture of tube and transistor equipment, mostly transistor tuners, some preamps, and both tube and transistor power amps, depending on my project's needs. I have designed and assembled many scratch-built preamps and amps over the years, but most of my work has been to rebuild and modify Heath and Dynaco amps because of their availability and low cost.

MAC ATTACK

About a year ago I was having my Harley-Davidson motorcycle serviced and struck up a conversation with the shop owner, who also happened to be a "tube head." We began comparing notes and listening to each other's equipment. He had been rebuilding and modifying McIntosh equipment for many years and was completely sold on their quality and sound.

After listening to his components, I decided I needed a MAC. He suggested one of MAC's auto transformer-transistor amps, and introduced me to his 2100. I really liked the heft and the sound of it, even driving my inefficient JBL 200s.

The 2100 is rated at 105W/channel into 4, 8, or 16Ω, courtesy of the audio transformers. It also has a front panel switch to bridge it to mono operation at 210W into 2, 4, or 8Ω. Other interesting specs on this amp include a THD maximum of 0.25% at all power levels (within the frequency range of 20–20kHz +0–0.25dB, and 10–100kHz +0–3.0dB) and IM distortion 0.25% at any frequency (from 20–20kHz at peak power levels up to 210W/channel).

The amp is compact, with a total of 47 semiconductors, including 12 TO3 power transistors. It weighs a hefty 57 lb and is built like a battleship! These amps are relatively common due to high consumer, commercial, and industrial use, and due to their ability to take extraordinary abuse.

Several months ago, I read in *audioXpress* the article on "Autoformers" (Jan. '03. p. 38) and found it interesting that the author did not comment that McIntosh had used this concept in transistorized amps for over 30 years! The concept has remained the same through this time, to provide a consistent load on the output transistors, regardless of the impedance of the speaker load (within reason).

I bought a "homeless" MC2100 via online auction (*Photo 1*). The seller stated it had "worked" when he brought it up on a Variac®, but after he cleaned it up a little it would blow fuses at about 50V.

INSPECTION

When it arrived, it was ugly (*Photo 2*)! I have seen some homely ST 70s and Mark IIIs, but this made them seem pristine. The cage, chassis, and even the base were rusty. Under layers of "grunge" on the chrome chassis was more rust. But my inspection showed that it was unmolested by soldering iron or drill. I obtained a copy of the service manual from a seller on-line and proceeded to check it out.

I examined the transformers, power-supply diodes, and power transistors. My motorcycle friend allowed me to test the plug-in driver boards in his 2100. They were fine and very quiet in his amp. I found one output transistor open, but also there were very low resistance readings from the cases of all the output transistors to chassis.

McIntosh relied on heavy anodizing to insulate the cases from the heatsinks and used no mica insulators. This was a common design issue with MACs and seemed to work just fine, unless the

PARTS LIST

3— packs NTE Thermo-pads (Mouser-526-NTETP0001)

OPTIONAL

2— 56,000µF 50V computer-grade electrolytics (Mouser-539-EGS50V56000) (C201, C202)
2— 500µF 25V electrolytics (Mouser-75-TVA1207) (C303, C304)
2— 100µF 15V electrolytics (Mouser-75-TVA1162) (C307, C308)
2— 10µF 25V electrolytics (Mouser-140-TG100M1E-0512) (C309, C310)
12— 2N3772 TO3 transistors (order 25-30 if you wish to match them) Electronic Goldmine (www.Goldmine.Elec.com) (Q15-Q26)
1— 100V 25A bridge rectifier (Mouser-625-GBPC25000W) (D201-D204)
1 each— input jacks (Mouser-161-0251 and 161-0252) red and black rings

CONTACT INFORMATION

Mouser Electronics—www.mouser.com, 1-800-346-6873
Electronic Goldmine—www.goldmine-elec.com, 1-800-445-0697
Bruce Brown—Tuninfork@aol.com, 1-605-582-6762

unit was exposed to moisture or corrosion. The previous owner's attempt at cleaning may have allowed some solution to seep between the heatsinks and the transistor cases. No wonder it blew fuses! I also found what appeared to be corrosion under the plastic film that encases the 39,000µF 40V power-supply caps on the top of the chassis.

I discovered that the MAC transistors were not very well matched. This might be by design or just age, but either way I decided to replace them all. I selected some matched 2N3772 output transistors (high voltage versions of the 2N3055, obtained from the Electronic Goldmine for $1.50 each), which will handle up to 80V.

These transistors were used in Dynaco ST and SCA 80 amps as well as ST120s and ST 150s. I had about 25 of them, so I matched them very closely for leakage and gain. When I installed these on the 2100 heatsinks, I used NTE Thermo-pads to make sure they would not ground to the chassis again (*Photo 3*).

I slowly brought the amp up on my Variac, and at about 60V I had music. I listened to the great sound for over a week and decided to keep the amp. Now I needed to do something about its appearance.

EXTREME MAKEOVER

I decided a complete rebuild was in order, and started to pore through some catalogs to find replacement power-supply capacitors. Mouser Electronics had some 56,000µF 50V computer-grade capacitors with the same diameter and terminal spacing as the original 39,000µF units in the 2100. (If you attempt to use different caps, be aware that the mounting system used by MAC means you either drill new holes in the chassis or figure out a different system.) My choice was fortuitous, since this increased the power-supply transient storage capacity. I also decided to replace all the electrolytics on the preamp board (see parts list) and phono input jacks crimped onto the front panel.

While I was waiting for parts, I stripped down the chassis for refinishing as follows: remove the plug-in driver boards and put them in a safe place. (You will need to drill out about ten factory rivets that hold the terminal strips and the octal socket on the front panel. I used a ⅛″ bit and drilled from the inside of the chassis.) Then use a soldering iron to disconnect the power transformer leads and the audio transformer leads from the terminal strip behind the speaker connection blocks. (A word of caution here: my wiring colors had faded and it was necessary to make a pictorial diagram of the chassis wiring and label leads as I removed them.)

Unbolt each of the transformers and pull the leads through the chassis. When you remove them you will have the center chassis/power-supply divider loose also. This will make the chassis much lighter and easier to handle (*Photo 4*). You can use cable ties to keep the wiring harnesses separate. When you unbolt the preamp board and the heatsinks, everything should be free and you can then gently remove all from the chassis. I also removed the tinnerman nuts from the logo on the base and set them aside.

The cage, chassis, and base went to the sandblaster, then to the powder coater. I had the chassis done in "antique silver," which looks like a hammered finish used on older electronic equipment. To preserve the grounding surface, I had the powder coater mask off the area where the chassis divider bolts back in.

The cage and base were done in a light textured satin black that is very durable and tough (*Photo 5*). I also masked off the decals on the tops of the transformers and painted them with three coats of Rustoleum™ light texture paint (*Photo 6*). When you get the chas-

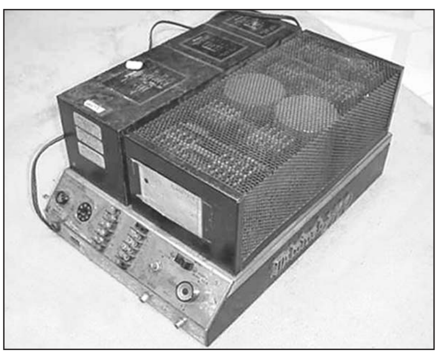


PHOTO 2: The McIntosh 2100 amp, upon first inspection.

PHOTO 3: Transistor replacements.

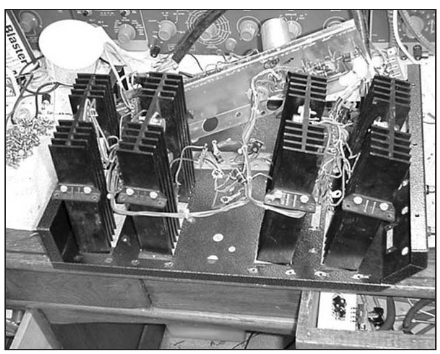


PHOTO 4: Disassembling the unit.

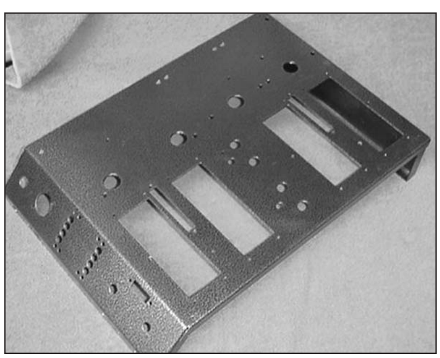


PHOTO 5: Refurbished chassis.

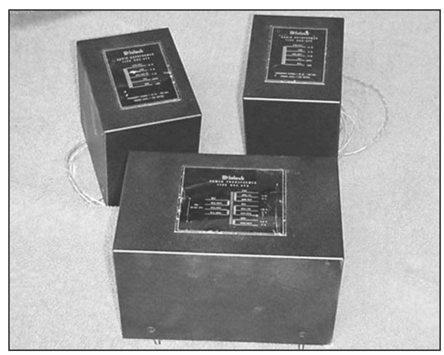


PHOTO 6: Sprucing up the transformers.

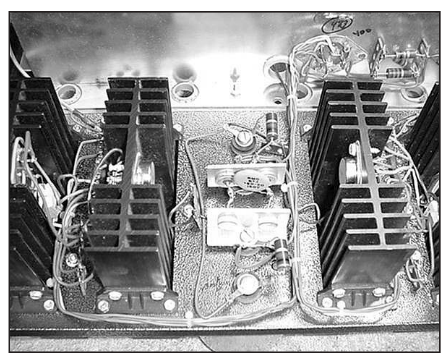


PHOTO 7: Installation.

PHOTO 8: Reassembled unit.

sis back, use a Dremel™ tool with a carbide bit to remove the powder coating from the underside of the chassis around all the mounting holes for terminal strips and heatsinks.

I took the opportunity to replace the front panel AC outlet with a lighted rocker switch that fit in the original mounting hole. (I found this switch at a local surplus electronics store; a standard slide switch will also work with a little modification.)

Another option now is to replace the large bridge diode assembly that mounts to the bottom of the large caps. Originally, McIntosh used four large diodes pressed into L channel aluminum; later they used a large epoxy bridge. You can use a 100V 25A bridge to update your amp (*Photo 7*).

Start reassembly by remounting the heatsinks, preamp board, and terminal strips with new hardware replacing the rivets. Next, remount the transformers and snake the wiring back to its original locations. As you re-solder the wiring, check several times to make sure it is correct (*Photo 8*). You can also use cable ties to secure everything.

After reassembly, I hooked up the amp to some speakers, a tuner, and the Variac, and brought the power up slowly. Again, at about 50V I had music! After checking various voltages and replacing the cage and base, I moved it to my listening system, consisting of a Carver CT-7 tuner/preamp, a Carver CD Player, and JBL L200 Studio Master speakers, which require at least 100W/channel to adequately drive them.

How does it sound? Great! It has tremendous headroom and rivals any transistor amp I have ever used. Best of all, it looks good, and I trust it will serve another 40 years without a problem. ❖

Adcom's GFP-565 Preamplifier, Part 4

This final installment describes the new phono preamp. **By Gary Galo**

Adcom's GFP-565 preamplifier came with an excellent phono preamp designed by Walt Jung. The circuit (*Fig. 8*) is no secret, since Linear Technology published a nearly identical topology on the front page of their LT 1115 op-amp data sheet (I thank Victor Campos for granting permission to reproduce the circuit back when he was Vice President for Product Development at Adcom).

The gain block is a Linear Technology LT1028C op amp (Adcom 8A, and predecessor to the LT1115), buffered by the high-current LT1010CT video buffer (Adcom 1A). R119, the 49.9Ω bias resistor, sets the LT1010 output stage for pure Class-A operation. D101, a J555 JFET current source, forces the LT1028 output stage into Class-A operation. The LT1028 was the quietest op amp available when the 565 preamp was designed—voltage noise was specified at 1.1nV√Hz at 1kHz. To keep the preamp noise as low as possible, the 565 phono preamp has a low impedance RIAA feedback loop.

The LT1010 buffer provides plenty of output current for linear drive of the low-Z network. C107 (bypassed by film cap C109) keeps the DC gain at unity, making the DC offset manageable. The LT1028C has very low input bias current (for a bipolar-input device), and low DC offset, allowing DC coupling of the output (R115 provides a slight offset trim for the circuit).

The RIAA network capacitors—C111, C113, and C115—are 1% tolerance Roederstein KP-series polypropylene types. The 565 phono preamp has excellent RIAA accuracy, ±0.1dB, 20Hz–20kHz. This was a very fine phono preamp by late-1980s standards, and it would be very easy for a modifier to make it worse. A "safe" modification might consist of simply replacing the 1% Roederstein/Resista MK2 resistors with better types.

So why design a new phono preamp? The original Adcom topology has one sonic limitation that can't be fully solved with parts substitutions. The biggest sonic snags in this design are the electrolytic caps that keep the gain unity at DC. My goal in designing the new phono preamp was to eliminate these capacitors by using a gain block with lower offset and lower input bias current, and controlling the residual DC offset with a servo.

I settled on Analog Devices' AD745J, an FET-input op amp with ultra-low voltage noise (2.9nV√Hz at 1kHz) and 12.5V/µS slew rate (compared to 11V/µS for the LT1028). Typical input offset voltage is 0.25mV, and input bias current is 150pA.

The input bias current is the critical spec. The LT1028C specifies this as 40nA. The significantly lower input bias current of the AD745 maintains stable

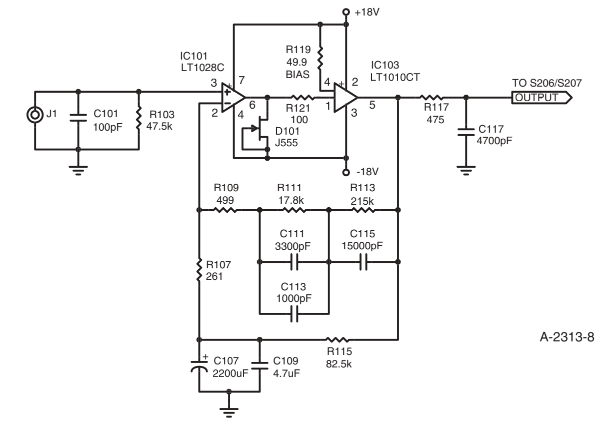


FIGURE 8: The original Adcom phono preamp. The preamp uses a Linear Technology LT1028 op amp buffered by an LT1010 video buffer. Linear Technology published a nearly identical circuit in the datasheet for the LT1115.

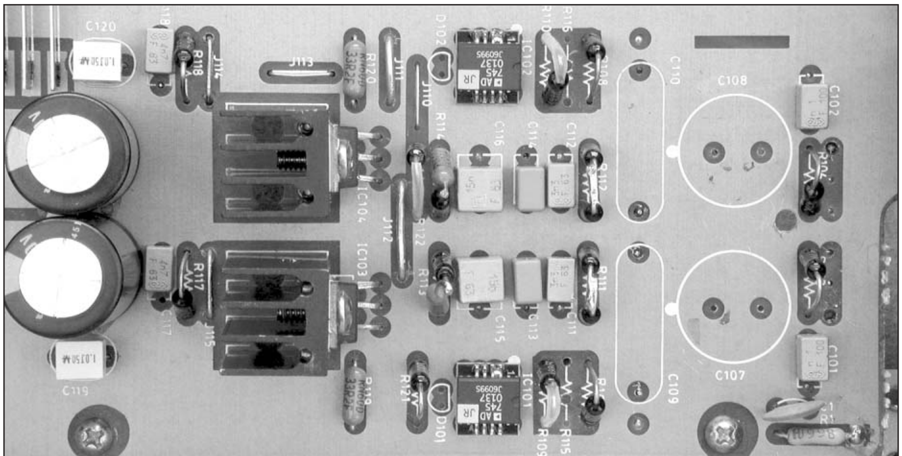
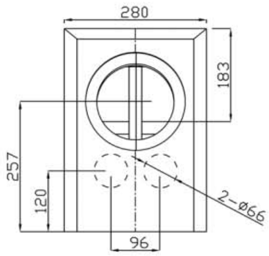


PHOTO 32: The modified phono preamp before the installation of the DC servo. C107 through C110 have been removed, and the LT1010 heatsinks have been replaced.

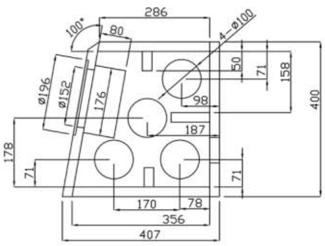
Swans Tempus Kit

Swans proudly presents their first European-style, independently developed loudspeaker kit. Due to exceptionally high standards for a kit speaker, TEMPUS is a top-level, world-class performer in all regards.

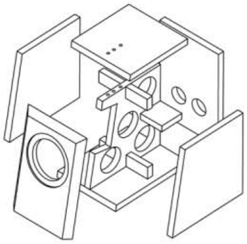
German sound magazine KLANG+TON featured the Tempus in the June 2002 issue, commending its excellent acoustic response.



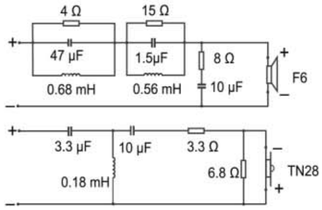
Front View



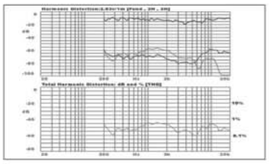
Section Side View



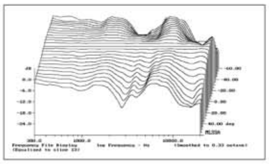
Explosion View



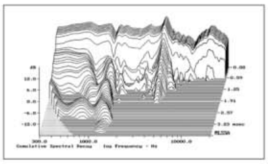
Filter Circuit



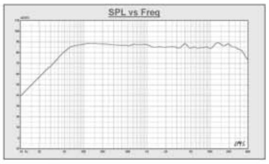
Tempus Harmonic Distortion



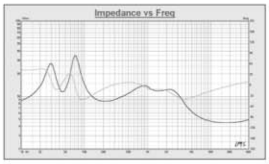
Tempus Horizontal response +60deg ~ - 60deg



Tempus Cumulative Spectral Decay



Tempus Frequency Response



Tempus Impedance

The Tempus project was initially conceived as a private work of the acoustic arts, to be executed independent of all professional affiliations. Despite a lack of traditional marketing and initial editorial commentary, the design has since received wide acclaim in the independent press and has gone on to tremendous commercial success. Tempus' reputation is therefore solely the result of unexpectedly high fidelity from a speaker of its modest origin and cost.

For Tempus' development, Swans agreed to supply premium components for what evolved into a rigorous electro-acoustics design program, executed to the highest European standards. The resulting design's performance was subjected to stringent laboratory confirmation and completely documented in the audio press, including revealing measurements not typical for any loudspeaker much less one of this type. Tempus has since gone on to claim top performance awards from acclaimed KLANG+TON magazine of Germany, during which time Swans cancelled all advertisements and further commentary.

Tempus is now one of Europe's most favored loudspeakers, leaving listeners admiring its naturally musical sound and excellent value. The design is already a classic, destined to sweep the audio world in a fashion similar to its current success.

The new Swans Tempus set includes:

- 2 x 1.1"/28mm Swans TN28 soft-dome tweeters with open-air housing, Neodymium motor, ferrofluid cooling, decompression chamber, integrated grille and German-made silk dome;
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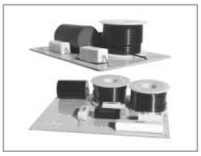
- 2 x completed crossover networks with premium-grade components;
- Two sets heavy input terminals;
- Two pairs internal cable sets;
- Complete manufacturer's instructions.



All Tempus kits are thoroughly pre-tested by Swans and come in matched acoustical sets.

Price:$599 with Tempus parts kit only;
$999 with Tempus kit and cabinets;
$1299 assembled Tempus.

hear different.

"Tempus"

SPECIFICATIONS

Frequency Response:	48Hz-20kHz
Sensitivity (2.83V/m):	86dB
Nominal Impedance:	8 ohms
Power Handling:	10-120W
Dimensions(HxWxD):	406x286x408 mm

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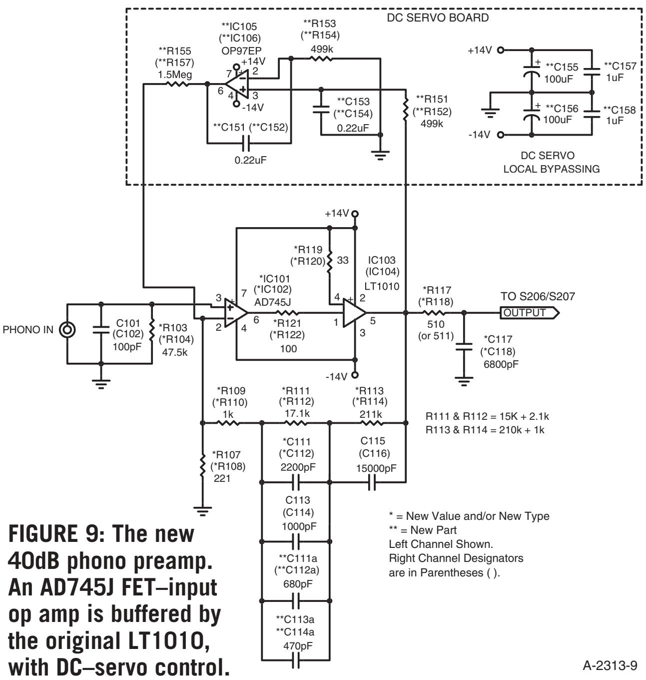

FIGURE 9: The new 40dB phono preamp. An AD745J FET–input op amp is buffered by the original LT1010, with DC–servo control.

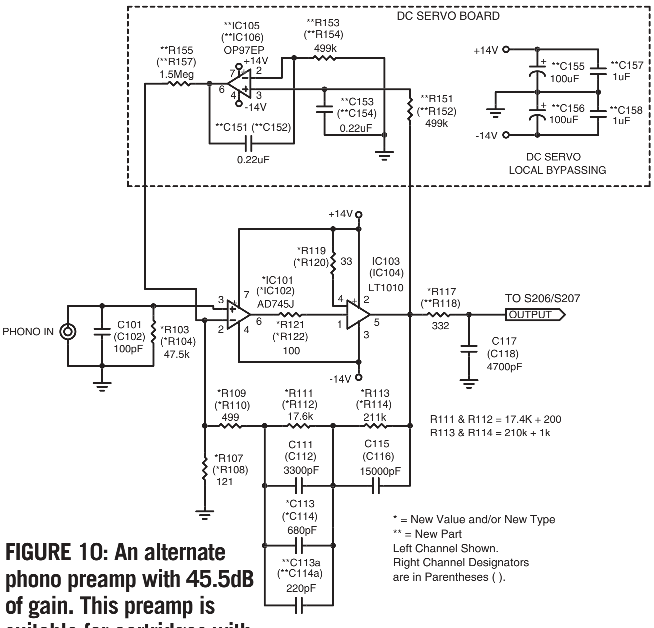

FIGURE 10: An alternate phono preamp with 45.5dB of gain. This preamp is suitable for cartridges with outputs around 2mV.

DC offset with source impedances ranging from very low-Z phono cartridges (below 100Ω) to the 47k cartridge terminating resistor. I believe that it's important to maintain safe DC operating levels even with the cartridge unplugged.

The excellent DC characteristics of the AD745 are possible because of the FET-input design. FET-input op amps are normally too noisy for high-gain circuits such as phono preamps. But the AD745 achieves its excellent noise performance with large-geometry FETs in the front end (the metallization photo shows that the FET differential input takes up well over one-third of the die space). The trade-off, as far as die space is concerned, is a less robust output stage than some other devices. But, buffering the op amp overcomes this limitation.

The AD745 does have higher voltage noise than the LT1028, but my measurements indicate that the new phono preamp is actually quieter than the Adcom original. This is probably due to the substantially lower current noise of the AD745, which results in lower measured phono noise with a low-Z source. You could use a current-feedback amplifier such as the AD811 or the discontinued BUF04 for the output buffer. However, I still like the sonic characteristics of the LT1010 in this application, so I retained it. I lowered the bias resistors to 33Ω, which improved the sonics of the LT1010 even further, but also requires additional heatsinking.

STABILITY ISSUES

Many op amps designed for high-gain applications are not unity-gain stable. The LT1028 is stable at voltage gains of two or higher (it can be used in unity-gain applications, with certain caveats described in the data sheet). Although RIAA preamplifiers normally have very high voltage gains at lower frequencies, at high frequencies the gain will eventually fall to unity and below as the reactances of the RIAA feedback capacitors approach zero.

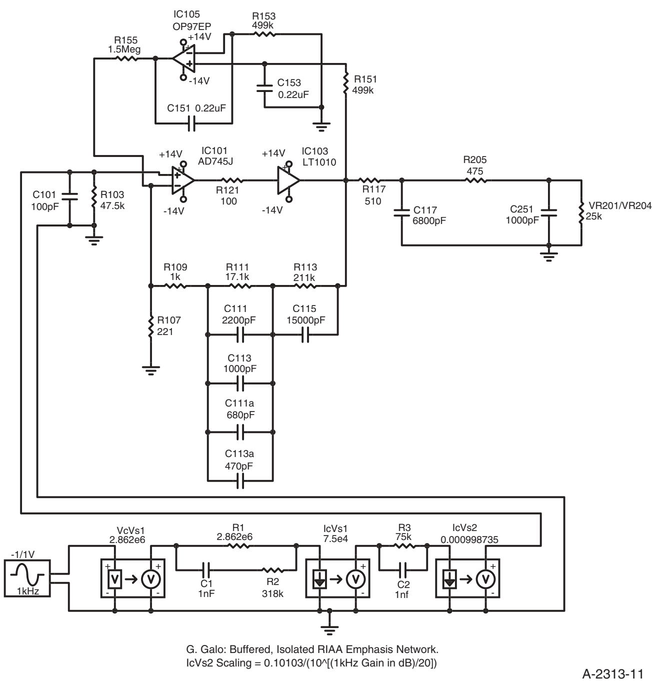

FIGURE 11: Simulation circuit for RIAA analysis of the phono preamps. The circuit on the bottom is a mathematically ideal RIAA emphasis model. Final values for R117 and C117 were determined in conjunction with the line stage loading.

A "gain-stop" resistor—R109 in the Adcom circuit—is added to prevent the voltage gain from falling below safe levels. You can determine the minimum high-frequency voltage gain in the Adcom circuit with the same formula used to calculate the voltage gain of any non-inverting amplifier:

$$\text{Gain} = R109/R107 + 1$$

The values used in the Adcom circuit prevent the voltage gain from falling below 2.9.

The AD745J is designed to operate at voltage gains of five or higher, so the values of R109 and R107 will not work. You could drop R107 to 121Ω, satisfying the AD745J's stability requirement, and raising the 1kHz gain of the circuit from the original 40dB to 45dB. This would be fine for lower-output cartridges (2 to 2.5mV), but moving magnet cartridges with outputs in the 5mV range would likely overload the preamp. Early on in my work with the AD745J in this circuit, I did drop R107 to 121Ω, which worked fine with my Grado Signature XTZII and Adcom XC-MRII cartridges. But, many builders would not find the 45dB preamp suitable.

I'm sure that many readers of *aX* are familiar with Stanley Lipshitz's 1979 AES paper "On RIAA Equalization Networks" (required reading for anyone designing RIAA phono preamps[1]). As the Lipshitz math shows, you can't add a gain-stop resistor—or change the value—without seriously affecting the RIAA accuracy. An RIAA network must be designed with this resistor taken into account, and any change in this resistor requires a redesign of at least part of the network.

The Lipshitz math is quite daunting, but you don't have to be a mathematician to make it work. A few years ago I designed a group of spreadsheets, using Microsoft Excel, to do the calculations for all four feedback RIAA topologies described in the Lipshitz article. With the spreadsheets to do the work, accurate RIAA design has become easy and painless. I use the spreadsheets along with SPICE-based circuit simulation in CircuitMaker 2000, which is my primary schematic capture and simulation program[2].

NEW PHONO PREAMPS

With the Lipshitz math in hand, I decided to redesign the Adcom RIAA circuits, and offer preamps with both 40dB and 45.5dB of gain. The RIAA feedback values are close to the original values, since I retained the large 15,000pF capacitor C115 (C116 in the right channel; for the rest of this article, parts designators in parentheses refer to the right channel).

Figure 9 shows the 40dB circuit and *Fig. 10* shows the 45.5dB circuit. In both preamps R109 ensures that the gain will never drop below five at high frequencies. It was not possible to get exact RIAA values from single, off-the-shelf parts. R111 and R113 are each made from two resistors in series. Four parallel capacitors are needed for the 40dB preamp; this number drops to three for the 45.5dB preamp. The Adcom PC layout accommodates only two capacitors in these locations, but you can easily solder the additional parallel caps to the bottom of the PC board.

The servo amplifier is an Analog Devices OP97EP, a device Walt Jung suggested for this application. The integrator R/C values produce a –3dB point of 0.22Hz for the preamps, and should keep the DC offset at low levels regardless of source impedance (I thank Walt for suggesting the integrator resistor values).

SIMULATIONS

Figure 11 shows my schematic for simulating the accuracy of the RIAA preamplifiers—the 40dB preamp is shown here. The circuit at the bottom is a SPICE model for a mathematically ideal RIAA emphasis network, with the bass and treble portions of the network isolated to prevent interaction. I based this network on an RIAA de-emphasis network Walt Jung designed(see sidebar).

At the time, he was doing RIAA simulation by comparing the outputs of a phono preamp design and a mathematically ideal de-emphasis network. But, CircuitMaker 2000 doesn't seem to support this (though CircuitMaker Version 6 did), so I essentially inverted his model to produce the emphasis model in *Fig. 11.* There seems to be a problem with Analog Devices' SPICE model for the OP97 op amp. I used the model for the OP297, which is the dual version of the OP97.

R1/R2/C1 set the mid- and low-frequency time constants at 3180µS and 318µS. R3/C2 sets the treble time constant at 75µS. With isolated networks, you don't need the Lipshitz math, since there are no interactions between the treble and bass networks—just plug in the real-time constants to determine the component values. VcVs1 and IcVs1 are scaled to match the series resistances that follow each device.

The IcVs2 scaling adjusts the output of the emphasis model to match the gain of the preamp. This applies 10mV to the input of the phono preamp, and makes the frequency response simulation curve lie around 0dB on the "Y" axis. Use this formula to determine the ICVs2 scaling:

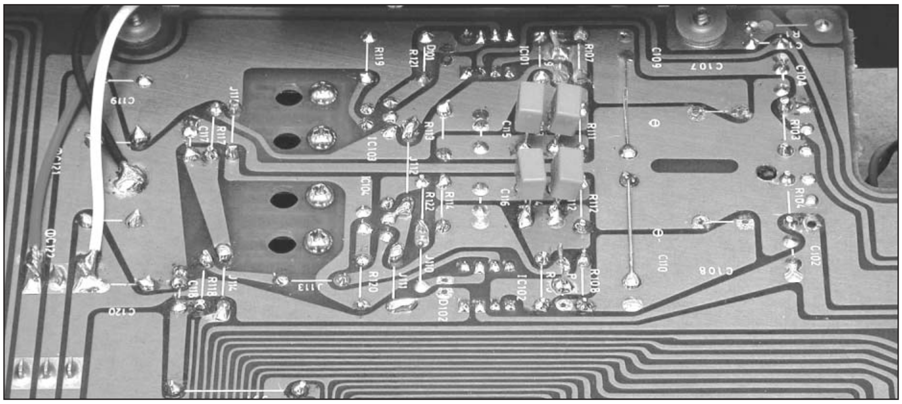

PHOTO 33: Bottom view of the phono preamp showing power supply connections to the DC servo, and parallel capacitors for the RIAA networks. C109 and C110 are replaced with jumpers.

$$\text{IcVs1 Scaling} = 0.10103/10^{[(\text{1kHz gain in dB})/20]}$$

The exact 1kHz gain of the 40dB preamp is 40.3, but this drops to 40.1dB with the line stage loading, so the required scaling is 0.000998735. For the 45.5dB preamps, the actual gain with line stage loading is 45.375dB, for a scaling of 0.000544127.

Figures 12 and *13* show the simulated RIAA accuracy for the 40dB and 45.5dB preamps. Cursors 1 and 2 are set to the lowest and highest points in the curve between 20Hz and 20kHz. The horizontal dashed lines mark the total spread of the error, which is around 0.012dB for both preamps. These circuits are accurate to within ±0.006dB (yes, that's six thousandths of a dB!), 20Hz to 20kHz.

Have I gotten carried away? I don't think so. The curves in *Figs. 12* and *13*

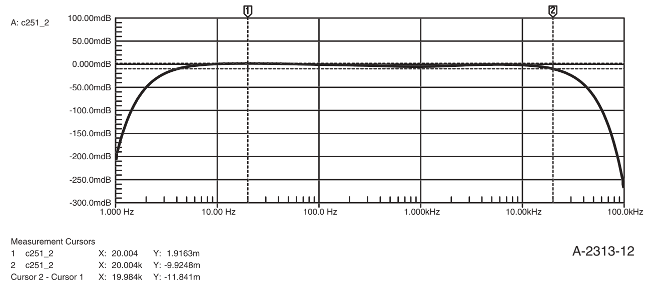

FIGURE 12: Simulated response of the 40dB phono preamp. The simulation shows the response to be accurate ±0.006dB, 20Hz to 20kHz. The tight simulated response ensures good results with real–world component tolerances.

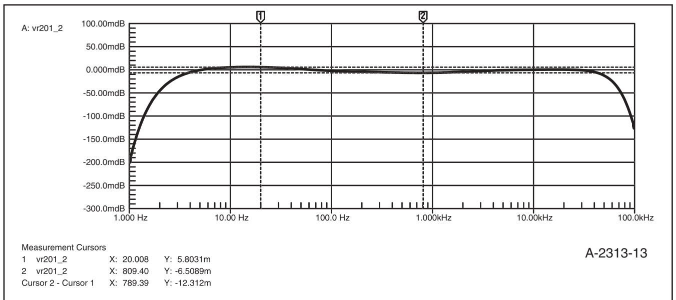

FIGURE 13: Simulated response of the 45.5dB phono preamp. The simulation shows the response to be accurate ±0.006dB, 20Hz to 20kHz.

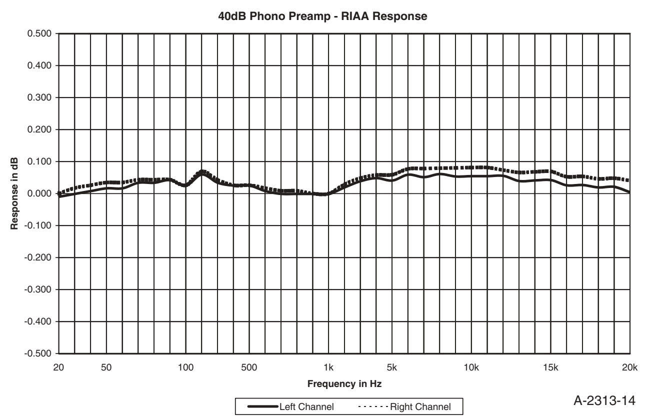

FIGURE 14: Measured response of the 40dB phono preamp. The graph was produced with a spreadsheet, taking errors in the generator, voltmeter, and inverse RIAA network into account. The response is accurate ±0.04dB, 20Hz to 20kHz.

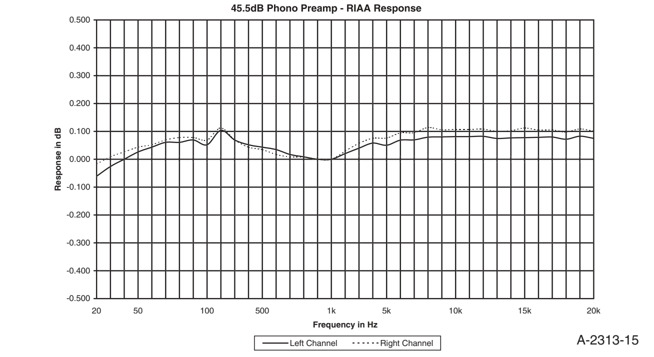

FIGURE 15: Measured response of the 45.5dB phono preamp. This preamp is accurate ±0.08dB, 20Hz to 20kHz.

can only be realized with zero percent tolerance components, which is impossible in the real world. The best that you can do with real-world components is 1% resistors and the 2.5% Wima FKP-2 capacitors sold by Welborne Labs.

My designs retain some of the 1% capacitors (C115) from the original Adcom circuit, which helps. But unless you're a manufacturer, 1% tolerances for the remaining caps will be prohibitively expensive, and generally unavailable in quantities less than 1000 per value. The Lipshitz math, with the help of my spreadsheet, makes it possible to design extremely accurate RIAA networks quickly and easily—it is really no more time-consuming to design a good one than a bad one. So why not do it right?

The Wima FKP-2 polypropylene capacitors are of the same type and construction as the Roederstein KP-series Adcom used, so there's no problem using them with some of the original Adcom capacitors. I hand-selected the 2.5% capacitors I bought from Welborne on my BK Precision 875B LCR meter, which has 1% accuracy. I found that out of six or eight 3.5% Wima caps, at least two measured close to dead-on. The Wima caps are fairly inexpensive, which makes this a practical approach.

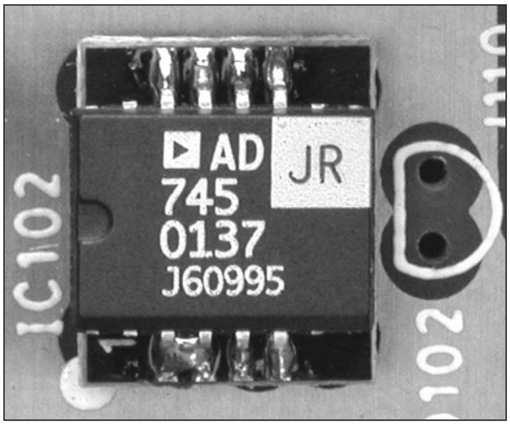

PHOTO 34: Close–up of the AD745JR in the SOIC package mounted on an Aries SOIC–to–DIP adapter. The four pins on each end of the op amp are cut off, and a solder bridge between pins 3 and 4 makes the connection to the inverting input.

The extremely tight accuracy in the simulation will ensure that the end result will still be very good with real-world components. Ideally, I'd like to see a finished product with an accuracy of ±0.1%. Since the late 1970s, there seems to be general agreement in both the "golden ear" and "scientific" camps that this level of accuracy will ensure that the frequency response errors of the RIAA circuit will not be audible. If you start out with accuracy of ±0.1% in a simulation, the end result will probably be quite unacceptable.

My simulations required three deviations from the ideal RIAA network values, as determined by the Lipshitz math. First, the capacitors in the treble portion of the network (C111, C113, and so on) were trimmed to compensate for the loop-gain error of the AD745J. Second, R113 needed to be adjusted slightly to compensate for the impedance of the DC servo.

I made these adjustments with circuit simulation after determining the nominal values using the Lipshitz math. Finally, I decided to treat the line stage loading as part of the circuit, including the bandwidth-limiting network R205/C251, along with the 25k load of the balance and volume controls. The output R/C network R117 and C117 was trimmed with circuit simulation to produce the flattest high-frequency response with the line stage loading as part of the system, with the output taken at the junction of R205 and C251.

This approach will produce optimum high-frequency accuracy only with the phono preamp connected to the line stage with the listening selector switch. This may be of concern to those who tape LP records. If you put the recording selector on phono and the listening selector in some other position, there will be a high-frequency error in the RIAA response at the tape outputs. But, if recording LPs while monitoring your recorder is a necessary part of your audio life, you can change R117 to 590Ω in the 40dB preamp, and 432Ω in the 45.5dB preamp.

BUILDING THE RIAA PREAMPS

Before you order parts, you'll need to make the same choice for the resistors that you did for the line stage in part 3. My recommendations are the same as before. Some of the photos show a mix of Holco H4 and the Vishay-Dale CMF RN-60 types. After I had switched from capacitor-coupled outputs to DC servo control, I made the necessary changes in the circuits only to find that the Holco values I needed were no longer available.

Ideally, I recommend using the same type of resistor throughout, with a couple of exceptions. The 33Ω bias resistors need not be exotic—the Vishay-Dale resistors are fine here. There's also no need to use anything more expensive than the Vishay-Dale or Holco resistors in the servo.

For the remaining resistors, I built one preamp with Caddock MK132s and two others with Holcos and Vishay-Dales. The Caddock resistors are really stellar in these circuits, but the other resistors still perform extremely well. If you use Holco resistors, you'll need to stand them on end, as you did for part 3. Use sleeving on the exposed leads.

Analog Devices has discontinued the 8-pin DIP AD745JN—they now offer this chip only in a 16-pin SOIC package, which they call the AD745JR-16. You can use the AD745JR-16 with the same Aries SOIC to-DIP adapter used in the regulator in part 2 (the 16-pin SOIC package is too wide to fit the Accutek Microcircuit adapter). But, you may still be able to buy the AD745JN.

Rochester Electronics specializes in discontinued semiconductors from most of the major manufacturers. As of this writing, they still had 13,800 AD745JN op amps in stock. With a $50 minimum order, you may consider many other devices of interest, including BUF04s and BUF03s.

COMMON STEPS

Begin construction with steps that are common to both the 40dB and 45.5dB preamps (*Photos 32* and *33*). For a few of the resistor replacements in these preamps, you'll use the same value, but substitute a new type.

- Remove D101 (D102; these parts won't be replaced).
- Remove R115 (R116; these parts won't be replaced).
- Replace R103 (R104) with 47.5k.
- Replace R121 (R122) with 100Ω.
- Replace R119 (R120) with 33Ω Vishay-Dale CMF type RN60.
- Remove C107 (C108; these parts won't be replaced).
- Remove C109 (C110; these parts won't be replaced).
- Install jumpers in the C109 and C110 footprints on the bottom of the PC board (*Photo 33*).

  Feel free to change C101 (C102) to suit the loading requirements of your phono cartridge, taking the capacitance of your tonearm cable into account. Use Wima FKP-2 capacitors from Welborne Labs.

Perform the following step if you are using the 8-pin DIP AD745JN:

- Replace IC101 (IC102) with the AD745JN.

Perform the following steps if you are using the 16-pin SOIC AD745JR-16 (*Photo 34*):

- Cut off the four pins on each end of the AD745JR-16 op amps. These are pins 1, 2, 7, 8, 9, 10, 15, and 16. The

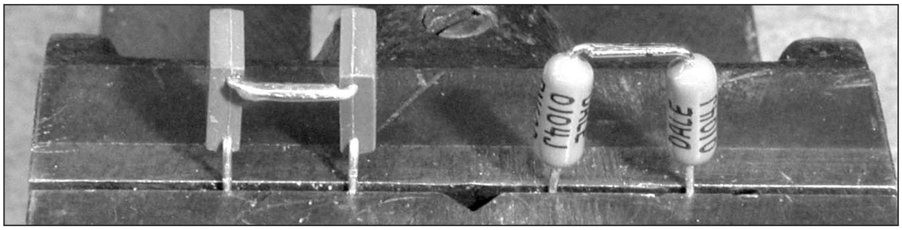

PHOTO 35: Method for soldering resistors in series. Caddock MK132 are shown on the left, and Vishay–Dale on the right. Series resistors are necessary to get the exact values needed for the RIAA feedback network.

TABLE 1
MEASUREMENTS ON 40dB SAMPLE BUILT FOR C. VICTOR CAMPOS

S.N.AC14755
LEFT PHONO TO TAPE OUT LEFT
THD (W/JUNG-LIPSHITZ INVERSE RIAA NETWORK; 2V OUT)

	WIDEBAND	W/80KHZ LP FILTER
20Hz	0.0022%	0.0016%
1kHz	0.0023%	0.0017%
10kHz	0.0033%	0.0029%
20kHz	0.0047%	0.0046%

RIGHT PHONO TO TAPE OUT RIGHT
THD (W/JUNG-LIPSHITZ INVERSE RIAA NETWORK; 2V OUT)

	WIDEBAND	W/80KHZ LP FILTER
20Hz	0.0022%	0.0016%
1kHz	0.0023%	0.0017%
10kHz	0.0032%	0.0029%
20kHz	0.0047%	0.0046%

PHONO TO TAPE OUT (2V IN > 2V OUT)	SMPTE (4:1)	1:1
Left	0.00175%	0.0016%
Right	0.0015%	0.0012%

Phono signal-to-noise ratio (relative to 2V out @ 1kHz)
–94.5dB *Unweighted* (left and right channels identical)
All measurements made with Sound Technology 1700B
By Gary Galo, 11/2/2002

eight remaining pins are 3, 4, 5, 6, 11, 12, 13, and 14 (these are the middle four pins on each side of the op amp).

- Solder the AD745JR-16 op amps to a pair of Aries 8-pin SOIC to DIP adapters.

  For some strange reason, Analog Devices made pin 3 the inverting input on the AD745JR-16. Logically, they should have made this input pin 4, so the eight pins in the middle of the package would exactly match the functions of their counterparts in the 8-pin DIP package. Pin 4 is unused, so this quirk is easy to fix (*Photo 34*).
- Make a solder bridge between pins 3 and 4 of the AD745JR-16 op amp on the Aries header. You'd be surprised how difficult it is to make a solder bridge when that's what you're trying to do! Now, pin 2 on the PC board footprint will connect to the inverting input of the op amp.
- Install the AD745JR-16 modules in the IC101 and IC102 footprints. Carefully observe orientation.

HEATSINK REPLACEMENT

- Remove the LT1010 buffers, IC103 and IC104. Unsolder all five leads and remove the screws that hold the heatsinks in place.
- Replace the 1″ heatsinks with the 1½″ Wakefield or Aavid types recommended in the parts list. These are the same type used for the pre-regulators in part 2. Use thermal compound or the Adcom sili-pads, and mount the LT1010s to the heatsinks with 6-32 hardware, without insulators. As with the pre-regulators, there's no need for insulating hardware, since the heatsinks do not make electrical contact with anything.
- Replace the 22AWG jumpers J111, J112, and J115 with 18AWG jumpers. This keeps the impedance of the power supply lines as low as possible. You may need to enlarge the PC holes with a #50 drill.

Now follow the instructions for the preamp you are building, 40dB or 45.5dB. In two of the steps for each preamp, you will connect two resistors in series to make the correct value. I recommend standing the two resistors on end in a vise, bending the top leads at right angles, and soldering the top leads together. Adjust the length of the top leads and the spacing between the resistors so the series assembly will drop into the Adcom PC footprint.

Photo 35 shows how this is done for both Caddock and Vishay-Dale resistors. Refer to *Fig. 9* for the 40dB, *Fig. 10* for the 45.5dB preamp, and *Photos 32* and *33* for both.

40dB PREAMP ASSEMBLY

- Replace R107 (R108) with 221Ω.
- Replace R109 (R110) with 1k.
- Replace R111 (R112) with 15k + 2.1k in series (R_T = 17.1k).
- Replace R113 (R114) with 210k + 1k in series (R_T = 211k). If you are using Caddock resistors, the series assemblies will straddle C115 and C116 (*Photo 41*).
- Replace R117 (R118) with 510Ω (Caddock) or 511Ω (Holco or Vishay).
- Replace C111 (C112) with 2200pF.
- Solder C111a (C112a)—680pF—to the bottom of the PC board, in parallel with C111 (C112).
- Solder C113a (C114a)—470pF—to the bottom of the PC board, in parallel with C113 (C114).
- Replace C117 (C118) with 6800pF.

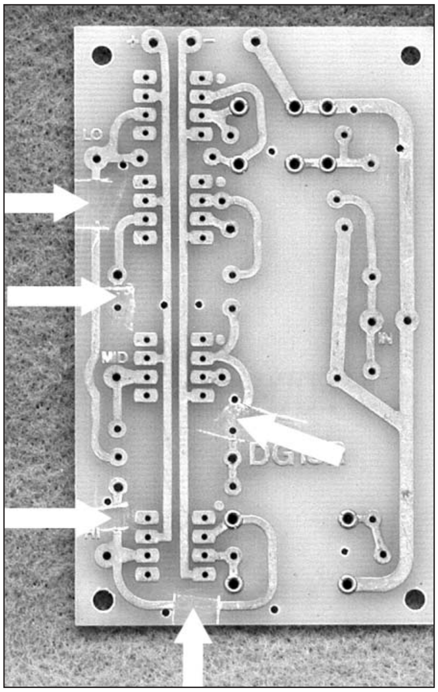

PHOTO 36: The Old Colony DG13R circuit board modified for use as the DC servo board. Five traces are cut, and several new holes are drilled to accommodate the servo circuit.

Note that C113 (C114) and C115 (C116) are unchanged—the original Roederstein KP capacitors are retained. *Photo 32* shows the preamp at this stage in the construction.

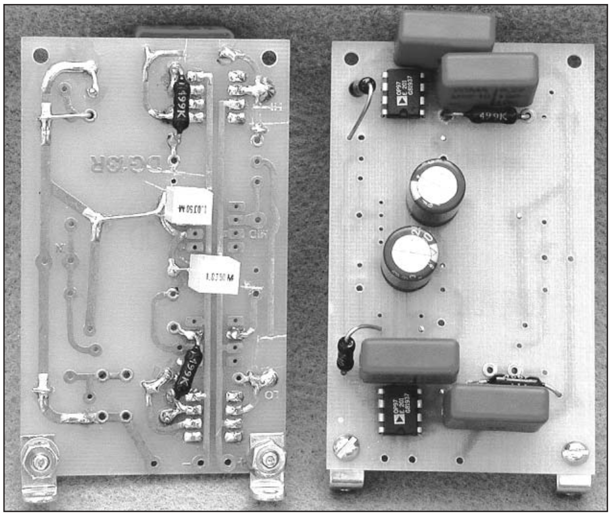

PHOTO 37: Bottom and top views of the DC servo board. R151, R152, C157, and C158 are soldered to the bottom of the board. Only one board is needed for both channels.

At this point, the phono preamp will operate, but with a significant DC offset. You should check the preamp on the bench to verify proper operation up to this point. Apply a 1kHz sine wave to each input at a level of 10mV, and monitor one of the tape output buffers. The level should be 1.05V, since the actual gain of this preamp is 40.1dB.

Check the DC offset at the phono preamp outputs. Maximum output offset for the 40dB preamp will be 1.5V, and most AD745J samples should be less. *Don't listen to the preamp at this point. The DC offset could cause damage to other components, especially your power amp and speakers!*

LISTENING TO GALO'S MODIFIED ADCOM GFP–565

It happens that I have had Adcom's GFP–565 preamplifier as part of my sound system for a number of years. When Gary Galo proposed, some time ago, that he write a modification article for upgrading this equipment, I was pleased and yet somewhat dubious that the product could be improved to any significant extent. I knew that the preamp was one of the Adcom products whose design and manufacture was the responsibility of C. Victor Campos. Mr. Campos does work that is difficult to improve.

I did know that this equipment—like any designed for the consumer market—was necessarily a compromise since only handcrafted, one-of-a-kind devices could be produced as super perfectionist products. Economics and competition are always issues in components designed for consumer markets.

Gary's work, now published in four installments in this magazine, was done over several years since Gary has a demanding day job as well as being involved in other professional audio activities. When he was near completion of his modifications, he asked if I would audition the finished project. Gary asked this knowing full well that I would say exactly what I thought about the result, keeping in mind that this was an audition and that my ears are not all they might have been in years past.

So I unpacked the two cartons containing one of his three modified units, the second box containing his new external power supply. I installed the new unit in the same rack with the rest of my system and listened carefully to three recorded samples, first through the unmodified preamp, then with Gary's modified unit in place and then again through the unmodified unit.

The first sample is from a Telarc SACD, Lorin Maazel and the Cleveland Orchestra performing Igor Stravinsky's "The Rite of Spring." Second, I listened to the New Budapest Quartet playing Bela Bartok's "String Quartet No. 1" (Sz40 Op. 7.) on a Hyperion Dyad CD. My third sample was a Westminster Laboratory Series LP (W LAB 7056) with Sir Adrian Bolt conducting the Philharmonic Promenade Orchestra playing Benjamin Britten's "Young Person's Guide to the Orchestra." The SACD and CD were played on a Sony SACD/DVD player (DVP–NS755V) while the LP was played on a Linn Sondek LP12, updated with a new motor from Origin Live (www.originlive.com) and a Linn Ittok LV II arm with a LINN ASAK cartridge. The preamp drives two Pass Zen stereo amps each driving one of my Thor speakers.

I am pleased to report that there are at least three notable differences in the sound from the two units. First, the individual instruments are more clearly defined and differentiated from each other. This is most noticeable on the Stravinsky, but almost as much on the CD. On the LP the differences were not quite as dramatic.

Second, the soundstage was quite obviously wider on all three of the samples. The response seemed to move outside the two Thors and to fill the space in the room to a greater degree than with the unmodified preamp. The LP is, although very high quality, a monophonic recording. The breadth and detail even on this sample was surprisingly good.

Three, the depth of the orchestral image was much greater. On the Stravinsky all the sound seemed to come from *behind* the speakers and to even go behind the rear wall itself. My room is not ideal for listening, being nearly square—with two large door openings, but the speakers are carefully placed equidistant from the side walls and the back wall. The illusion of the orchestral spread was very firm and stable, and coupled with the increased definition of the individual instruments, the effect was deeply satisfying. The quartet presence was of the four players sitting in, and slightly behind the speakers, and quite vividly reproduced. This preamp has the ability, in my system, to help you forget that the music is being reproduced.

I had only one small objection to the modified result. The separate power supply is housed in a stock aluminum box which has a very slight vibration at what sounds like 120Hz. A slight pressure on the top of the box quiets the noise very effectively. A remedy should be easy.

I believe the modifications Gary has made to his units are a very worthwhile undertaking, even considering the amount of work involved. His modified Adcom GFP 565 is probably the best sounding preamp I have ever had the pleasure of auditioning.—E.T.D.

45.5DB PREAMP ASSEMBLY

- Replace R107 (R108) with 121Ω.
- Replace R109 (R110) with 499Ω.
- Replace R111 (R112) with 17.4k + 200Ω in series (R_T = 17.6k).
- Replace R113 (R114) with 210k + 1k in series (R_T = 211k). If you are using Caddock resistors, the series assemblies will straddle C115 and C116 (*Photo 41*).
- Replace R117 (R118) with 332Ω.
- Replace C113 (C114) with 680pF.
- Solder C113a (C114a)—220pF—to the bottom of the PC board in parallel with C113 (C114).

Note that C111 (C112), C115 (C116), and C117 (C118) are unchanged—the original Roederstein KP capacitors are retained. *Photo 32* shows the preamp at this stage in the construction. At this point, the phono preamp will operate, but with a significant DC offset. You should check the preamp on the bench to verify proper operation up to this point. Apply a 1kHz sine wave to each input at a level of 5mV, and monitor one of the tape output buffers.

The level at the output of the phono preamp should be just under 0.96V. Check the DC offset at the phono preamp outputs. Maximum output offset for the 45.5dB preamp will be 3V, and most AD745J samples should be less. *Don't listen to the preamp at this point. The DC offset could cause damage to other components, especially your power amp and speakers!*

DC SERVO CONSTRUCTION

The DC servo is the same for both preamps. I built the servo on one of Ed

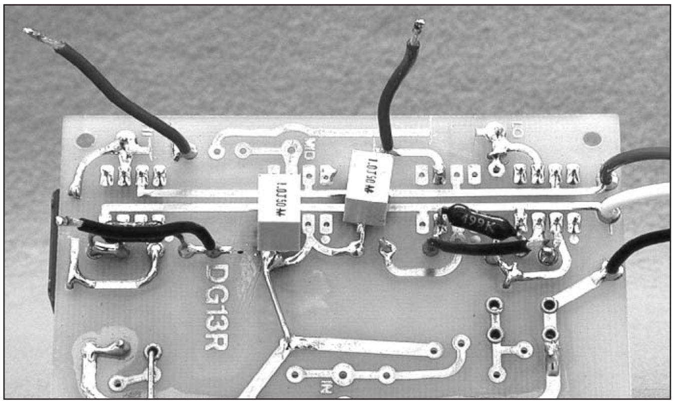

PHOTO 38: Bottom view of the DC servo board with input, output, and power supply leads attached.

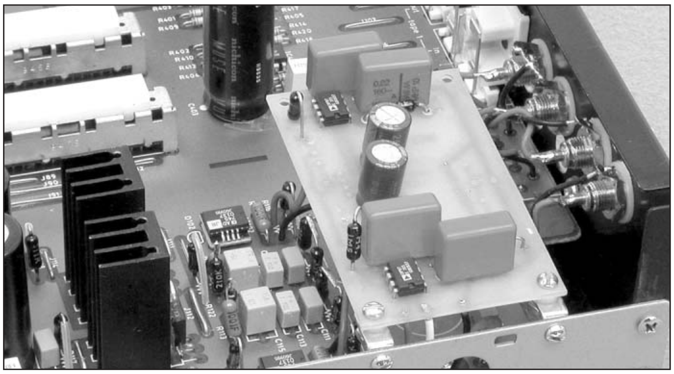

PHOTO 39: Side view of the completed phono preamp with the servo board installed. The servo board is mounted to the preamp's metal side rail with angle brackets.

MEASURING RIAA ACCURACY

If you can afford an Audio Precision System 1 or System 2, its built-in algorithms make RIAA measurement relatively easy. For the rest of us, measuring RIAA accuracy is a tricky and tedious process, requiring the right equipment and a lot of patience. I have developed a procedure using conventional equipment that works very well.

For many years I have used the Jung-Lipshitz inverse RIAA network, which is a passive device I built from the Old Colony kit, with 1%-tolerance parts (no longer available)[3]. But, matching your generator source impedance to this network can be frustrating, and a source of error. Jung and Lipshitz offered two versions of the network, with component values trimmed to accommodate source impedances of 0Ω or 300Ω.

The problem is that most generators have output impedances in the 500 to 600W range. I decided to actively buffer my Jung/Lipshitz network. I ran some computer simulations to check the accuracy of the buffered device against Walt's mathematically ideal RIAA de-emphasis network, and found that the best accuracy was with the 0Ω source impedance network driven directly from IC1's output *(Fig. 16)*. My simulations showed an excellent response across most of the spectrum, with an error of only –0.024dB at 20kHz *(Fig. 17)*.

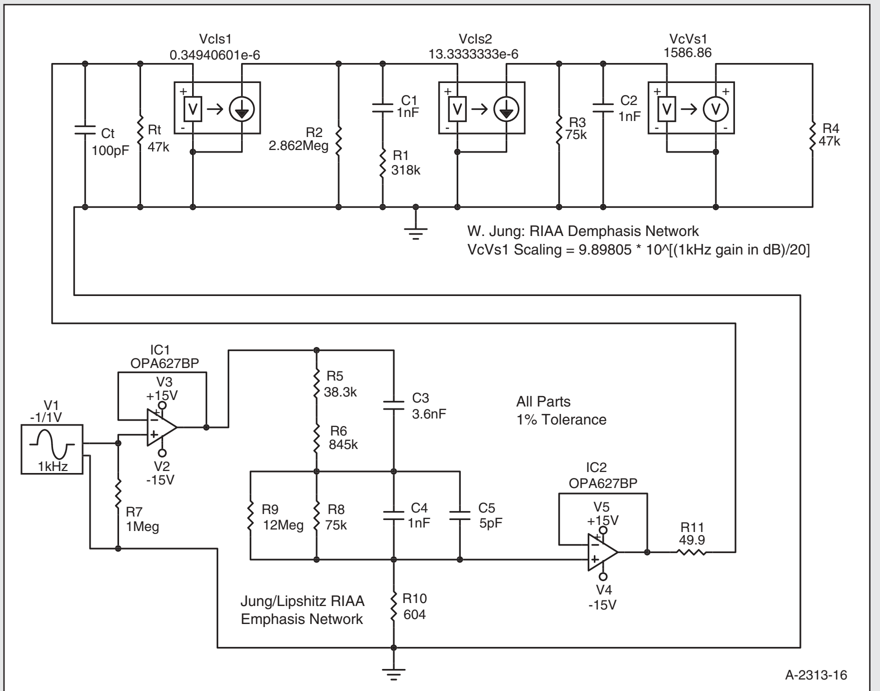

FIGURE 16: Simulation circuit for testing the buffered Jung/Lipshitz inverse RIAA network. The network is checked against a mathematically ideal RIAA de-emphasis model.

The op amps for IC1 and IC2 need to be high-current devices capable of driving a 600Ω load, which is the terminating impedance of the network. The op amp should also be a low-offset, FET-input device, to allow DC coupling. A noisy op amp can cause meter readings to wander, particularly at lower frequencies. I had this problem with Analog Devices AD845, so I changed the op amps to TI/Burr-Brown OPA627BP devices, which can deliver 45mA of output current, with an input voltage noise of 4.5nV√Hz at 10kHz.

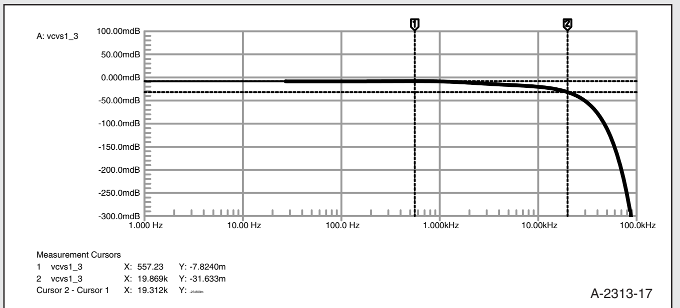

FIGURE 17: Error of the buffered inverse RIAA network, compared to the mathematically ideal de-emphasis model. The network is ruler-flat to 1kHz, and down only 0.024dB at 20kHz.

The difficulty in measuring RIAA response is the need to resolve differences in the hundredths of a decibel. Most signal generators do not maintain exactly the same output at all frequencies, and most meters have variations, as well. Accurate, high-resolution dB meters are rare, and the popular Loftech TS-1 and TS-2 have only 0.1dB resolution, making them useless for RIAA measurements. The best choice is an AC DMM with 20kHz bandwidth and 1mV resolution. I used the generator in my Sound Technology 1700B and a Tenma (MCM Electronics) 72-410A bench-type, true-RMS DMM.

I first tested the accuracy of the generator and DMM as a system, after warming up the test equipment for two hours. I set the generator to 1V output at 1kHz, and fine-tuned the output so the DMM read *exactly* 1.000V. I then ran a decade frequency response test from 20Hz to 20kHz, noting the exact voltage reading at every frequency. Over the course of two days, I re-checked the errors several times to see whether the measurements were repeatable. They were.

I plugged these error figures into a spreadsheet (Microsoft Excel), which allowed me to automatically correct for the generator/meter error. If the generator and meter read 1.007V at 20kHz, as an example, my correction column has –0.007V as the correction needed.

To measure the preamps, I inserted my active Jung/Lipshitz inverse RIAA network between the generator and the phono input on the preamp, set the generator output to 1V, and connected the preamp's main output to the DMM (remember that I treated the phono preamp plus line-stage loading as a system). The inverse network's 40dB insertion loss cuts the level down to 10mV at the preamp's phono input.

The preamp's volume control is carefully adjusted to make the DMM read 1.000V at 1kHz. Then I ran a complete decade frequency response check from 20Hz to 20kHz (in 1kHz steps between 10kHz and 20kHz), writing the results down in the "measured" column in the spreadsheet. I put a formula into the spreadsheet to take preamp measurement and necessary generator/meter correction to give a "corrected" response in the next column.

The spreadsheet converted the voltage measurements to dB, using the formula:

$$dB = 20 \log (E1/E2)$$

where E2 is the 1.000V reference at 1kHz and E1 is the measurement at the other frequencies. This gave the correct ±dB indication in the "response in dB" column. I also entered the errors of the buffered inverse RIAA network above 1kHz, using data from my computer simulation. I did this for both channels, and then used Excel to make a graph with both the left and right curves *(Figs. 14* and *15)*. This procedure is extremely painstaking, but it seems to be accurate and repeatable—GG.

Dell's venerable DG13R electronic crossover boards, sold by Old Colony. I still find this op-amp PC board to be extremely versatile, and I have adapted it for a variety of projects over the years.

You need only one board for both channels. You can also use a small perfboard, if you wish. You can easily adapt the DG13R board for this circuit by cutting a few traces, adding a couple of jumpers, and drilling a few new holes (*Photos 36*, *37*, and *38*).

- Cut the five traces on the DG13R board shown in *Photo 36*.
- *Photo 36* also shows the locations of extra holes that you must drill in the PC board. Use a #55 drill.
- Assemble the servo components on the board, using *Photos 37* and *38* as a guide.
- R151 and R152 are soldered to the bottom side of the PC board. All other servo parts are mounted on the top of the board.
- C155 and C156—the 100µF electrolytic supply bypass capacitors— should be mounted on the top of the board.
- C157 and C158—the 1µF film bypass caps–are soldered to the bottom of the board, in parallel with C155 and C156.
- Solder 2″ lengths of 18AWG hookup wire for the input and output connections for each channel. I used black wire for the inputs and red for the outputs (*Photo 38*).
- Solder three 18AWG hookup wires for the positive and negative supply connections, and ground. Make these wires long enough to reach the bottom of the PC board, where you will solder them to the main preamp supply bus. I used red wire for positive, white for negative, and black for ground (*Photo 38*).

Check the assembly of the servo boards very carefully to make sure that each channel matches the schematics in *Fig. 9* or *10*. Clean the PC board with CaiCleen TRP DG7S-6 Cleaner. Cleaning is extremely important, because residual solder flux can cause leakages in these high impedance circuits.

SERVO BOARD MOUNTING

The DC servo input connections are taken from holes left vacant by the removal of R115 and R116. You must drill two new holes for the output connections:

- Drill a small hole with a #50 drill through the PC board between the vacant R115 holes. Make sure that this hole lands in the middle of the PC trace that goes between the R115 holes on the bottom of the board. Scrape enough lacquer off the PC trace to ensure a good connection.
- Drill a small hole with a #50 drill through the PC board between the vacant R116 holes. Make sure that this hole lands in the middle of the PC trace that goes between the R116 holes on the bottom of the board. Scrape enough lacquer off the PC trace to ensure a good connection.
- Enlarge the R115 and R116 holes closest to C112 and C113 with a #50 drill.

You can mount the DC servo board on the preamp's metal side rail with two small angle brackets, 1/8-nylon spacers and 4-40 hardware (*Photos 39*, *40*, and *41*). The board will sit directly above the old C107 to C110 footprints on the main PC board.

- Mount two angle brackets to the end

PARTS LIST

NOTE: ALL RESISTORS ARE BUILDER'S CHOICE:

Vishay–Dale CMF Type RN60 (Welborne Labs, Mouser 71–RN60D–Value)

Holco H4 (older non–ferrous type, if available, Welborne Labs, Michael Percy Audio, Parts Connexion)

Caddock MK132 (Welborne Labs, Michael Percy Audio, Parts Connexion)

Vishay S–102 (Michael Percy Audio)

DC servo (same for both preamps):

(1)	DG13R circuit board, Old Colony PCBC–4
(2)	Analog Devices OP97EP op amp, Newark 66F5763 or Analog.com
(4)	0.22µF (220nF) 160V Wima MKP–10 capacitors, Welborne Labs WM214 (C151, C152, C153, C154)
(4)	499k ½W resistors (R151, R152, R153, R154; use Vishay–Dale CMF type RN60 or Holco H4)
(2)	1.5M ½W resistors (R156, R157; use Vishay–Dale CMF type RN60 or Holco H4)
(2)	100µF, 25V DC Nichicon KZ, Michael Percy Audio (C155, C156)
(2)	1.0µF, 50V DC Mallory 168–series, Newark 89F1692 (C157, C158)

MISC.

Angle brackets, Mouser 534–616

⅛″ nylon spacers, Digi–Key 875K–ND

4–40 hardware

18 AWG hookup wire: Welborne Labs TWR, TWB and TWW (Welborne Labs);

40dB phono preamp:

(2)	Wakefield 634–15ABP 1½″ heatsink, Newark 91F3674—or— Aavid 581102B00000, Digi–Key HS303–ND (for LT1010 buffers)
(2)	Analog Devices AD745JN op amp, 8–pin DIP, Rochester Electronics—or—Analog Devices AD745JR–16 op amp, 16–pin SOIC, Analog.com (IC101, IC102)
(2)	Aries 8–pin SOIC/DIP adapters, Digi–Key A724–ND (adapters needed only for the AD745JR–16)
(2)	47.5k resistor (R103, R104)
(2)	33Ω ½W resistor (R119, R120; use Vishay–Dale CMF type RN60 or Holco H4)
(2)	100Ω ½W resistor (R121, R122)
(2)	221Ω ½W resistor (R107, R108)
(4)	1k ½W resistor (R109, R110, R113, R114)
(2)	15k ½W resistor (R111, R112)
(2)	2.1k ½W resistor (R111, R112)
(2)	210k ½W resistor (R113, R114)
(2)	510Ω ½W resistor (R117, R118; 510Ω is the Caddock value carried by Welborne; use 511Ω for Holco or Vishay–Dale)
(2)	2200pF 100V 2.5% Wima FKP–2, Welborne Labs WM9 (C111, C112)
(2)	680pF 100V 2.5% Wima FKP–2, Welborne Labs WM6 (C111a, C112a)
(2)	470pF 100V 2.5% Wima FKP–2, Welborne Labs WM5 (C113a, C114a)
(2)	6800pF 63V 2.5% Wima FKP–2, Welborne Labs WM12 (C117, C118)
	45.5dB phono preamp:
(2)	Wakefield 634–15ABP 1½″ heatsink, Newark 91F3674—or—
	Aavid 581102B00000, Digi–Key HS303–ND (for LT1010 buffers)
(2)	Analog Devices AD745JN op amp, 8–pin DIP, Rochester Electronics—or—Analog Devices AD745JR–16 op amp, 16–pin SOIC, Analog.com (IC101, IC102)
(2)	Aries 8–pin SOIC/DIP adapters, Digi–Key A724–ND (adapters needed only for the AD745JR–16)
(2)	47.5k resistor (R103, R104)
(2)	33Ω ½W resistor (R119, R120; use Vishay–Dale CMF type RN60 or Holco H4)
(2)	100Ω ½W resistor (R121, R122)
(2)	121Ω ½W resistor (R107, R108)
(2)	499Ω ½W resistor (R109, R110)
(2)	17.4k ½W resistor (R111, R112)
(2)	200Ω ½W resistor (R111, R112)
(2)	210k ½W resistor (R113, R114)
(2)	1k ½W resistor (R113, R114)
(2)	332Ω ½W resistor (R117, R118)
(2)	680pF 100V 2.5% Wima FKP–2, Welborne Labs WM6 (C113, C114)
(2)	220pF 100V 2.5% Wima FKP–2, Welborne Labs WM3 (C113a, C114a)

of the servo board with the power supply connections. Use nylon spacers, 4-40 machine screws, lock washers, and nuts. The Mouser brackets recommended in the parts list have one hole tapped for a 4-40 machine screw. I used the tapped hole to mount the board/bracket assembly to the preamp's metal side rail.

You'll need to drill two holes in the preamp's side rail to mount the board/bracket assembly.

- Position the board as shown in *Photo 39*, and mark the locations of the two mounting holes.
- Drill the two holes with a ⅛″ drill. I suggest removing the metal side rail before drilling, and re-mounting it after. The side rail is held in place with six screws, and is easy to remove.
- Put the metal side rail back in place, but don't mount the servo board assembly just yet.

Trim the servo input and output leads when you connect them to the main PC board, to keep them as short as possible (*Photos 40* and *41*).

- Solder the left servo input lead to the hole previously occupied by the R115 resistor lead, closest to C111.
- Solder the left servo output lead to the newly drilled hole next to the input lead.
- Solder the right servo input lead to the hole previously occupied by the R116 resistor lead, closest to C112.
- Solder the right servo output lead to the newly drilled hole next to the input lead.
- Route the three power-supply wires to the bottom of the main PC board, between the main PC board and the side rail.
- Fasten the servo board assembly to the side rail with two 4-40 machine screws, using the tapped holes in the angle brackets. There should be plenty of clearance between the main PC board and the components on the bottom of the servo board.
- Solder the positive and negative supply wires, and the ground wire, to the main supply buses on the bottom of the PC board (*Photo 33*).

The phono preamp and DC servo modifications are now complete. Carefully re-check all of the assembly to make sure that your preamp—including the DC servo and its connections to the main PC board—matches the schematic diagrams. Clean the main PC board with CaiCleen TRP DG7S-6 Cleaner.

TESTING

Now it's time for some tests. Move the listening selector to Video/Aux (just to get it out of the way, for now). Put shorting plugs in both of the phono inputs and check the DC offsets with a digital multimeter (DMM) at the outputs of each channel. The easiest place to read the left channel is at jumper J114; measure the right channel at the junction of R118 and C118. The DC offsets in the three preamps I built are very close to 0V. The noise from the phono preamps will make a DMM wander a few tenths of a mV either side of zero on the 200mV scale (or whatever your meter's most sensitive DC scale happens to be).

The 45dB preamp makes my DMM wander a little more than a mV either side of zero, since the noise level is higher in the preamp. But, you can also look at the offset on a scope set to 5mV or 1mV per division. With your probe grounded, carefully set the trace in the center of the screen, and then look at the offset with your probe.

On my preamps, I see random noise above and below the zero line, but no real shift in the DC level. The offset readings should be close to the same regardless of the source impedance. In other words, there should be little difference in the readings with shorting plugs or with the phono inputs wide open (when you change the source Z, the servo does take a few seconds to settle).

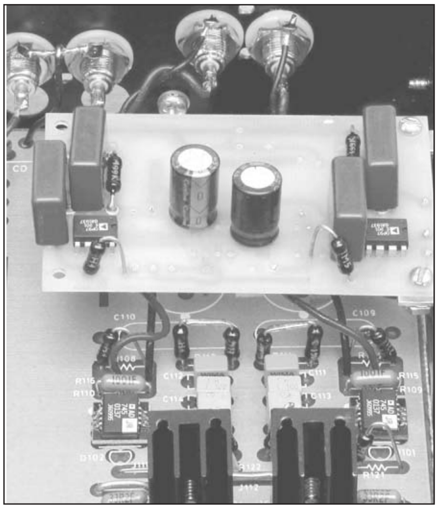

PHOTO 40: Close–up of the phono preamp showing the input and output connections to the servo board. This preamp was built with a combination of Holco and Vishay–Dale resistors.

After you've put the covers back on the preamp, you can still check the phono DC offset at the tape outputs. Put the recording selector in the CD position. Check the DC offsets at either of the left and right tape outputs, and note the readings. If you used OPA2604 op amps for the tape output buffers, the output offsets will be no more than 5mV.

Now move the recording selector to the phono position to read the combined tape buffer and phono preamp offset. The offset at the tape outputs will wander above and below the readings you obtained with the recording selector in the CD position, but about the same amount they did when you checked the offset directly from the phono preamp outputs. If your tests match these, the DC servo is operating properly. I highly recommend running the preamp on the bench for 24 hours and re-checking the DC offset.

If you have a distortion analyzer, you can compare the THD and IMD of your preamp to the measurements in *Table 1*, which shows the measurements I made on the preamp I modified for Victor Campos, which has 40dB of gain. I used a Jung/Lipshitz passive inverse RIAA network, and adjusted the generator output at 1kHz for 2V at the tape outputs. My noise measurements were made relative to 2V at the tape outputs. To measure the residual noise, I removed the signal generator from the phono input and inserted 100Ω resistor plugs in the phono inputs, simulating a low-impedance cartridge.

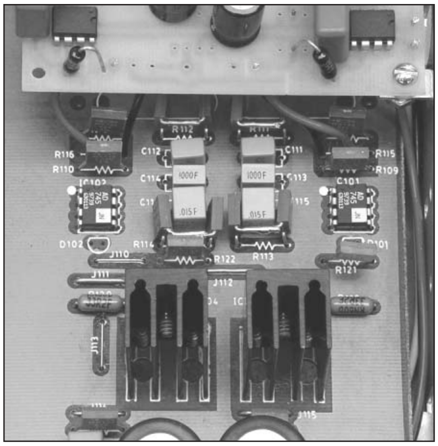

PHOTO 41: Close–up of a phono preamp built with Caddock MK132 resistors. The series resistors that make up R113 and R114 straddle C115 and C116.

My measurements show noise from the 40dB preamp to be 94.5dB below 2V out, *unweighted*. Adcom specifies the noise for the original preamp at 95dB below 2V, but A-weighted. The modified preamp has lower noise than the original.

I discuss my measurement procedure for RIAA accuracy in the sidebar that accompanies this article. The results are shown in *Figs. 14* and *15*. *Figure 14* is the response of the 40dB preamp. The 20Hz to 20kHz RIAA accuracy is ±0.036dB for the left channel and ±0.04dB for the right. *Figure 15* is the response of the 45.5dB preamp. The 20Hz to 20kHz RIAA accuracy is ±0.0825dB for the left channel and ±0.0655dB for the right.

Both preamps are well within the target of ±0.1dB, and every bit as good as the Adcom originals as measured by the manufacturer with an Audio Precision System 1 (the 40dB measurements are actually an order of magnitude better than the Adcom originals). With the conclusion of this preamp modification project, there's nothing left for you to do but listen and enjoy the music. ❖

VENDORS:

Old Colony Sound Laboratory
PO Box 876
Peterborough, NH 03458-0876
1-888-924-9465
1-603-924-9464
1-603-924-9467 (FAX)
www.audioXpress.com
custserv@audioXpress.com

Rochester Electronics, Inc.
10 Malcolm Hoyt Drive
Newburyport, MA 01950
978-462-9332
978-462-9512 (FAX)
www.rocelec.com
sales@rocelec.com

(Remaining vendors are listed in Part 1)

REFERENCES

1. Lipshitz, Stanley P. "On RIAA Equalization Networks," *Journal* of the Audio Engineering Society, V. 27, No. 6, June 1979, pp. 458-481.

2. Although I have found CircuitMaker 2000 to be a good program, their technical support leaves a lot to be desired. There are a number of known issues with the program that the manufacturer—Altium, Ltd.—seems to have no interest in fixing (www.circuitmaker.com and www.altium.com), but I have learned how to work around them. The program does offer a great deal for the money, including schematic capture, mixed-signal simulation, and PC layout.

3. Lipshitz, Stanley P., and Walt Jung. "A High Accuracy Inverse RIAA Network," *Audio Amateur*, 1/1980.

Canto Sirena, Pt. 2

Here are the parts and construction details you need to build this phono preamp with split passive RIAA equalization. **By James Lin**

From one viewpoint, an amplifier is a machine that modulates the power supply in accordance with the input signal, so a quiet, stable power supply is important. Active regulation is one way to satisfy this requirement; however, since an active regulator is an amplifier with the output being the power-supply voltage, it can contribute its own noise and sonic signature. Morrison specifically recommends against a regulated supply because "regulated preamps always sound bleached out and barren to me," opting instead for a choke filtered power supply[1].

POWER SUPPLIES

Although many consider tube rectifiers obsolete, they do have some advantages, such as a clean switching behavior, meaning less noise generation. In addition, tube rectifiers with indirect cathode heaters turn on gradually, providing a delayed and gradual ramp-up of plate voltage, which is gentler on circuit tubes. You can design turn-on circuits that do the same thing, but these are more complex than a simple filament supply. Some audiophiles also believe that tube rectifiers "sound better." On the other hand, tube rectifiers do produce heat, require a filament supply, and have a higher impedance than solid-state diodes, which means decreased voltage regulation with varying loads and will eventually wear out. Still, I believe the advantages outweigh the disadvantages.

The damper diode is a subset of tube rectifiers. They were used in the horizontal amplifiers of tube television sets[12], and were designed to have minimal overshoot characteristics, low impedance, and outstanding noise characteristics[13]. Barbour has written that "if you must use a vacuum tube rectifier, the best performance overall probably comes from damper diodes. They are ideal for DIY, since they are cheap and plentiful[14]."

Since damper diodes are single diodes, you must use two for a full rectifier power supply, resulting in a relatively high filament current—a potential disadvantage. These rectifier tubes feed into a choke/capacitor filter. According to the *Radiotron Designer's Handbook*, in order for the rectifier tubes to have continuous current flow, thus limiting peak current, the value of the first choke must be above a critical value $L \geq R_L/(6*\pi*f)$, where R_L is the load resistance presented to the power supply and f is the supply frequency[15].

For this preamp, R_L is approximately 9kΩ, so with a 60Hz AC power line, L must be at least 8H. Following this first filter, there are individual choke/capacitor filters for each channel. The capacitors in the first two sections should be large to decouple the high voltage supply from the power line down to subsonic frequencies. Resistors R1 and R2 drain the supply over about 10 minutes after turn-off. Resistor/capacitor filters located on each preamp circuit board accomplish final smoothing and isolation.

There are two other supplies beside the high voltage supply—the negative supply for the first stage current source and the filament supply. Usually, these use solid-state rectifiers and resistor/capacitor filters; however, choke/capacitor filters can provide superior noise filtering.

For the negative supply, I also used fast recovery diodes for their better noise characteristics. The negative voltage supply has input resistor R3 to decrease current surge at turn-on, and R4 to drain this supply at turn-off. The power-on indicator LED is driven by this supply, and I chose resistor R5,

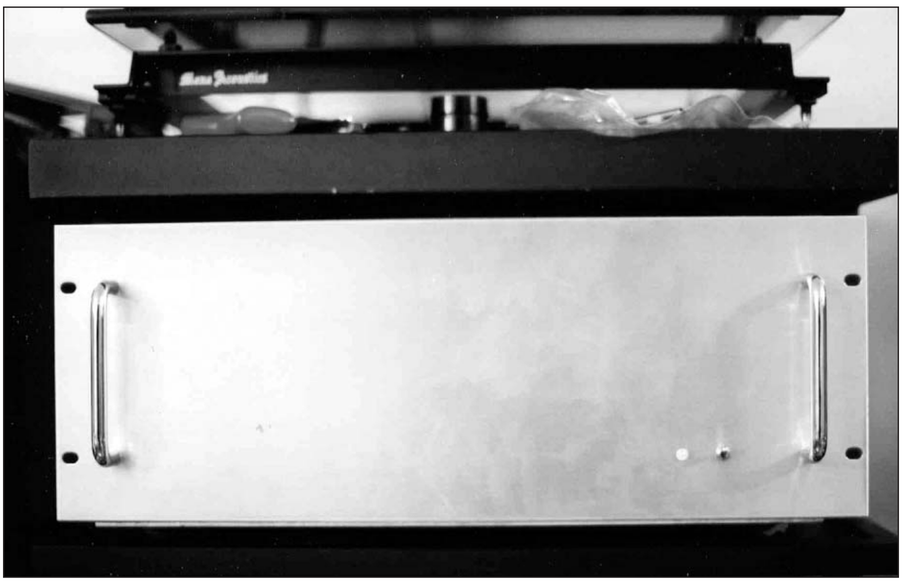

PHOTO 1: Front view of Canto Sirena phono preamp.

which determines the current through the LED, to give a visible indication without being too bright.

The filament supply uses a similar topology, but with an initial capacitor/resistor combination C7/R6 to fine-tune the voltage. Charles King reviewed data that running the filaments above nominal voltage greatly decreased tube life, whereas running it as much as 10% below increased tube life without significantly affecting the sound[16]. I recommend running the filaments between 5.7 and 6.2V to extend tube life while allowing for voltage drops during high power demand periods (e.g., heavy summer air conditioning). One incidental advantage of using chokes in the filament supply is that turn-on occurs over a few seconds, minimizing thermal shock.

It is a very good idea to use transformers that have an electrostatic shield, which significantly reduces interwinding capacitance and bypasses a large amount of AC line noise and grunge to ground. This helps lower the noise floor. You can identify the shield on a transformer schematic as a line between the primary and secondary windings terminating in a ground symbol. Enclosed transformers and chokes also help minimize the amount of electromagnetic interference (EMI) that is splattered around the power supply. These features are readily available in surplus government and industrial transformers and chokes, which you can obtain from a number of surplus dealers at a reasonable cost.

The final power-supply design is shown in *Fig. 3*. The capacitors across the power switch C11 and C12 absorb any spark that may occur when the switch is opened, thus preventing a "pop" in the system at turn-off. With these in place, there are no significant turn-on or turn-off transients. The major disadvantages of this supply are size and weight. The final preamplifier in one box weighed about 45 lb, nearly as much as my power amp, and draws about 65W.

TUBES

There are variants for each of the tubes used, with similar or identical specifications. For the standard 6SL7, there is also the military version (often referred to as JAN, for Joint Army-Navy) VT-229, the ruggedized military 6SL7WGT, and the military-industrial 6SU7/6188, which has closely matched sections. A similar tube is the 5691, which is an industrial version with longer specified life (as long as it is run within somewhat conservative conditions). One caveat is that the 5691 draws twice as much filament current as the 6SL7.

The 6SN7 also has a military version VT-231, as well as a 6SN7W or WGT, again indicating a ruggedized version. In addition, there are also two later versions, the 6SN7GTA and GTB, which have a higher plate voltage and wattage specifications, and the 5692, which is an industrial version. Ironically, even though the 5692 has extra bracing, both Morrison[1] and Barbour[9] report these tubes to be microphonic. Wafer-base 6SN7s have also been reported to be more microphonic.

The 5692 is specified for a 10,000-hour life span; however, this is only if it is run under very conservative conditions. Under those same conditions, a 6SN7 would probably also have an extended life span. The British also had the similar B65, and the military DV181 and CR1988.

The 6J5 is a metal envelope tube that also comes in a G version (ST-shaped glass envelope, like a Coke bottle) and GT version (cylindrical glass envelope). Barbour recommends the glass 6J5G or GT over the metal 6J5s, which are "more 'hazy' sounding[9]." The British version of this tube is the L63.

Tube choice can make a difference in the sound of this preamp. Tube connoisseurs report that new old stock (NOS) tubes generally sound better. (New old stock refers to components that were manufactured years ago—hence "old stock"—but never used.) Not only can different brands of tubes sound different, but tubes of the same brand made in the 40s and 50s are reported to sound different from tubes made in the 70s and 80s, presumably due to differences in construction and materials. Different authors seem to prefer different variants. If you're interested in experimenting, you could try buying some used, tested tubes, and see what you like best.

PARTS

I agree with Barbour that the two most important elements in sound quality are tube types and circuit design[9]. My experience is that parts make a small difference, which nevertheless can be important to some. I followed Morrison's recommendation to use wire-wound resistors whenever possible[1], and metal films elsewhere. Wire-wound resistors are very quiet, stable, and have very low temperature coefficients, which means that their values will change very little with current and voltage variations (i.e., music).

I used Panasonic and Nichicon electrolytics for most of the power supply capacitors. Anyone who thinks this is a sonic compromise can try oil capacitors or Black Gates for considerably more money. I think this is a minor tweak given that the preamp has resistor isolation and filtering with polypropylene capacitor sections located on the circuit board for each tube, but I haven't tried it myself so I could be wrong.

On the preamp circuit board I used Axon metallized polypropylenes for the power-supply caps, Auricaps for coupling caps, and Reliable RT polystyrenes for RIAA network caps. It is especially important to use close tolerance, matched parts for the RIAA network components, both for accurate equalization and close channel matching. I recommend obtaining the 1N5311 current source diode from Handmade Electronics. I tried a lower-cost part from Mouser, and it was significantly noisier.

These parts are not inexpensive;

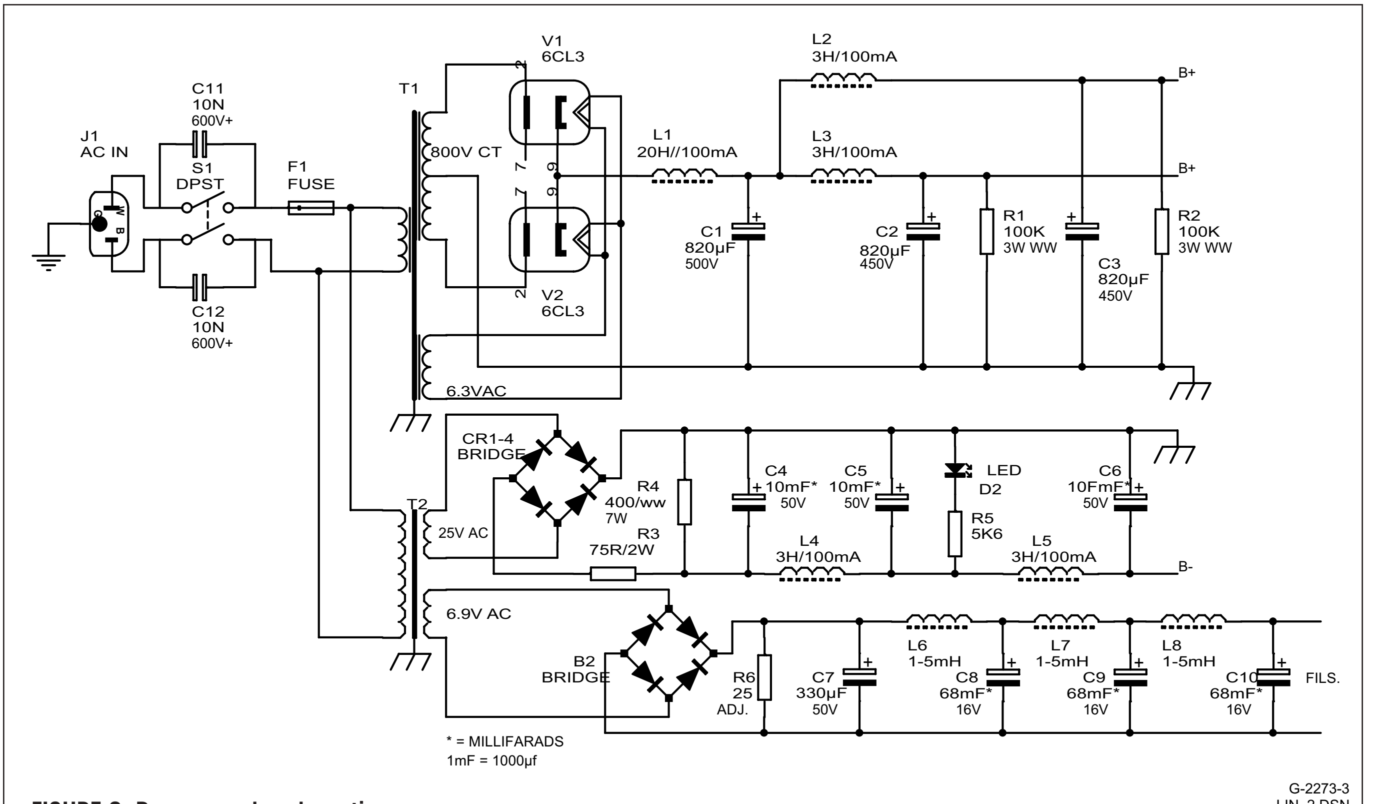

FIGURE 3: Power-supply schematic.

however, there are not a lot of them. I figure if I'm going to spend this much time and effort building something from scratch, hang the expense. It's still a lot cheaper than buying a high-end phono preamp.

For transformers and chokes I scrounged parts from surplus dealers and eBay. For example, I used an Eico HF-50 power amp transformer for the high voltage transformer. Since surplus part supplies vary, feel free to substitute equivalent or better parts. Interestingly, I found the cost was about evenly divided between circuit and power-supply parts.

CONSTRUCTION

Before beginning, it is important to determine whether your phono setup is wired for balanced operation or single-ended. Most cartridges are inherently balanced, with the exception of the Deccas, which have a common negative connection. Therefore, if the tonearm negative connections are separate—and separate from the tonearm ground connection—they should be able to be used in a balanced circuit. In my tonearm the stock wiring has been replaced with balanced wiring and a separate ground wire, but it uses standard phono plugs for compatibility.

The safest way to determine this is with phono cartridge leads disconnected, so there is no chance of zapping your cartridge with the current from your ohmmeter. If the negative leads of your phono cartridge are tied together either in the cartridge itself or in the tonearm wiring, or connected to the ground wire, then you should change the 56kΩ input resistor R2 to 47kΩ, omit the 150kΩ resistors R1 and R6, and connect pin 4 of the 6SL7 directly to ground.

I built the preamp circuit on terminal boards from Welborne Labs, following some rules of thumb. First, keep input and output components separated. Second, keep input and output components at right angles to each other to minimize interaction. Third, leave enough space for ventilation of heat sources—resistors and tubes. Fourth, use component leads for connections whenever possible to minimize the number of solder joints. Fifth, place power-supply caps near each tube. *Photo 2* shows the layout of the circuit board from the top and bottom.

The input is at the end with the smaller polypropylene power-supply cap, and is basically laid out following the circuit diagram. I mounted several components such as the 1N5311 (banded end toward the negative supply), RIAA resistors R6 and R7, coupling capacitor C5, power-supply isolation resistors R11, R14, and R17, and cathode resistor R20 on the underside of the board, and soldered input resistor R2 directly across the input tube socket. I used a bus ground wire running along one edge of the board, connected to the chassis at the turntable ground binding post. The power-supply ground wire also connects to the chassis at that point. I isolated both input and output sockets from the chassis by washers and connected the output grounds to the bus wire ground wire for each channel.

I compromised by putting everything into one Bud "Rak-Mount" chassis box, 7″ high by 17″ by 17″. It has a 0.13″ front rack mount panel, relatively thick 0.09″ aluminum sides and back, and has provision for internal shelves of the same thickness. However, the top and bottom are constructed of relatively thin 0.05″ aluminum.

I needed to figure out how to squeeze everything inside this box without causing hum problems. The flimsy bottom didn't seem ideal for the usual construction technique of mounting everything on the bottom of the box. In addition, the transformers and some of the chokes were designed to be mounted on the top of a chassis, with the terminals protruding through the underside, and thus couldn't be mounted on the bottom. I therefore decided to mount everything on the back or on vertical internal shelves.

I mounted the preamp circuits on the back wall of the box, which allowed short signal wiring at the input and outputs. This also allowed the tubes to be mounted horizontally, which according to David Manley minimizes tube shot noise[17]. I then distributed the power supply on two internal shelves, placing the high voltage power transformer at the front of the box, as far away from the preamp circuits as possible (*Photo 3*). I built the power supply in sections, and tested each section by temporarily hooking each transformer up to AC and using a resistor load—about 9kΩ/25W for the high voltage circuit and 2.5Ω/20W for the filament circuit—to simulate the estimated preamp circuit load to confirm it had been wired correctly.

These resistor loads were made up using series and parallel combinations of inexpensive power resistors. I used bus ground wires for each section, connecting each section at one point and running the power-supply ground to chassis ground at the input. These shelves served to shield the power supply from the signal circuits, as well as stiffening the box.

I fastened the preamp boards to the rear of the box with 1″ standoffs, and rear-mounted and wired the phono sockets to the circuit board before final assembly. The standoffs are loosely attached to the circuit boards, the sockets are fed through and the external nut is threaded on to draw them up tight against the back, and the standoffs are then attached to the back wall with screws and tightened. The assembled preamp is shown in *Photo 4*.

This layout served a number of functional purposes—chassis stiffening, separation of power supply and signal circuits, some shielding, and short signal paths—so I'm pleased with the way it worked out. After building this project, mounting everything on the bottom of a box like everyone else seems, well, boring! I also stiffened the bottom with a 0.1″ sheet of aluminum, which I epoxy-glued to the chassis bottom sheet, and used EAR Isodamp equipment feet for some vibration isolation.

For safety's sake, the AC power cord should be of the three-wire variety with the safety ground wire securely attached to the bare metal chassis. This prevents electrocution in the unlikely event of a major component failure resulting in line or high voltage touching the chassis.

POWERING UP

I recommend that you first turn the unit on using a Variac® to gradually increase the voltage, watching for sparks, smoke, and other signs that something

PARTS LIST

Preamplifier (1 channel)

Capacitors (400V minimum rating)
C1 – 10µF metallized polypropylene
C2 – 3n9 polystyrene 1%
C3, C6 – 39µF metallized polypropylene
C4 – 1nF polystyrene 1%
C5 –100nF
C7 –1.0µF

Resistors
(1/2W unless otherwise noted, w-w = wire-wound)
R1, R6 –150kΩ
R2 – 56kΩ 1 %
R3, R5 – 100kΩ/3W w-w
R4 – 200Ω trimpot
R7, R8 – 330 or 332kΩ 1%
R9 – 30kΩ/3W w-w
R10, R15 – 82kΩ 1%
R11, R13 – 15kΩ/3W w-w
R12 – 10 kΩ/5W w-w
R14, R17 – 3kΩ/3W w-w
R16 – 2200Ω 1%
R18 – 301kΩ 1%
R19 – 25kΩ/3W w-w
R20 – 374Ω w-w
R21 – 250kΩ

Tubes
V1 – 6SL7/6SU7/5691/6188
V2 – 6SN7/5692
V3 – 6J5/GT

Other
1 – 1N5311 3.6mA constant-current diode
3 – octal sockets

Power Supply (both channels)

Transformer(s)
800VCT/100mA
6.3V AC/2.4A
6.3V AC/5A
24V AC/300mA

Capacitors
C1 – 820µF/450V
C2, C3 – 820µF/450V
C4, C5, C6 – 10,000µF/50V
C7 – 3300µF/50V, adjust for filament voltage between 5.7 and 6.2V
C8, C9, C10 – 68,000µF/16V
C11, C12 – 10nF/600V or higher

Chokes
L1 – 20H/300Ω/100mA
L2, L3, L4, L5 –3.0H/200Ω/100mA
L6, L7, L8 – 1-5mH/0.7Ω/1A-11A swinging choke

Resistors
R1, 2 – 100kΩ/3W wire-wound
R3 – 75Ω/2W
R4 – 400Ω/7W wire-wound
R5 – 5600Ω
R6 – 25Ω, 5W, adjust for filament voltage between 5.7 and 6.2V

LEDs
1 – T1 ¾ green LED

Rectifiers
CR1-4 – 1A/100V fast recovery diodes
CR5 –15A/50PIV bridge
V1, V2 – 6CL3/6CK3

Other
2 – Novar sockets
1 – 2.0A slo-blo fuse

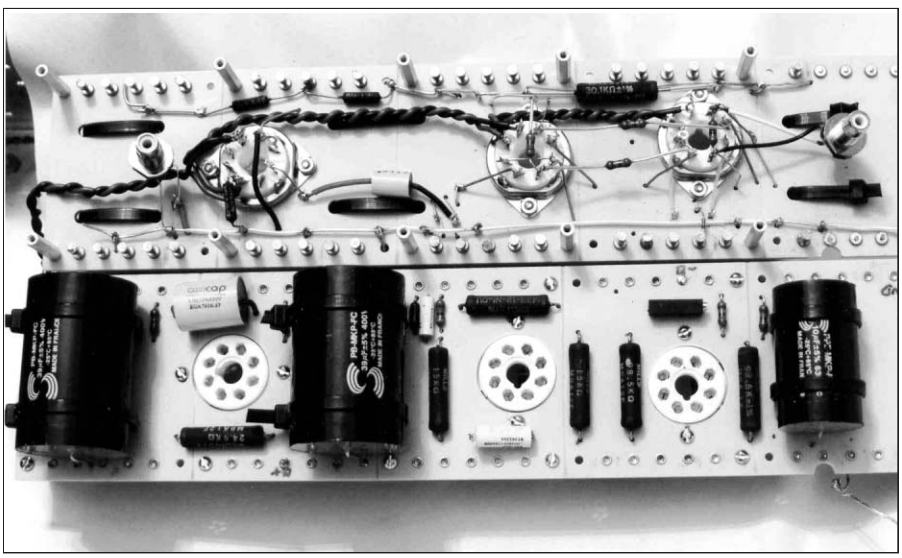

PHOTO 2: Circuit board layout from top and bottom. The small power-supply capacitor is at the input end.

is wrong. If a Variac is not available, at a minimum the fuses should be in place. With a Variac, note that the high voltage will not come on until the Variac is turned up most of the way, since there needs to be enough voltage to turn on the cathode heaters in the tube rectifiers.

Assuming all goes well, check that all the voltages are correct. When doing this, connect the voltmeter ground lead to the chassis ground, make sure you're insulated from any conductive surface, and take the positive test lead in one hand and put the other hand firmly in your pocket. The high voltage power supply has over 100 joules of energy storage, and you definitely don't want to fool around with it. To give you an idea, medical defibrillators use between 25 and 100 joules to convert heart rhythms. I'm not trying to scare you, just telling you to take precautions.

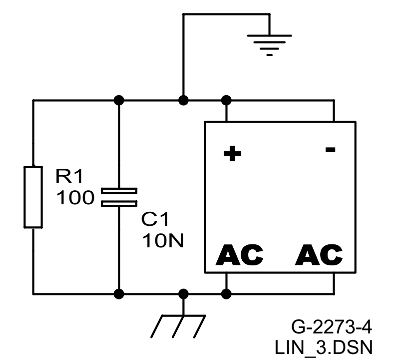

FIGURE 4: Ground lift schematic. Bridge rectifier should be high current, e.g., 25A, any voltage rating. Circuit ground is at top, chassis ground is at bottom.

With a line voltage of 122V, I measured the heater supply at 6.1V, the high voltage supply at 380V, negative supply about 25V, the 6SN7 cathode voltage around 85V, plate voltage around 280V, and the 6J5 plate voltage around 140V. It isn't necessary to measure the 6SL7 plate voltages because this tube is direct-coupled to the 6SN7, so the plate voltage will be a few volts higher than the 6SN7 cathode voltage. Results should be within 10% of this if you've wired the circuitry correctly. You should set the AC balance pot on the cathodes of the 6SL7 tubes at their midpoint initially, and adjust them to obtain the same 6SN7 diff amp plate voltages. Once this is done, the next step is to place the preamp in your system and try it out.

TROUBLESHOOTING

I had a couple problems which were not initially apparent. You may not have the same problems, but I mention them just to illustrate what can go wrong. Initially, there was quite a bit of hum and one channel made a series of random, loud noises, due to a cold solder joint to the ground terminal. It's always a good idea to check each solder joint by pulling and pushing on it a little to make sure it's solid. In this case, I forgot to follow my own advice.

Next, I noticed an intermittent whistle in one channel, and tapping on the outside of the chassis produced a thumping sound out of the loudspeakers, due to a couple of microphonic tubes. You could find the guilty tube or tubes by tapping each tube in turn while the preamp is in your system, but I think it is easier to do this on the test bench with the aid of an oscilloscope (less traumatic on the ears). Connect the oscilloscope with one output channel, then very gently tap each tube in that channel in turn with a plastic pen or a wooden pencil, being careful not to touch anything else. Whacking away at a hot tube could damage it, so be gentle.

Now, all tubes are microphonic to a degree. If you tap them, you will get a signal. But a bad tube will give you a much bigger signal for a longer period of time.

Then, during some tube swapping, I ran across a very noisy tube. I localized this by swapping tubes between channels. The noise moved with the guilty tube.

Finally, I noticed some humming and buzzing. I installed a partial ground lift consisting of a 100Ω resistor in parallel with a 0.01µF capacitor between circuit ground and chassis ground. A further safety improvement suggested by Rod Elliot is to add a high current bridge rectifier in parallel with the ground lift[18] (*Fig. 4*).

With this in place, hum and buzz were inaudible even with the volume full up and my ear next to the speaker. On one record, I heard residual hum that disappeared when I lifted the needle. It was on the recording and not from my system!

The original design has been criticized for low-frequency "instability." Kevin Carter of K & K Audio reported that one sample put out subsonic pulses resulting in "interminable woofer 'breathing,' " which he diagnosed as due to power-line sags and overshoots. I hadn't noticed any problems, but then my subwoofer has an infrasonic filter built in. So I powered the preamp through a Variac, monitored the output via an oscilloscope, and found that twiddling the voltage dial will indeed produce infrasonic output from the preamp, confirming the diagnosis.

To investigate the practical effect of this, I then plugged the preamp directly into the AC line and monitored the out-

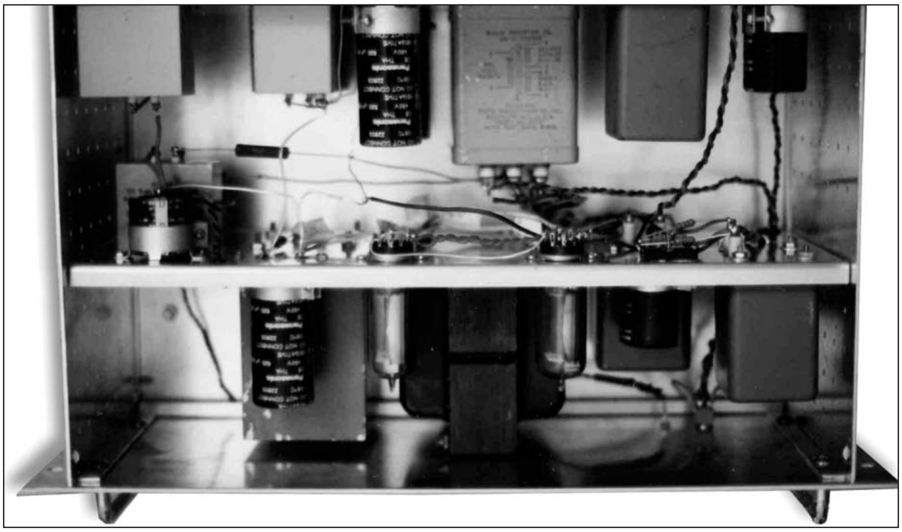

PHOTO 3: Power supply. Front of preamp is at the bottom of photo (author's note: note rack handles at front!).

put for about 20 minutes. With my first power supply, the output voltage showed infrasonic fluctuations of 20–50 peak mV with spikes to 100–200 peak mV every few minutes. Increasing the capacitance in the first two stages of the high-voltage supply to the values listed decreased the output voltage fluctuations to around 5–10mV peak, with occasional spikes to 50mV peak every few minutes.

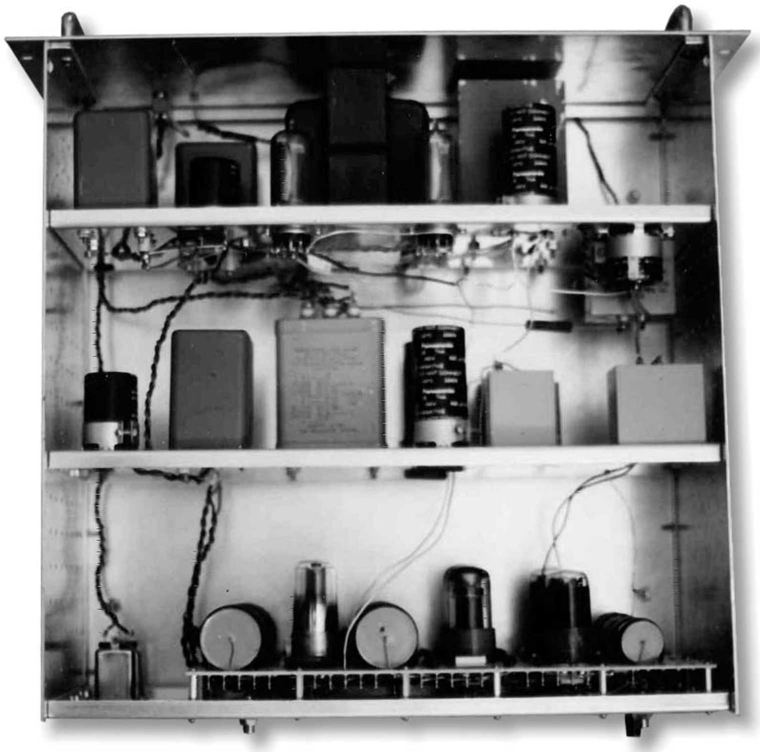

PHOTO 4: Preamp construction. Front of preamp is at top of photo.

To put this into perspective, a "typical" record warp of .003″ at 4Hz[19], which is about the thickness of a human hair, when played by a 1mV/cm/s flat transducer through this preamp would produce an infrasonic output of about 80mV. This is a good reason to limit infrasonic response in an LP-based system[19].

MISCELLANY

Since one of the goals of this design was accurate RIAA equalization, I measured both channels using the Old Colony inverse RIAA kit, which is accurate to ±0.1dB. With the oscillator amplitude set by eyeball using the built-in meter, it was within 1–2%, so the overall measurement is only accurate to within ±0.2dB or so.

Because the RIAA kit is single-ended, I needed to temporarily convert the input from a balanced configuration to single-ended. I found that the output was flat within ±0.1dB between 40Hz and 10kHz for both channels, and about 0.25dB down at 20Hz. This is within the error of my measurement setup, so that I may have been measuring the errors in my instruments as much as the errors in the preamp. These results were obtained without any trimming or tube selection, so they should be typical. They also support Dr. Lipshitz' comment that accurate design is the way to go[5].

Measured input resistance was 47kΩ. Gain at 1kHz was about 49dB, and visible sine wave overload distortion occurred at about 150mV RMS input. Initially, one channel had about 1.3dB more gain than the other. One unavoidable consequence of a no-feedback design is that variations in tube parameters can result in channel imbalances, but with some tube swapping this was reduced to about 0.3dB. Channel imbalances of 0.4–0.5dB are not uncommon in cartridges. For example, in an issue of *Hi-Fi Choice* from the 1980s, about one-third of over 100 cartridges tested had a channel imbalance of more than 0.5dB[20].

Calculated input capacitance is about 130–140pF, due to the Miller effect in a no-feedback design. As already mentioned, calculated output impedance is about 8kΩ, so a low capacitance output cable is a must.

A few words about reliability: tubes can be very reliable—consider how long the average TV picture tube lasts. All the tubes in this design are industrial grade, and run very conservatively. The power supply gives gentle turn-on and low-filament voltages for maximum tube life. Assuming no defective components, this design should run as specified for many years. In fact, Diego Nardi rates the 6SL7 life span as 50–100,000 hours[21], and Eric Barbour has written that preamp tubes can last for 150,000 hours or more if they are not abused[22]! Therefore, I think 5000–10,000 hours of tube life is a conservative estimate. Assuming no defective components, this design should run as specified for many years.

Just a few words about the sound. While the builder's ear is always biased, the sound stage seems bigger, depth cues seem to be clearer, and the overall sound seems smoother than my old preamp. More to the point, inner instrumental lines seem easier to follow as distinct lines—more music! Preamp noise is essentially inaudible from the listening seat at normal levels.

One more thing, the name: Canto Sirena is Italian for Siren Song, because I applied a little Roman rigor to the original design, but I believe the fun is still there. ❖

SOURCES

Angela Instruments
10830 Guilford Road, Suite 309
Annapolis Junction, MD 20701
www.angela.com
tubes, sockets

Antique Electronic Supply
6221 S. Maple Ave.
Tempe, AZ 85823
www.tubesandmore.com
tubes, sockets

Digi-Key Corporation
701 Brooks Ave. South
Thief River Falls, MN 56701-0677
www.digikey.com
GI852 rectifiers, rectifier bridges

Fair Radio Sales
PO Box 1105
1016 E. Eureka St.
Lima, OH 45802
www.fairradio.com
Surplus transformers, chokes, tubes

Handmade Electronics
PO Box 9114
Allentown, PA 18105
www.hndme.com
Current source diode, tube sockets, resistors

Michael Percy Audio
11731 Stillwater Creek Road
Nevada City, CA 95959
www.percyaudio.com
Mills resistors, metal film resistors, matched film capacitors, phono sockets, EAR equipment feet

Mouser Electronics
1000 N Main Street
Mansfield, TX 76063-1511
www.mouser.com
Bud RAK-mount box, electronic parts

Old Colony Sound Laboratory
PO Box 876
Peterborough, NH 03458
888-924-9465
www.audioXpress.com

Parts Connexion
2885 Sherwood Heights Drive, Unit # 72
Oakville, Ontario
Canada L6J 7H1
www.partsconneXion.com
Audio Note and Mills resistors, tube sockets, other DIY parts

Surplus Sales of Nebraska
1502 Jones Street
Omaha, NE 68102
www.surplussales.com
Surplus transformers, chokes, tubes

Welborne Labs
PO Box 260198
Littleton, CO 80126
www.welbornelabs.com
Terminal boards, resistors, capacitors, tubes

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18. Elliot R. "Earthing Your Hi-Fi—Tricks at Techniques," at http://sound.westhost.com/earthing.htm.

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22. Barbour E. "Testing Preamp Tubes for the Real World," *Glass Audio* 3 (2): 16–28, 1991.

Why Speakers Have Slanted Fronts, Part 3

A look at which real-world models work and which don't.

By G. R. Koonce

Part 3 examines a few systems that were actually developed and compares the polar plot with the system directivity plot so you can see which approach you find more useful. In all these systems the summation was done at a point 60″ away from the front panel to compare with testing results. Generally you would work with a fixed summation distance closer to your actual listening distance. Experience has shown that increasing the summation distance will generally widen the usable vertical lobe a small amount. Thus a system that looks good at 60″ will look as good or better at a greater distance.

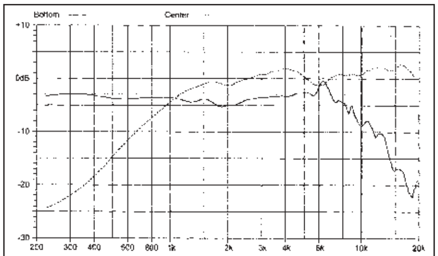


FIGURE 17: Bare driver responses for system BT2W-RS5. B-2330-17

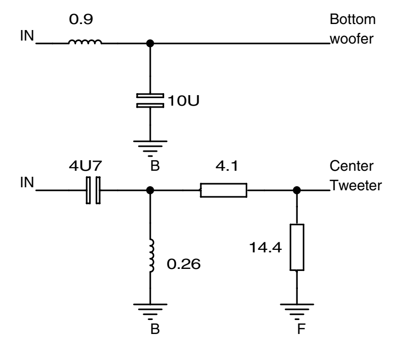


FIGURE 18: Crossover networks for system BT2W-RS5. B-2330-18

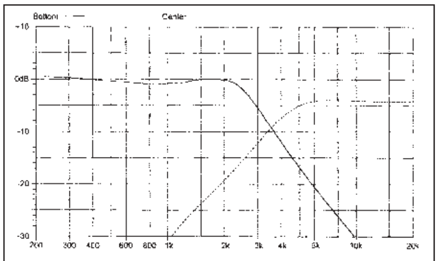


FIGURE 19: Crossover network electrical responses for BT2W-RS5 system. B-2330-19

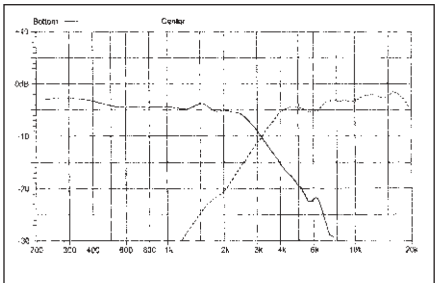


FIGURE 20: Acoustic on-axis responses for drivers with crossover for BT2W-RS5 system. B-2330-20

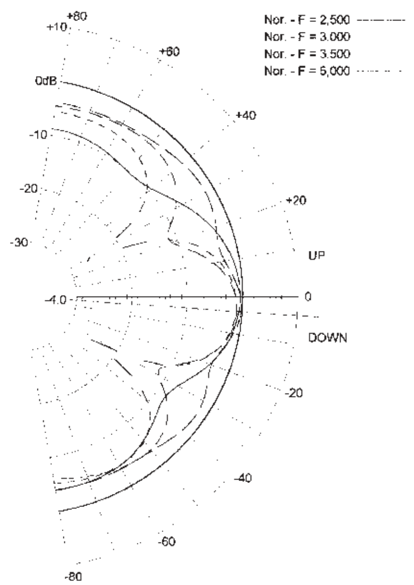


FIGURE 21: Vertical polar plot for system BT2W-RS5. B-2252-21

SOME REAL SYSTEMS

The BT2W-RS5 system is a small woofer two-way using the SEAS CA11RCY woofer of about 4½″ with a Vifa D25AG-35-06 1″ dome tweeter. *Figure 17* shows the on-axis frequency responses for the bare drivers indicating a wide overlap range.

The developed CO was second-order with an L-pad on the tweeter *(Fig. 18)*. Note here that the tweeter is marked as "center" because this modeling software is designed to handle up to three-way systems. I will still refer to the tweeter as

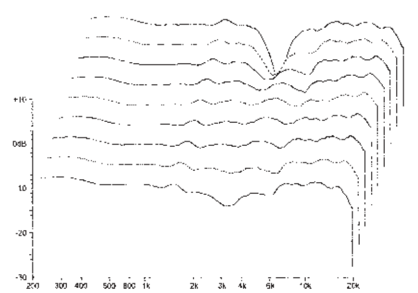


FIGURE 22: Vertical directivity plot for system BT2W-RS5. B-2330-21

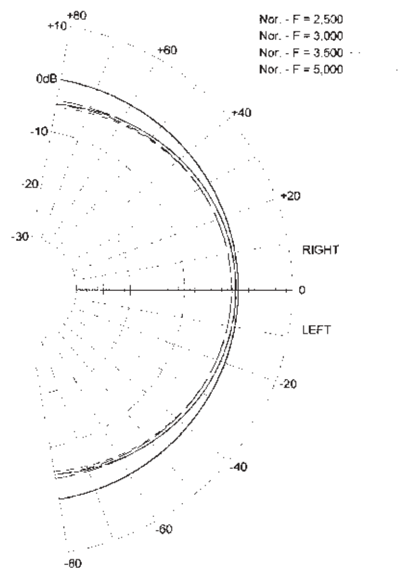


FIGURE 23: Horizontal polar plot for system BT2W-RS5. B-2252-23

the top driver. The electrical response of the two CO networks loaded by their driver is shown in *Fig. 19*. Slight peaking is used in the LP to flatten the top end of the woofer response.

Figure 20 shows the on-axis acoustic responses of the drivers with their CO network showing a near LR2 acoustic CO at about 3,200Hz. Note, however, that this acoustic CO shape sums with

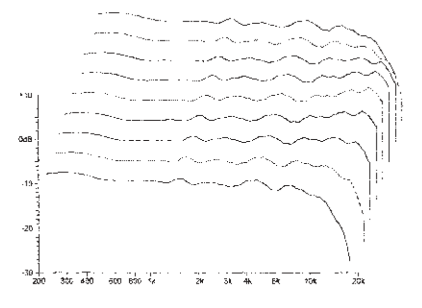


FIGURE 24: Horizontal directivity plot for system BT2W-RS5. B-2330-24

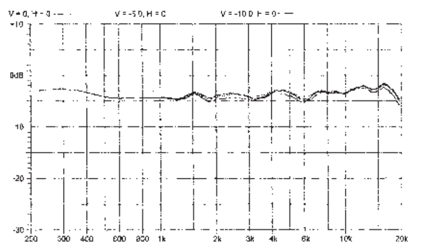


FIGURE 25: Response of system BT2W-RS5 at 0, –5, and –10 degrees vertically off-axis. B-2330-25

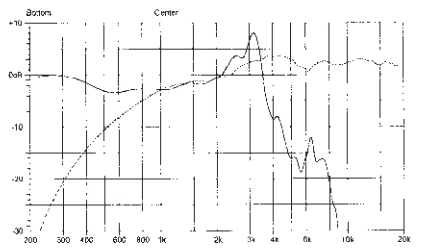
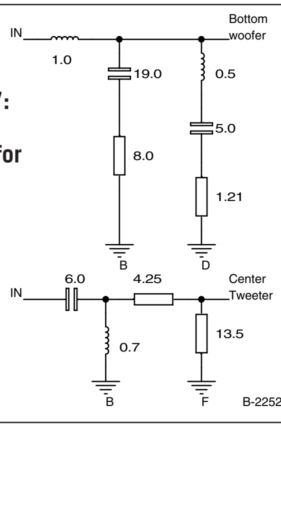


FIGURE 26: Bare driver responses for system Sys119. B-2330-26

FIGURE 27: Crossover networks for system Sys119.



B-2252-27

both drivers wired with the same polarity. So much for that rule of thumb about second-order COs needing the tweeter inverted. From what you have seen in Part 2 you would expect the main lobe somewhat below on-axis.

Figure 21 shows the vertical polar plot

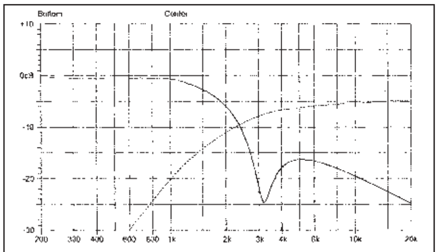
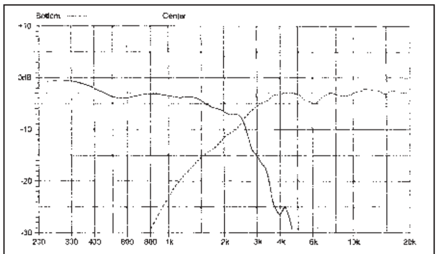


FIGURE 28: Crossover network electrical responses for system Sys119. B-2330-28



B-2330-29

FIGURE 29: Acoustic on-axis responses for drivers with crossover for system Sys119.

indicating that for the frequencies plotted the main lobe is at about –4° and the system response looks good from about on-axis to at least –10°. *Figure 22* is the vertical directivity plot and agrees well with the polar plot showing the system response is usable from +5° to –10°. *Fig-*

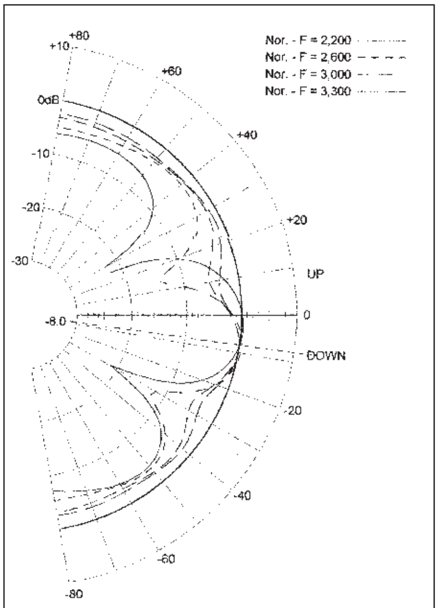


FIGURE 30: Vertical polar plot for system Sys119. B-2252-30

ure 23 is the horizontal polar plot showing the system response is near circular for the frequencies plotted.

The horizontal directivity plot (*Fig. 24*) agrees with the polar plot and shows the system response only falls off at high frequency due to the directivity of the tweeter. Finally, *Fig. 25* shows the response of this system at vertical angles of 0, –5°, and –10° and a horizontal angle of zero degrees—a nice flat and smooth response that was verified in testing. Clearly, when you deal with a small woofer and a tweeter that goes low in frequency, developing an acceptable system is relatively easy.

Best use of this system requires building it so the seated listener will be on an angle of about 4° below the tweeter axis (–4°). Later we will look at ways to accomplish this.

AN 8″ WOOFER TWO-WAY SYSTEM

Now for a much tougher challenge. Two-way system #119 uses a Carboneau #24882 8″ woofer with a Vifa D27TG-35-06 1″ silk dome tweeter. *Figure 26* includes the on-axis acoustic responses of the bare drivers, showing just how much fun an 8″ woofer two-way system can be. Through most of the frequency overlap range of the drivers, the woofer shows a nasty response peak about 15dB above the basic woofer response. The CO design must account for this peaking.

Figure 27 shows the CO that was developed. The HP is a straight second-order with an L-pad. The LP looks like a first-order with Zobel and a series tank

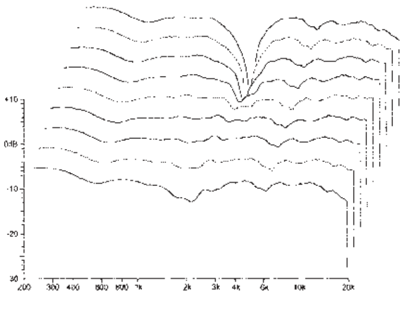


FIGURE 31: Vertical directivity plot for system Sys119. B-2330-31

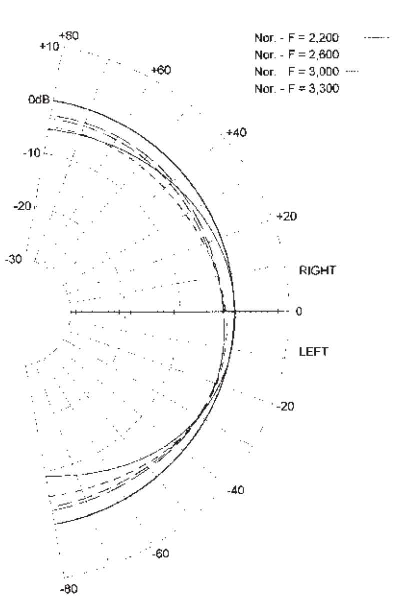


FIGURE 32: Horizontal polar plot for system Sys119. B-2252-32

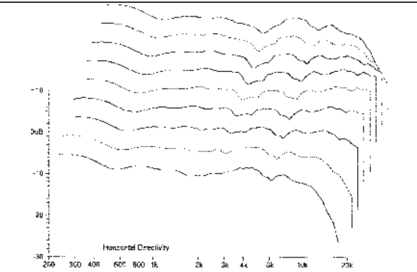
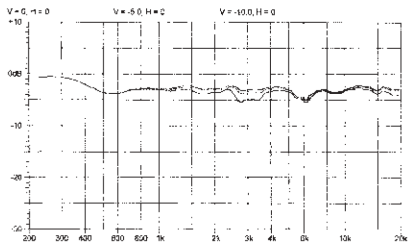


FIGURE 33: Horizontal directivity plot for system Sys119. B-2330-33



B-2330-34

FIGURE 34: Response of system Sys119 at 0, –5, and –10 degrees vertically off-axis.

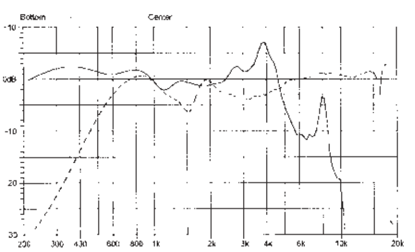


FIGURE 35: Bare driver responses for System-S. B-2330-35

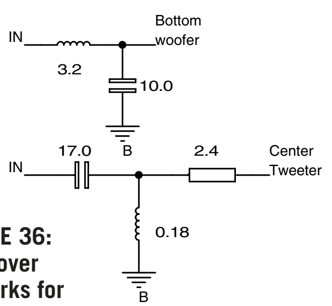


FIGURE 36: Crossover networks for System-S. B-2252-36

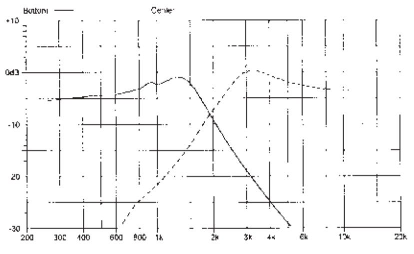


FIGURE 37: Crossover network electrical responses for System-S. B-2330-37

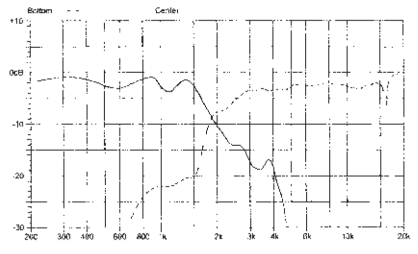


FIGURE 38: Acoustic on-axis responses for drivers with crossover for System-S. B-2330-38

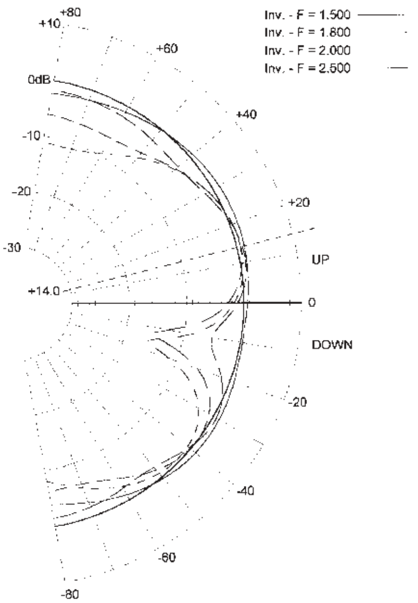


FIGURE 39: Vertical polar plot for System-S. B-2252-39

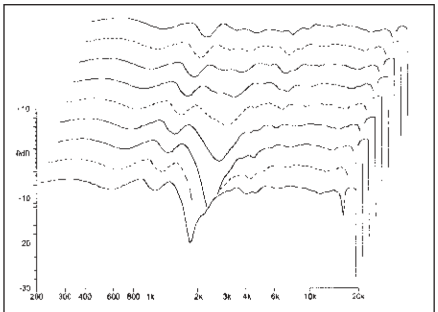


FIGURE 40: Vertical directivity plot for System-S. B-2330-40

circuit across the woofer.

Actually, this is what I call a "strange" second-order CO (without the tank circuit). It may look like a first-order, but it behaves like a second-order, where the resistor in series with the large value capacitor is used to control the Q of the network. The shunt tank circuit is used to suppress the peak in the woofer by dropping its impedance in the peak's frequency range.

Figure 28 shows the CO networks' electrical responses loaded by their driver. The notch about the woofer's peak frequency is clearly visible. *Figure 29* shows the on-axis acoustic responses for the drivers with their CO network. The CO frequency is about 2.6kHz and the tweeter has nearly a second-order slope, while the woofer response starts with a low slope and then increases to a high slope—clearly asymmetrical CO slopes.

Figure 30 is the vertical polar plot showing the main lobe at about –8°. Note again that both drivers are driven with normal polarity. *Figure 31* is the vertical directivity plot showing the system good at negative angles (Down), but not at positive angles (Up). This agrees with the polar plot, but I believe it is much easier to understand. One point to note is that the directivity plots give you information about what frequencies you should plot on the polar plots.

Figure 32 is the horizontal polar plot for the system and looks reasonably smooth even with the tweeter offset 1″ from the woofer's centerline. *Figure 33*, the horizontal directivity plot, confirms that the system goes away smoothly in the horizontal plane. *Figure 34* shows the system frequency response at vertical angles of 0, –5 and –10° and a horizontal angle of zero degrees relative to the tweeter's axis. This system should be built so the seated listener is on a vertical line in this 0 to –10° range. I actually built this system which sounded very nice.

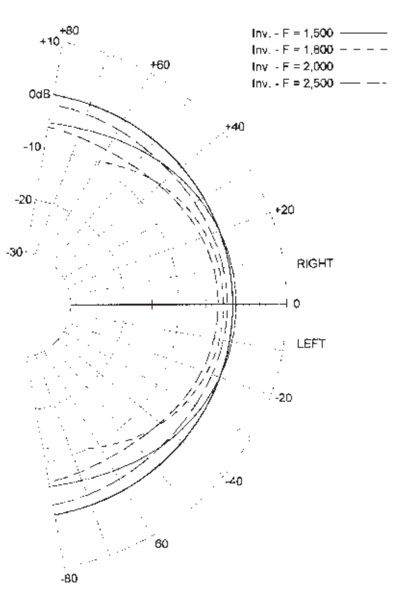


FIGURE 41: Horizontal polar plot for System-S. B-2252-41

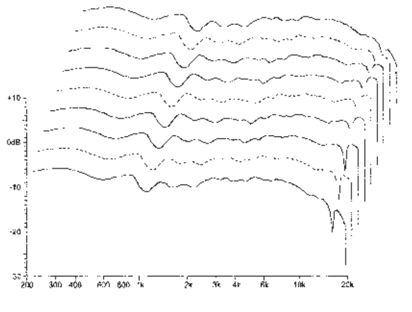


FIGURE 42: Horizontal directivity plot for System-S. B-2330-40

AN UNUSUAL SYSTEM

System-S is a two-way system using an

MCM 55-1550 8″ woofer and an Audax TW025V2 low-resonance 1″ dome. *Figure 35* shows the bare driver on-axis responses, which are sure not encouraging. The woofer has a nasty peak at 4kHz and the tweeter does go down to 1kHz, but has a nasty dip at 1.5kHz. The developed CO (*Fig. 36*) contains only five components and it is a second-order LP and a second-order HP with series padding on the tweeter.

Figure 37 shows the electrical responses of the CO networks, which show considerable peaking, being almost M-derived networks. This gives a very fast fall-off in the first octave from CO, but can have some adverse effects, especially on system input impedance and power handling. The on-axis responses of the drivers with their CO networks are shown in *Fig. 38*. The acoustic CO point is about 1.9kHz with both responses down near 6dB at CO. The slopes around CO are quite high as the CO network responses combine with the driver anomalies.

Figure 39 shows the vertical polar plot for this system with the tweeter wired inverted polarity. The system response has a wide lobe centered near +14°. The vertical directivity plot (*Fig. 40*) confirms a system that has a nice, consistent response from slightly above on-axis to at least +20°. Here is a floor-standing system you can build with a vertical front panel!

The horizontal polar plot (*Fig. 41*) shows the system degrades gracefully in the horizontal direction in the CO frequency region. The horizontal directivity plot (*Fig. 42*) shows that it degrades gracefully at all frequencies over the ±40° range. Note that *Fig. 42* is not a directivity plot in the horizontal plane through the tweeter's center. The system response at zero degrees vertical was slightly degraded so *Fig. 42* is the horizontal directivity about a vertical angle of +20°. It is the response moving along a cone making a 20° angle to

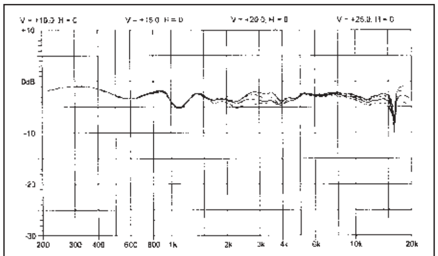


FIGURE 43: Response of System-S at +10, +15, +20, and +25 degrees vertically off-axis. B-2330-40

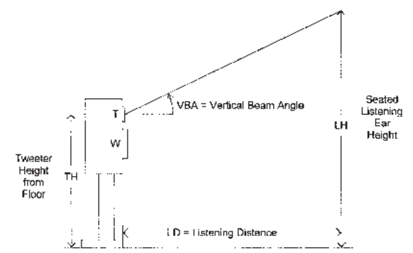


FIGURE 44: Conditions for a stand-mounted enclosure. B-2330-44

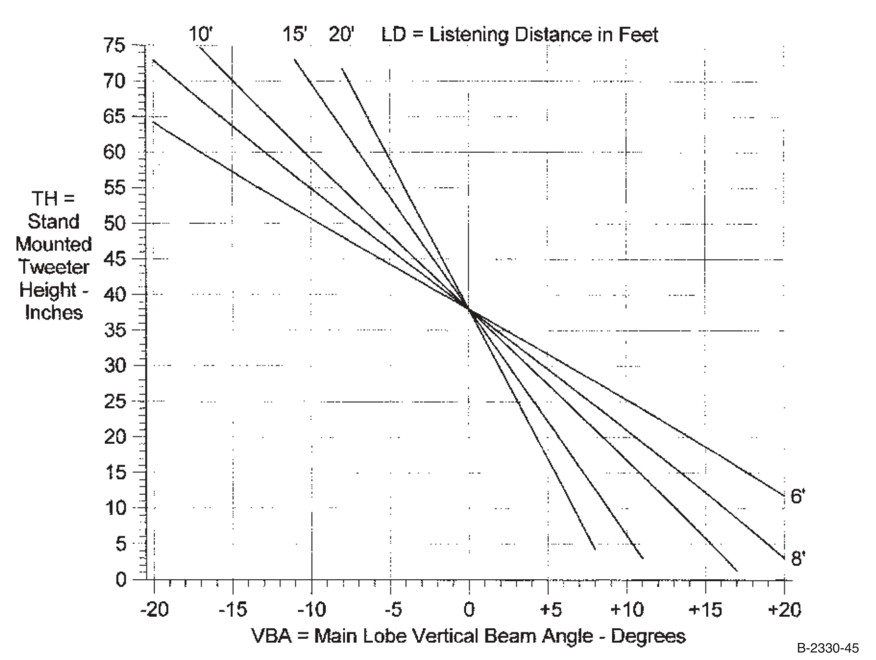


FIGURE 45: Plot to allow establishing tweeter height for stand-mounted enclosures.

horizontal. This is the equivalent of a horizontal polar plot made some distance above the tweeter.

Figure 43 shows the system response at +10°, +15°, +20°, and +25° vertical angle and zero degrees horizontal angle. The performance of this system was verified in testing and I built System-S using a slightly modified form of the CO which worked just fine. I just loved being able to build one pair of boxes with a vertical front panel.

SUMMARY

This work has demonstrated that the performance of COs (passive or active) with real drivers can be far from the ideals predicted for various electrical CO function shapes. To assure a system that degrades gracefully in the horizontal plane for good imaging, a near vertical alignment of drivers is best. Offsetting a midrange or tweeter by an inch or so to spread out diffraction at the cabinet edges is not a problem, but large lateral offsets can be harmful.

Generally you will obtain a system response lobe in the vertical plane that is only "good" over about 15° to 25°. It is important to build and use the enclosure in such a way that the listener is placed within this useful vertical lobe, surely for seated listening and, you hope, also for standing listening.

You have also seen that implementing your favorite electrical CO function—either active or passive—does not mean you will get the acoustic CO function that you wanted. This means all rules of thumb about inverting certain drivers for certain CO functions are questionable at best and generally dangerous to follow.

Generally, it is clear that the bigger the driver, which forces a bigger CTC spacing, the lower you need to try to make the acoustic CO frequency. Many dome tweeters may test to go low in frequency, but unless the maker recommends such application, going low with the tweeter may be dangerous. To properly work down low, the tweeter needs some form of suspension on the dome and "flying" lead wires.

HOW TO PUT THE LISTENER ON THE OPTIMUM AXIS

To accomplish putting the listener on the best axis, you must know the location of that axis. The end of the article addresses this question. Here we assume you have used a nearly vertical alignment of the drivers for a nice wide, smooth horizontal lobe. You also know the location of the vertical lobe, and I will refer to the center of this lobe as the vertical beam angle (VBA) relative to the tweeter axis being zero degrees.

Your goal is to position the enclosure front panel so that the VBA, or close to it, points at the seated listener. You also hope the standing listener will still be in the usable vertical lobe. Here are three ways to accomplish this:

1. Build the enclosure with a vertical front panel and place it on a stand of the proper height. This is really practical only for small enclosures.
2. Build floor-standing enclosures with the front panel tipped at the proper angle. David Weems has developed an alternate way to accomplish this same thing. You build a rectangular box and sit it on very low stands that tip the entire enclosure.[11]
3. Build enclosures with the drivers side-by-side. This interchanges the horizontal and vertical lobes. You build mirror image boxes that are used with the tweeters either both outboard or both inboard. This allows rotating the enclosures to sit the listener on the now narrow horizontal lobe.

   This approach clearly violates the concept of having a wide and smoothly degrading horizontal response, but many listeners who have heard such systems liked the sound very much. They offer a very wide listener "sweet spot," but do have an image a bit less specific than more conventional systems. I do not cover this approach in this work, but it is documented in reference 12.

The equations for solving the stand-mounted box and the tipped front panel floor-standing box are shown in the equations sidebar included with Part 4 of this article. For those who don't like the math, I offer some plots that will accomplish the same thing. The height of a seated listener's ears varies from maybe 34 to 44″, the plots use a value of 38″. Similarly, the height of a standing listener can vary a bit. I chose a value of 66″. Listening distances range from 6 to 20′, which is the distance from the enclosure front to the listener.

CALCULATING STAND HEIGHT

Figure 44 shows the conditions for a stand-mounted enclosure with a vertical front panel and the tweeter above the woofer. The up angles are positive in this figure and have been exaggerated for clarity. You should select a stand height that will put the tweeter at the proper height to aim the VBA at the seated listener.

Figure 45 shows plots of tweeter height from the floor (TH) versus vertical beam angle (VBA) for a variety of listening distances (LD). Find the VBA on the horizontal axis and move vertical to the closest LD for your application. Move horizontally from this point to the vertical axis and read the tweeter height from the floor (TH). You need to select stands that will mount your enclosures with the tweeter at the specified height. It is clear that if the VBA is positive, as shown in *Fig. 44*, then the tweeter must be below 38″ while a negative VBA means the tweeter must be above 38".

Consider an example using the small BT2W system discussed earlier. The VBA for this system was –4°. For a 10′ listening distance, in *Fig. 45* you would move vertically from –4° to the LD = 10′ line and then horizontally to get a TH value of 46″. This means the required stands should place the tweeters at about 46″ off the floor.

Now, if you prefer lower stands, you could build the enclosures with the tweeter mounted below the woofer, which works just fine. That is, turn the boxes upside down. This changes the sign of the VBA so it is now +4°. Using *Fig. 45* again for +4° and LD = 10′ yields a lower TH of 30″. This shortens the required stands somewhat, but remember the tweeter is now lower in the enclosure.

We'll wrap up this four-part series next month with some construction and positioning tips. ❖

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11. Weems, David B., and Koonce, G. R., *Great Sound Stereo Speaker Manual—Second Edition*, McGraw-Hill, 2000, p. #43.

12. Koonce, G. R., "Side Saddle: A New Two-Way," *SB* 3/97, p. #10.

Product Review

The Usher CP8871

By James Moriyasu

Usher is not a name that quickly brings to mind a state-of-the-art listening instrument. The word commonly refers to one who escorts people to their seats. However, in the Merriam-Webster online dictionary, that definition is third on the list. The first explanation of usher is: "an officer or servant who has the care of the door of a court, hall, or chamber." Having had the opportunity to live with these speakers for several months, I like to think of them as gatekeepers to a chamber of listening pleasure.

Readers will recognize the Usher Compass Dancer CP8871 for its distinctive, curve-sided cabinets, displayed in Usher's full-page ads. Also quite prominent in the ad was the fact that the crossover design, prototype testing, and final fine tuning for the Usher 8871 was performed by Joe D'Appolito, a regular contributor to *audioXpress*. Needless to say, Joe is more than just a contributor. Usher's brochure describes him as "a world renown authority in audio and acoustics." Sales brochures can contain a fair amount of hyperbole, but with regard to Joe, this wouldn't be the case.

I've been familiar with Joe's designs for more than 15 years. My interest started with the Swan IV design article in *Speaker Builder*, which I studied and built. Since then I have pored over his designs with much interest and have employed some of his ideas in my own projects. You might consider me a reviewer who has the advantage of being familiar with the work, but wonder whether I lack enough familiarity to be an unbiased critic.

NOT FOR THE FAINT OF HEART

The CP8871 is manufactured by Usher Audio Technology of Taipei, Taiwan. You'll find more information on the company at: www.usheraudio.com

The Usher CP8871 is part of the Compass Dancer series of loudspeakers, which is a subset of the Compass series. It is a large speaker, weighing 76kgs, or 167.2 lb, and is 52.75″ tall. While it is just 12.4″ wide, it is 33.5″ deep, so it does take up a bit of floor space (*Photo 1*).

Fit and finish is excellent. Your eye is drawn to the vast expanse of curved hardwood, which sweeps to the rear and to the top, which reminds one of the prow of a ship. At the front you can't help but notice the large radiused area above the tweeter and the generous beveled edges on either side of the tweeter or mid/bass. Also worth noticing is the thick, solid wood butcher-block plinth that forms the base of the cabinet (*Photo 2*).

DRIVERS

The Compass Dancer series of loudspeakers all use the same distinctive cabinet design, but their sizes vary from 57–87.5kgs. Within the series there are two variations. The AC models use Accuton drivers from Thiele and Partner, while the CP models use Usher drivers. The CP8871 uses two Usher 8955 8″ woofers, one 8945 7″ midrange, and one 9950 1″ soft dome tweeter.

Worth noting here is that the Usher drivers bear a striking resemblance to Scan-Speak drivers. The 9950 tweeter, for example, has almost the same plastic beveled ring as the Scan-Speak D2905/9300 just beyond the surround. The pattern of mounting holes and structural screws is more or less identical. The cones and the baskets of the mid/bass and woofers are very similar in appearance. For example, the damping material appears to have been smeared or swirled on the cone surface, much the way Scan-Speak cones look. And the baskets have the familiar

PHOTO 1: The Usher CP8871 loudspeaker.

PHOTO 2: Note the unique cabinet design and driver treatment of this high-quality unit.

six spokes.

As it turns out, after contacting Usher, I was informed that they and Scan-Speak use the same cone source. Usher also said that their drivers are pair-matched to within 0.02dB, and that they have thicker plating for the motor.

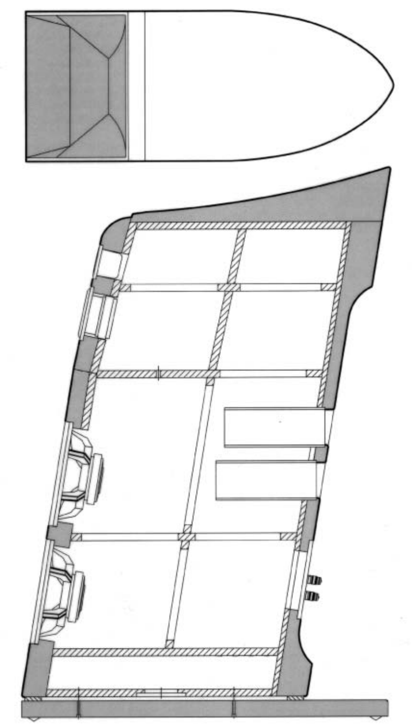

FIGURE 1: Compass Dancer CP8871 side view cross section.

The 8955 woofer features a carbon fiber/paper cone, vented pole piece, and distortion-reducing rings at the pole piece. It has a very large magnet, the biggest I have ever seen on an 8″ woofer. Resonant frequency of the driver is specified at 24Hz; the 18mm voice coil and 6mm gap result in a maximum excursion (X_{MAX}) of 6mm, one way. Moving mass at 23.5g is relatively light for a driver of this size, resulting in a Bl of 7.47 and total Q (Q_{TS}) of 0.32.

By comparison, the Scan-Speak 21W/8555-00 has an Fs of 20Hz, X_{MAX} of 6.5mm, moving mass of 32g, Bl of 8.2, and Q_{TS} of 0.31. So while this Usher driver may resemble the Scan-Speak in appearance, there isn't that much similarity in driver parameters.

The 8945 7″ mid/bass also has a very large magnet, which is specified at 1.1kg! And while its appearance is similar to that of the Scan-Speak 18W/8545-00, its driver parameters are dissimilar. The same can be said about the 9950 tweeter: while it looks like a Scan-Speak it has very different driver parameters.

Regardless of whether the Usher drivers look like Scan-Speak drivers but measure differently, it is my opinion that these are still very high quality drivers in terms of features and build quality. More than likely they are well designed and produced. Do they sound as good as a Scan-Speak? That is a difficult question to answer, because other factors such as cabinet and crossover design come into play.

The Usher drivers are now available for purchase from Euphase Audio and Zalytron at: zalytron.com, www.euphase.com. The Euphase website includes photos and detailed specifications and measurements of the drivers that were done by Vance Dickason (May '01 *Voice Coil*). Some designers have already posted designs on the web. For example, try MurphyBlaster Productions at: murphyblaster.com

CABINET

The CP8871 cabinet is definitely not your father's speaker cabinet. Any resemblance to your basic rectangular box is hard to find. *Figure 1* is from the Usher brochure on the Compass Dancer series. The front baffle is approximately 2¼″ thick and comprises four layers of MDF (2 × 25mm, 1 × 15mm, and 1 × 9mm). The curved sides are built with 25mm MDF covered by 3mm of hardwood. The inside edge of the mid/bass hole is not relieved or chamfered.

The inside of the cabinet is braced with shelf braces made of ¾″ MDF with "window" frames approximately 1½″ in width. The surfaces are lined with a ⅛″ asphalt damping sheet, followed by a layer of 1″ egg-crate foam, which is covered with ½″ layer of light density felt.

Overall, the cabinet construction could be characterized as "robust," but certainly not of the "take no prisoners" approach. This is quite apparent when you rap the front baffle or the sides with your knuckles. The front is very quiet due to its narrow width and extra thickness; however, the wide side panels sound relatively loud by comparison. They could benefit from more bracing or being made thicker.

The twin ports are cardboard and are not flared. This is a glaring deficiency for a speaker of this caliber. Numerous studies have well established the benefits of flared ports.[1,2] Loudspeakers with such vents exhibit a significant reduction of "chuffing" noises and generate 1–3dB more output below 40Hz.

The wiring to the drivers is with Jenn Heu Super OFC 16 gauge speaker cable, which is terminated with female push connectors that are not soldered into place. The input panel is composed of a 5⁄16″ thick, gold-plated, satin-finished aluminum plate with two pairs of satin-finished gold plated locking binding posts with jumpers that you can remove for bi-wiring. It is a very attractive combination, and the input panel is fastened with 14 screws, giving it an industrial-grade look.

The drivers are held in place with a type of Allen-socket machine screw that is flat on top but has rounded sides. T-nuts anchor the machine screws. The drivers are flush-mounted with the front baffle and blend in well with the polished black front surface. The ½″ removable loudspeaker grilles are covered with a black fabric and held in place with male/female fasteners.

The top of the speaker is solid butcher-block and is between 1¼″ to 5″ thick. It is quite attractive if you like the look of laminations of solid wood. The plinth is also solid butcher-block and has five ⅜″ threaded inserts made of brass. Brass cones 1⅛″ wide and 1⅜″ tall with ⅜″ threads are provided.

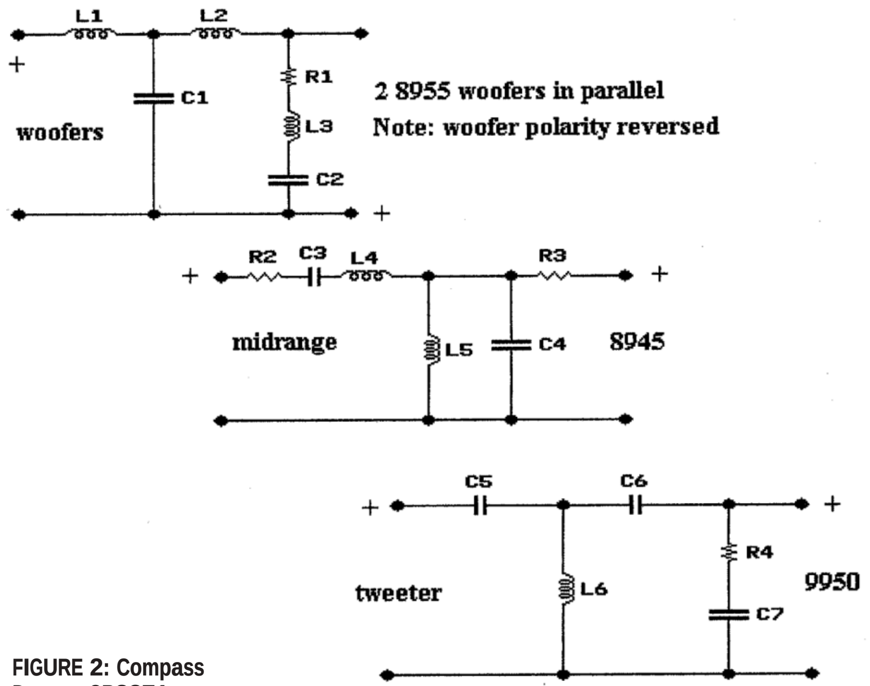

FIGURE 2: Compass Dancer CP8871 crossover schematic.

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CROSSOVER CONSTRUCTION

The woofer crossover is on a separate printed circuit board and is screw-mounted to the side panel nearest the lower woofer. All of the components are soldered to the board; nylon ties are used to secure the inductors, only. A quarter-inch layer of foam rubber insulates it from the cabinet side. It uses air core inductors for the coils that are in series with the driver and has a transformer core for the large conjugate inductor that is in parallel with the woofer.

Oddly enough, all three coils are oriented in the same direction and are placed quite closely to each other. There is only a ½″ space between the transformer coil and the smaller of the air core inductors. The larger air core is about an inch from the smaller one and about 3″ from the transformer coil. My concern was that this arrangement would not result in the lowest level of crosstalk between the inductors[3].

The capacitors are electrolytic and there are two (what appear to be) sand-cast, 15W resistors for the conjugate. The resistors are suspended on metal tabs that serve as leads and as mounting brackets that keep the resistors ½″ from the board. No glue or damping compound is used to keep the components from vibrating.

The mid/bass and tweeter crossover circuits are on a printed circuit board adjacent to the woofer board separated by a vertical, ¾″ MDF brace. Air core coils are used throughout, along with some type of film capacitor. The supposed sand-cast, 15W resistors are also used. Again, the inductors are all oriented in the same direction with their poles perpendicular to the plane of the circuit board. A pair of the coils is less than an inch apart; a third is about 5″ away.

Aside from the orientation of the inductors, it appears that great care was taken in the design and implementation of the crossover. A relatively minor modification for this speaker would be to re-orient the coils to reduce the crosstalk between them. Perhaps higher quality resistors could be used, also.

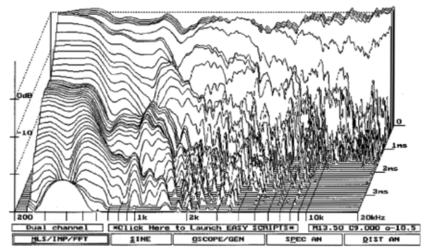
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FIGURE 6: Cumulative spectral decay of CP8871 over 3mS.

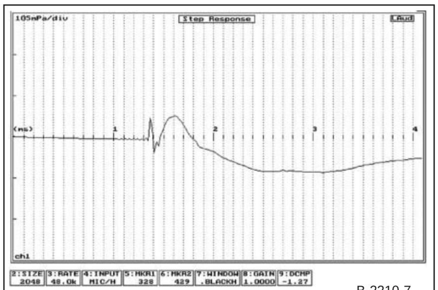
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FIGURE 7: CP8871 system step response; measured from 2 meters.

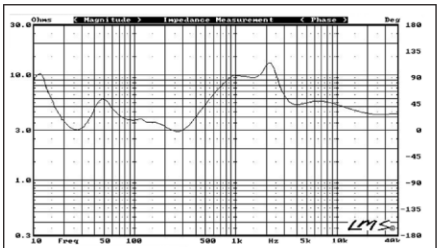
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FIGURE 3: Full-range loudspeaker system impedance.

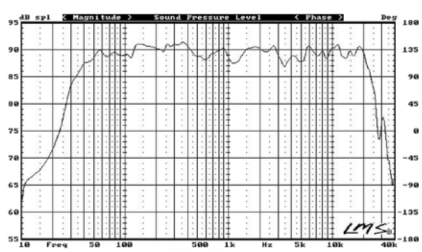
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FIGURE 4: Full-range loudspeaker system on-axis SPL; sine wave gated measurement from 2 meters with 4W; spliced to a ground plane measurement at 245Hz.

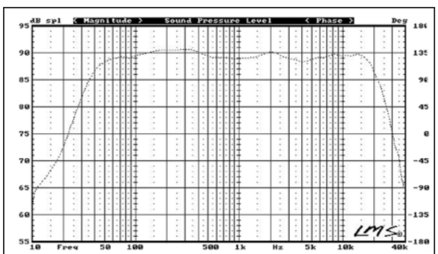
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FIGURE 5: Full-range on-axis SPL measurement smoothed by one octave.

CROSSOVER TOPOLOGY

Figure 2 shows the crossover topology without the component values. The woofer crossover is third order electrical. It is made up of L1, L2, and C1. It has a series RLC conjugate or shunt that is parallel with the woofers, which is made up of R1, L3, and C2.

L1 and L2 appear to be 16 or 14 gauge air core coils, while L3 is a transformer core. C1 and C2 are large electrolytic capacitors. R1 is a 15W sand-cast resistor. The woofers are wired in parallel with their polarity reversed; that is, the negative terminal of the woofers connects to the positive of the crossover output.

While it is preferable to connect drivers with "normal" polarity—that is, with the positive on the driver to the positive of the crossover—sometimes it is impossible to do so with a particular crossover topology and have the driver outputs sum properly. This is due to the differences in driver phase, I think, and may be specific to these drivers. I have built three-way systems with the woofer wired with normal polarity so, apparently, the ability to do so does vary with the driver set.

The series RLC conjugate is used to damp or flatten the woofer's upper impedance peak. Without the shunt the woofer's sound pressure level would typically rise by up to 2dB between 50–150Hz.

The midrange crossover is second order electrical with no RLC conjugate. C3 and L5 make up the high-pass section, while L4 and C4 form the low-pass part. L4 and L5 are air core inductors, while C3 and C4 are film capacitors. R2 and R3 are 15W sand-cast resistors. The midrange is connected with normal polarity.

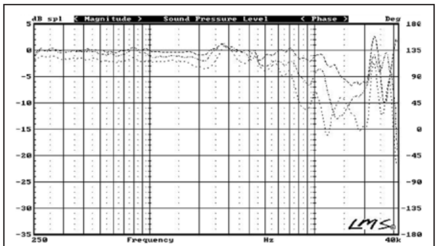
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FIGURE 8: System 15, 30, and 45° horizontal off-axis SPL measurement referenced to on-axis.

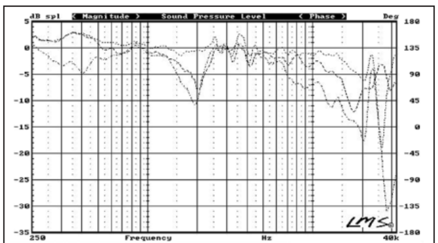
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FIGURE 9: System 15, 30, and 45° vertical off-axis SPL measurement referenced to on-axis; measured below the tweeter.

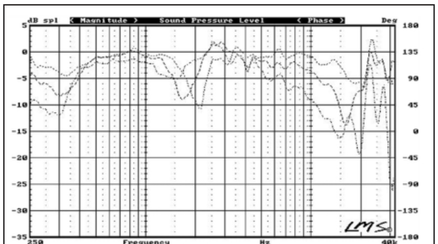
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FIGURE 10: System 15, 30, and 45° vertical off-axis SPL measurement referenced to on-axis; measured above the tweeter.

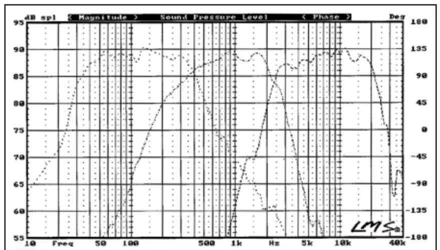
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FIGURE 11: Individual driver responses.

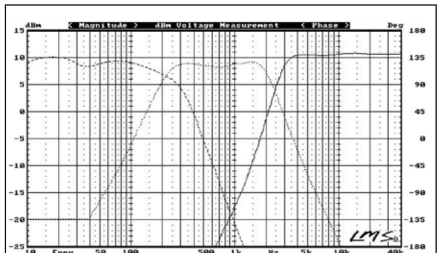
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FIGURE 12: Individual driver crossover voltage transfer functions.

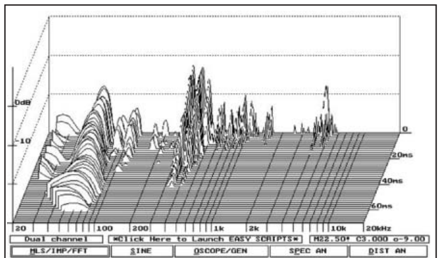
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FIGURE 13: Cumulative spectral decay of accelerometer measurement of front panel.

However, the placement of R2 raised my eyebrows. A resistor's placement in a crossover network can affect the shape of the network's transfer function—in particular, the damping at the "knee" of the transfer function. So placement of the resistor can result in a sharp or more rounded "knee" at the crossover point[4]. The need to shape the transfer function might be the reason why R2 is at the front of the crossover network.

The trade-off with this placement is power handling, because the resistor must bear the full bandwidth and power of the input signal. In general, I guess, placing a resistor after a capacitor would be preferable, because most loudspeaker capacitors are rated at 100V or more and can, presumably, do a better job than a 15W resistor when it comes to handling power. Placing a resistor after an inductor would probably be preferable, too, since the inductor would attenuate a portion of the input signal.

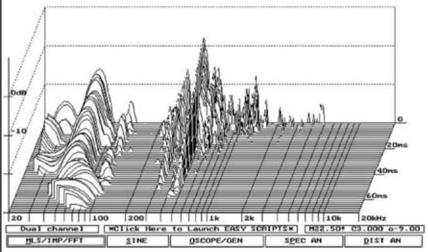

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FIGURE 14: Cumulative spectral decay of accelerometer measurement of side panel.

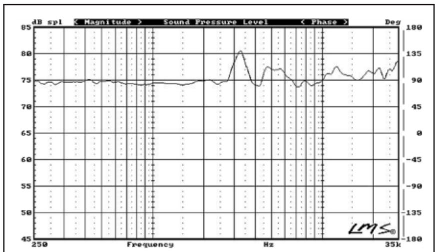

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FIGURE 15: Difference curve of speaker system SPL with and without grille.

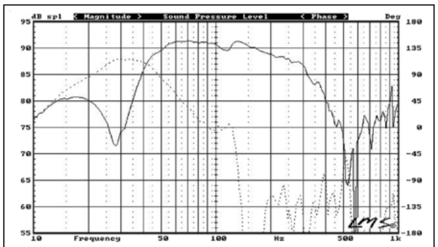

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FIGURE 16: Near-field measurement of woofer and port.

Coincidentally, while working on this review, I received a call from a friend who had burned out the resistor in the tweeter crossover of both of his Tannoy TD12 loudspeakers. Upon examination of the crossover it appeared that the resistor was placed ahead of the capacitors, so they had to handle the full power from the amplifier. And it didn't help that the resistors were tiny, the size of a dime only thinner, and only rated at 10W. Fortunately, I was able to supply him with a replacement pair of 15W Lynx resistors.

The tweeter crossover is third order electrical with an impedance equalizer, or zobel, in parallel with the tweeter. It includes capacitors C5 and C6 and inductor L6. Capacitors C5 and C6 are film, and L6 is an air core coil. The tweeter is connected with normal polarity. The zobel, R4 and C7, flattens the rising impedance of the tweeter. Without a zobel most tweeters will exhibit a rising response above 6–8kHz.

MEASUREMENTS

I measured the impedance of the speaker system with a LinearX VI box. This allows measurements at realistic power levels. In this case I measured at 4W into 8Ω, or 5.66V. As you can see in *Fig. 3* the curve is mostly between 3 to 10Ω with a peak at 13Ω at 2200Hz. Because the woofers are wired in parallel, the impedance drops to 2.9Ω at 269Hz and is just around 3Ω at the box tuning at 27Hz. There is an anomaly in the curve just above 100Hz with a peak centered at 115Hz. I will examine this in more detail later.

Aside from this bump, the impedance curve is relatively smooth. Bumps in impedance curves can often indicate a resonance of some type. There could be a resonance in the cabinet volume, port, or cabinet panel. Or there could be a breakup mode in the driver's cone or a basket/magnet interaction.

Figure 4 is a full range, on-axis sound pressure level (SPL) measurement of the speaker system. I raised the speaker 7′ off the ground, with the microphone on-axis with the tweeter and two meters from it. A gated sine wave measurement was made with Loudspeaker Measurement System (LMS) from LinearX. The voltage for the measurement was scaled up to be comparable to a measurement at one meter with 1W. The curve was smoothed with ¹⁄₁₀ octave averaging. All of the SPL measurements were taken with the grille frames removed.

The microphone for measurement was the standard LMS microphone. To make sure it is accurate, I compared it against another standard LMS microphone that I use only for calibration purposes. The mikes were ±0.1dB below 2kHz and ±0.5dB above that level.

I spliced the gated measurement to a ground plane measurement at 245Hz. The ground plane measurement was made at 2m with the speaker positioned vertically. I mention this because with the double woofers, a ground plane measurement with the speaker lying on its side is very different above 100Hz than with the speaker standing up.

Aside from a 1dB dip at 3.5kHz, the speaker's SPL curve is relatively smooth since it is ±1.5dB between 40Hz and 20kHz. Above 500Hz the dips and peaks are mostly caused by diffraction so the SPL curve suggests the individual driver responses are smoothly integrated. Sensitivity is around 89dB. Below 500Hz, however, there is a serious bump at 125–150Hz, which is probably related to the anomaly in the impedance curve.

Figure 5 is the SPL curve of the system, but this time with the data smoothed by one octave averaging. I did this to get a better perspective on the system's relative octave by octave differences. The system appears to be centered at 89dB with a broad 2dB hump between 100Hz and 400Hz. There's a narrow peak and dip at 2kHz and 4kHz, respectively.

Above 5kHz the response is flat to 15kHz. Relative to 89dB the low-frequency response is down 3dB at 38Hz, which suggests good but not great bass extension. The broad hump at 100–400Hz suggests the speakers will sound a little forward in the mid bass.

I measured the system's impulse response with Liberty Audiosuite from Liberty Instruments. This measurement was taken from two meters and on-axis to the tweeter with the speaker on a manual forklift raised 7′ above the ground. The impulse was then processed to show how the speaker system behaves over a 3ms period. *Figure 6* shows the cumulative spectral decay above 200Hz. The system's decay is very good with no major signs of resonances.

Figure 7 is the step response of the system. The initial spike at 1.37ms is the tweeter, which is followed by the mid/bass, about 200µs or more, later. The woofer is the broad, inverted hump that follows the mid/bass. Because the woofers are wired in reverse polarity, their impulse response is negative relative to the tweeter and mid/bass.

I also made SPL measurements 15, 30, and 45° off-axis from the tweeter, horizontally and vertically. I then subtracted the measurements from the on-axis measurement to produce a difference curve. *Figure 8* shows the system's horizontal response to be very broad with the 15° difference curve down 3dB at 12kHz and the 30° difference curve down 3dB at 7.5kHz. This broad dispersion is mostly due to the tweeter that is crossed over at 2.3kHz.

However, there is a 1.5dB peak at 2.7kHz and a 2dB dip at 4.5kHz in the 15° difference curve, which I suspect is related to cabinet edge diffraction. So despite the large rounded-over area above the tweeter and the wide bevels,

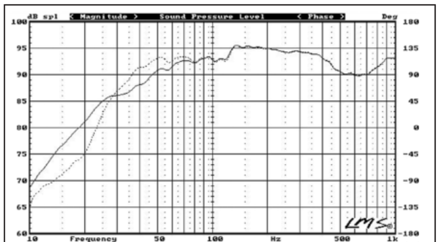

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FIGURE 17: Comparison of system with port blocked (solid line) and system with port open.

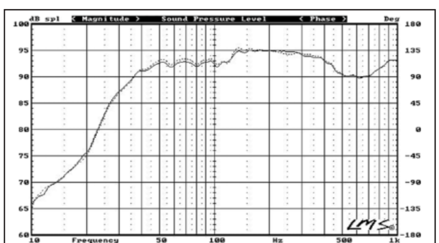

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FIGURE 18: Comparison of system with extra damping material (solid line) and system without extra damping.

there are still significant levels of cabinet edge diffraction.

Figures 9 and 10 are the system's response below and above the tweeter. Below 5kHz the dips in the response curve are due to inter-driver cancellation caused by the varying path lengths between the drivers. Overall, the system exhibits very good dispersion and should deliver a smooth and broad power response to a room.

Individual driver responses are shown in *Fig. 11*. The woofer crosses over to the mid/bass at nearly 400Hz, while the mid/bass crosses over to the tweeter at 2.3kHz. The woofer's response rolls off at a fourth order rate above 300Hz but the mid/bass rolloff is about second order till 200Hz, then it steepens to about a fourth order rate. The asymmetrical nature of the woofer to mid/bass crossover may explain why the woofers must be wired with reverse polarity. The rolloff slopes of the mid/bass to tweeter crossover appear to be symmetrical, however.

In *Fig. 12*, which shows the crossover voltage transfer functions, you can see that the woofer and mid/bass cross over at 220Hz, and this time the woofer has the gentler roll off slope to start with, which then transitions into a steeper slope at 300Hz. The voltage transfer functions for the mid/bass and tweeter are average in damping, in my experience. In other words, I've seen voltage transfer functions with a sharper "knee" and ones with a more gradual "knee." To what extent the shape of the "knee" influences the subjective quality of the speaker is uncertain, as far as I know. Still, I prefer as rounded a "knee" as possible.

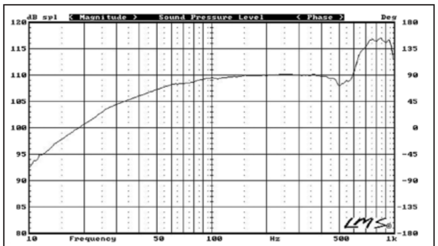

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FIGURE 19: Near-field SPL measurement of 8955 woofer in free air.

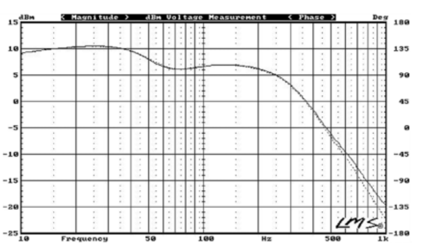

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FIGURE 20: Comparison of woofer crossover voltage transfer function with inductors interacting (solid line) versus inductors not interacting.

CABINET PANELS

While the well-built cabinets look great, they still produce sound when rapped with knuckles. Admittedly, it takes a hard rap to do so, and your knuckles will hurt, but it is still instructive to look at the cabinets with an accelerometer. I attached a Measurement Specialties ACH-01 accelerometer with double-sided carpet tape to the front and then the side of the speaker cabinet.

By the way, this is not the foam type of double-sided tape that is 1⁄16″ thick. This type of tape is very thin and holds the accelerometer very tightly to the panel and provides little if any damping. *Figures 13* and *14* show a primary resonance at 50–60Hz and another series of higher Q resonances at 300–500Hz. The side panel resonance at 400Hz is about 3–4dB higher than the front panel.

Compared to plain, undamped, or unbraced ¾″ medium density fiberboard, these measurements are about 10–15dB lower. These cabinet panel resonances should be low enough so as not to affect the overall sound of these speakers[5]. This expectation is based on a study of cabinet panel resonances that showed SPL of resonances from an untreated ¾″ MDF cabinet panel can be as high as the sound produced from the driver and, thus, affect the direct SPL measurement of the driver. The study showed that resonances from a triple-layer ¾″ MDF cabinet panel were about 10dB less than single-layer MDF, which was why a triple-layer panel did not affect the direct SPL measurement of the driver. So while the 8871 cabinet panels are not vibration free, the level of the vibrations are probably low enough so as not to color the direct sound of the speaker system.

The grille frames for the 8871 are ½″ thick and are covered with a black fabric. How they affect the speaker's response is shown in *Fig. 15*, which is a difference curve of the speaker system with and without the grille. As expected, there is a 5dB variation around 3kHz and 1–2dB bumps and dips above that frequency. Most of the variation is caused by the grille frame, with some effect above 10kHz due to the grille fabric. Since the on-axis response of this speaker has a 2–3dB dip between 3–5kHz, they may actually sound smoother with the grille frame on.

ANALYSIS OF WOOFER DIP

The woofer aberration between 100–150Hz is quite noticeable on SPL and impedance measurements and could be a significant source of subjective coloration. What is causing this problem? Well, it may be caused by a resonance of the woofer magnet and the driver basket. Or, maybe it is a resonance caused by the ports. And since the cabinet is rather tall, it could be caused by an internal standing wave resonance.

To evaluate each of these possibilities, I started with near-field measurement of the woofer and port (*Fig. 16*). The dip at 100–125Hz is still there, and it turns out that the port resonance occurs at nearly the same frequency. However, the port resonance is down by nearly 15dB, so it seems unlikely that this could be causing the dip.

I then stuffed the port with rags and measured the system SPL. This is shown as the solid line in *Fig. 17*. While the response of the woofer changes below 100Hz, the dip at 100–125H remains. So these two observations suggest the port isn't the cause of the aberration.

So could it be an internal standing wave? A half wavelength at 100–125Hz would be about 53–66″. While these speakers are tall, that's about 10–15″ more than the longest internal dimension.

Still, I stuffed another pound of Dacron damping material into the bottom of the speaker and measured by ground plane. The results are shown as the solid line in *Fig. 18*. There is perhaps a difference of a half-decibel but no significant change in the area of interest. So it may be that I didn't have enough material to make a difference or the problem isn't caused by an internal standing wave.

That leaves a magnet/basket resonance as the potential culprit. To find out, I removed the woofer from the cabinet and mounted it on large wooden L-brackets that were attached to the miter gauge slots on my table saw.

Then I measured SPL near-field. *Figure 19* shows no trace of the dip at 100–125Hz. So this near-field measurement seems to rule out the magnet/basket resonance. Also, I mounted the woofer in a sealed box and again there was no dip at 10–125Hz. So it appears that the magnet/basket resonance is not the cause.

I was hoping for a clear-cut explanation of this problem, but the tests were inconclusive. My best guess is that the dip at 100–125Hz may be caused by an internal standing wave. At these frequencies damping materials are not very effective, so it is difficult to attenuate standing wave resonances.

ANALYSIS OF INDUCTOR CROSSTALK

While it is well known that inductors can interact with one another, I was curious to what extent the problem occurs in loudspeakers. I set up a simple experiment by connecting a 0.70mH air core coil to my test amplifier and running 2.83V through it. I placed another 0.70mH air core coil adjacent to the coil with the signal and attached a voltmeter.

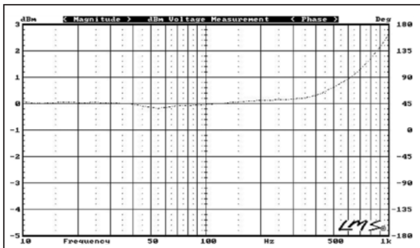
With both coils facing upwards so that their poles were perpendicular to the test bench, the unpowered coil registered 0.057V. By turning one of the coils 90° apart and with the poles of the coils not pointed toward one another, the voltage dropped to zero. With both coils stacked 0.493V was generated in the unpowered coil.

While 0.057V doesn't seem like much, a similar test with a 2.0mH ferrite core coil generated 0.470V. So the level of the induced voltage in an adjacent coil can be quite significant. I also repeated this little experiment with a 4.0mH air core coil and the 0.70mH air core coil. Only 0.025V was generated in the 0.70mH coil, but 0.183V was generated in the 4.0mH coil.

I was a little surprised at these results, considering that an expert had once told me that air core coils generated more crosstalk than cored coils because there was nothing to block the magnetic field. In any case, crosstalk between inductors can vary with the type of inductor and can be significant or modest.

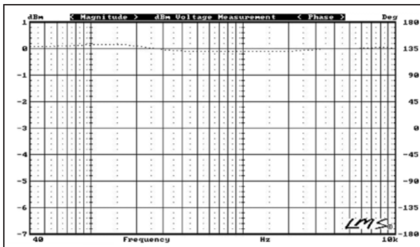
To further evaluate the effect of inductor crosstalk, I reproduced the woofer and midbass crossovers on my prototype boards. I then measured the voltage transfer functions with and without the coils cross talking. In other words, in one instance the coils were next to one another with the same orientation, poles perpendicular to the test bench, and in the other instance one coil was 90° to the other.

For the woofer crossover I lined up a 4.0mH air core, a 0.70mH air core, and a 9mH ferrite core, which is similar to how they appear on the crossover board in the speaker. These aren't the exact values of the 8871 crossover, but are



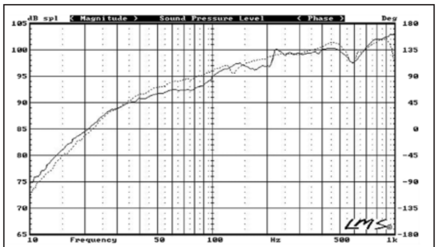
B-2210-21

FIGURE 21: Difference curve of voltage transfer functions in *Fig. 21*. Scale expanded.



B-2210-22

FIGURE 22: Difference curve of mid/bass crossover voltage transfer functions with and without interaction of inductors.



B-2210-23

FIGURE 23: Comparison of ground plane SPL measurement of 8955 woofer in 54″ tall box (solid line) versus 23″ short box.

within 20% of the stated values. To reduce the crosstalk I then turned the 0.70mH coil 90°. The results for the woofer crossover are shown in *Fig. 20*.

Above 150Hz the crosstalk increases the voltage from about a tenth of a dBm to more than 2dBm at 1kHz. Below 150Hz there is little variation; you can see this in *Fig. 21*, which has an expanded dBm scale and shows the difference between the two curves. In this instance, the two air cores were in series with one another and in series with the woofer. The 9mH ferrite core was part of an RLC shunt circuit.

For the midbass crossover, I lined up a 4.0mH coil and a 0.70mH air core. The midbass difference curve (*Fig. 22*) shows a difference of between a tenth to two-tenths of a dBm from 50Hz to 10kHz. In this instance the 0.70mH air core was in series with the midbass and the 4.0mH coil was in parallel.

Thus, it appears, at least in this simple experiment, that inductor crosstalk has a relatively minor effect on the voltage transfer function of a crossover. However, you still must wonder what effect this has in a real situation of playing music through a loudspeaker. Purists would most certainly consider any opportunity for inductor crosstalk to occur to be a less than optimal execution of crossover construction.

SUMMARY & CONCLUSION

From an engineering standpoint, the CP8871 does many things well, but it has several imperfections. I'm sure it wouldn't be presumptuous to say that a speaker of this caliber (and price) should be free of any obvious engineering shortcomings. But I've identified six in the construction of this loudspeaker.

As noted earlier, the wiring to the drivers is not soldered but is terminated with push terminals. This may be a minor point and also a matter of personal preference, but soldering the connections would ensure a more secure and reliable termination.

The inside edge of the mid/bass hole isn't chamfered or relieved. Because the front baffle is quite thick, at 2¼″, and the mid/bass magnet is quite large, the rear opening of the driver basket is seriously impeded. This arrangement could cause anomalies in the midrange response of up to 2dB[6]. I'm not sure how serious a problem this is from a subjective quality standpoint, but it is certainly measurable and does affect the smoothness of the midrange SPL.

And from what I see in advertisements, some loudspeaker designers take great pains to minimize the problem. They use either a midrange with a smaller magnet or a larger basket opening. Or barring the use of a specially designed midrange driver, the rear opening is chamfered or relieved to make the rear opening of the driver hole larger.

Much research has come out in recent years proving beyond a doubt of the superiority of flared ports. While they don't eliminate compression, they substantially reduce port noise and distortion, and speaker systems with them produce more bass. So I was surprised to see plain old cardboard tubes in the CP8871. When you consider that you can pay about $12 retail for a very nice 3″ flared port from Madisound or Meniscus, it seems to be cost effective for a manufacturer to implement them, especially in an expensive loudspeaker system.

The placement of resistor R2 ahead of the capacitor and inductor in the mid/bass crossover compromises power handling, because R2 takes the full bandwidth of the signal. To lower the chance of burning out R2, it might make sense to use a resistor with a higher power rating or to parallel two resistors with twice the value. Also, while I'm not an expert on the subject, the quality of the resistors used does not appear to be of the highest grade. So simply using better quality resistors may noticeably improve the subjective

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sound quality of the speaker.

The inductors on the crossover board are placed too close together and are oriented in the same direction, which is causing interaction with each other. While my simple empirical test showed minor variations in the voltage transfer function of the crossover, eliminating the interaction could improve the sound of the speaker. By simply changing the orientation of the inductor, crosstalk can be significantly reduced.

The woofer's resonance problem at 100–125Hz remains a bit of a conundrum. I'm pretty sure that it is due to an internal standing wave. To further verify this notion, I built a 54″ tall enclosure similar in height to the CP8871 enclosure. Internally, this test box measures 8.5″ wide and 9.5″ deep and it was stuffed with 1½ lb of dacron fiberfill.

I installed one of the 8955 woofers and measured the SPL one meter from the enclosure and with the microphone on the ground. The solid line in *Fig. 23* shows the results. There is a broad dip between 140–200Hz. Compare this result to the dotted line that is the SPL of the woofer in an enclosure measuring 23″ tall by 11″ wide by 10″ deep.

Aside from a small notch at 135Hz the SPL of the woofer in the smaller cabinet is much smoother. This comparison does seem to offer additional proof that the resonance at 100–125Hz is caused by an internal standing wave resonance. The smaller cabinet should also have a standing wave resonance, but it would take place at more than twice the frequency since its largest dimension is less than half the tall enclosure's longest dimension. A dip at the higher frequency is not obvious, maybe because the damping material more readily absorbs the higher frequency.

Thus, it seems that tall enclosures can be problematical because they support standing wave resonances that are low enough in frequency to be difficult to dampen. A probable solution to this problem is to partition the enclosure so as not to have any dimensions longer than, say, 24″.

Also, while not an obvious engineering flaw, the somewhat non-flat on-axis frequency response puzzles me. Either my microphone calibration is way off or perhaps there has been substantial deviation in driver parameters since the initial design. This might suggest that manufacturing quality control might have slipped. Or maybe it was done deliberately since speakers with more bass emphasis are more attractive to some listeners.

Nevertheless, the Compass Dancer CP8871 is an impressive speaker. Its sound is quite good and its appearance is quite appealing. I don't profess to be a "golden ear" but I think it does a very good job in many ways. I've enjoyed listening to these speakers and would gladly keep them.

However, my wife would like to "reclaim" her living room, which is 17′ wide and 20′ long with a 14′ high vaulted ceiling. These speakers tend to visually dominate a room even of this size. Thus, these loudspeakers aren't for everyone, especially if your listening room is on the small side and you don't have an understanding spouse. But the sound fit quite well in our room, so I would highly recommend them if you don't mind their physical size. ❖

XpressMail

OCTOBER ISSUE

So readers aren't confused, there should be "ohm" signs (Ω) after all the "4" and "8" load resistance figures in the first column of page 17 of "Modifying Peavey's TKO 80 Bass Amp" (October 2003).

In Fig. 3a of Ben Poehland's "The SOAP Factor," p. 34, in addition to the two emitter ballast resistors, the lower NPN output transistor also requires a collector resistor to help make the lower common emitter output stage mirror the characteristics of the upper emitter follower.

In "Borbely's RIAA Preamp with Tubes," p. 36, passive RIAA equalization predates the excellent transistorized version in Dick Marsh's *TAA* 3/80 article. There is a passive RIAA EQ circuit using a 7025 twin triode in the RCA Receiving Tube Manual RC-19 (1959), "Preamplifier For Magnetic Phonograph Pickup," p. 364, schematic 19–15. I like Joe Tritschler's use of the PC board for mounting the tube circuitry.

I'm not sure why Joe prefers 5% carbon comp resistors for their sound. I've heard that opinion mainly from guitar players who modify their amplifiers. Even applying his 1/3 derating won't alleviate the 45dB noise index disadvantage over metal film resistors, or the 35dB noise index disadvantage over carbon film types (see "Choosing and Using Electronic Parts: A Survival Guide, Part 2," *aX* 12/01, Fig. 6, page 40). In addition, the inevitable change in resistance of the carbon comp resistors with time, temperature, and humidity will continuously degrade the low RIAA response error he worked so hard to achieve.

Sprague hasn't made the Orange Drop caps since 1968. They were originally introduced in 1959 at the Sprague Barre, Vt., plant, which became SBE (Sprague-Barre Electronics) in 1968 after a management buyout of the plant and tooling. Vishay acquired the remainder of Sprague Electric of North Adams, Mass., in 1992.

The SBE Orange Drop is a family of orange epoxy encapsulated radial lead film caps, comprising both film-foil and metalized polyester, polypropylene, and polystyrene capacitors. To make sure you get the polypropylene types, you must specify the 7xx series part numbers. There are a few SBE 4xx series polyester caps being hawked on eBay as "Sprague polypropylene" caps.

Caveat emptor!

Chuck Hansen
Ocean, N.J.

IMPEDANCE TESTING

In Bill Fitzmaurice's articles he talks about impedence testing of the speakers used. Was there an article I missed on how this is done? If not, how about an article about how to do it? Or is this something that requires a huge batch of test equipment, besides a scope, a DMM, and a signal generator? I suppose some sort of frequency level meter would be needed as well.

I have several drivers that I would like to use in several of Bill's boxes, but I don't know the specs, and can't tell whether they would be appropriate or not without testing.

Thanks for the mag, and keep up the great job.

Will Morrison
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Bill Fitzmaurice responds:

Impedance testing has been covered by myself and others in these pages and in Speaker Builder *on a number of occasions, so you should be able to find an article in the archives. It's also well covered in many speaker building books, David Weems's being one that comes to mind ("Designing, Building and Testing Your Own Speaker System," available from Old Colony Sound Lab). However, as for T/S specs, there is much more involved than impedance testing. The details are also covered in many books, but there well may be an easier way to determine specs for specific drivers. Try logging on to the Audio Asylum site at www.audioasylum.com/forums and put a post on to the Speakers or High Efficiency Speakers forum; chances are good that someone may either have the specs you're looking for or know of a link to where you can find them.*

TUBE AUDIOPHILES

I am a fairly long-term subscriber and sometime contributor to *audioXpress* and to *Glass Audio* before it. I realize that the genesis of the present magazine and its format and content were probably an economic necessity. However, the present format and content does not please me nearly as much as the old *Glass Audio*. I subscribed to *GA* because I am a vacuum tube audio enthusiast and hobbyist, and the present mix of articles holds less interest than the former. Also, I really miss the expanded items for sale columns, from which I had purchased many parts and components in the past.

Having said this, let me add that I will continue to be a subscriber to *aX*, which still holds some interest. I do, however, miss the old *GA*.

Jerry Boncer
Carlsbad, Calif.

Edward T. Dell responds:

Each of the 12 annual issues of audioXpress *contains two* GA *articles. Note the little symbols on each article. That is 24 articles each year.* GA *was a bimonthly with four articles in each issue. You can do the math.* aX *is 72 pages; most of the* GA *issues were far less than that.*

I am sorry you are not happy, but have you noticed that audio magazines are dying left and right? We are the last audio construction print magazine on the planet and will continue to do our very best.

Thank goodness for your magazine! Since subscribing to *Glass Audio* several years ago, I have had more fun and learned more about home-built audio than I thought possible. Now I am in the process of acquiring back issues/CDs as I can afford them. Just wish I had subscribed to all three magazines years ago, as the combined format has opened up a whole new world of speaker building, which was not a big interest before. Please keep up the good work and publish everything on CDs. Thanks!

I am very interested in building a version of Rick Spencer's "A Five-Channel Tube Home Theater Amp" (*aX* 6/02, p. 42). I would like to ask Mr. Spencer if he has tried the 6L6 version mentioned and also what preamp design he uses or recommends.

D. Mark Detrixhe
Fredericksburg, Tex.

Rick Spencer responds:

Thank you for your interest in the five-channel tube amplifier. There have been a few of the amplifiers built using the 6L6 (or 5881) tubes in place of the 12V6s. I built one for a gentleman upon his request for a five-channel tube unit with more power per channel than the 12V6s or 6BQ5s could deliver. He had a good stock of Sylvania 5881s on hand.

I built the amplifier using them with the Hammond #1620 output transformers in place of the #1608s. With a few minor circuit changes and a different Plitron B+ transformer, the amp delivers a warm and full sound. It really has a lot of punch in the bass and midbass ranges and seems to have a good reserve of sonic energy when it handles DVD movies with a heavy rock audio track. On the "Jurassic Park" soundtracks, the T-Rex impact tremors are very deep and solid indeed!

Regarding the type of preamp designs that you can use with the amplifier, some hobbyists have tried a five-channel passive type, which, of course, is the most simple to construct. They are also basically trouble-free, require no power source, and you never have to remember to turn them off or on. Installed between the signal source and the amp, it gives you the ability to control the level of each channel with relative ease, and at a very low cost.

The other preamp design I tried with the amp is the one in Glass Audio *2/91 (p. 1) by Bruce Rozenblit. It works just fine and so does the one by Joseph Still in* aX *3/01 (p. 36). I built the line stage only, and, as with all of Mr. Still's designs, it performs great! If you like to experiment, you can build either of these units using octal type tubes instead of the 9-pin types. The choice is yours.*

I wish you the best in your endeavor to build your own version of the five-channel tube amp, and, if I can help you out by furnishing an upgraded parts list and schematic diagram, let me know. By the way, I am very pleased to hear that you are having fun learning about building your own audio equipment! We hobbyists are all very fortunate to have a fine magazine like audioXpress *around to inspire us!*

SUB REQUIREMENTS

Reading the article "Servo Dual Voice Coil Subwoofer" (*aX* 11/03, p. 18), I was particularly interested in the aspect of acceleration and velocity, which I have never found in any other article or design. As explained, it made perfect sense to me and I believe this to be the right approach.

In order to build such a subwoofer could I use the Adire DPL 12, which happened to be advertised in the same issue of *audioXpress*? This driver has very similar parameters as the Peerless mentioned in the article. Mr. Ferguson mentioned that in order to get a reasonable sound level, you must use at least a 12″ driver. I also understand that the smaller one can make the box, the better, from a distortion point of view. Is this correct? Is it also correct to say that the smaller the box the less SPL?

For your information, I play mostly classical music in a normal-size room with a multi-amplified setup with battery-driven French 8W monster amplifiers and Scan-Speak speakers. Classical music requires less power than some modern music, but the sound level of Mahler, for instance, with a big orchestra, is pretty impressive. That is the reason I believe your subwoofer should improve my system.

M.G. van Lanschot
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Daniel Ferguson responds:

To summarize, you are considering building a closed-box servo subwoofer using the Adire Audio dual voice coil DPL12 woofer and are asking whether or not this driver is suitable. You also stated that, as you understood it, the smaller the box size the better from a distortion standpoint.

The driver in question has Thiele-Small parameters which are essentially perfect for a normal closed box application—Qts = 0.4, Fs = 16.3, Xmax = 14.3mm. These are measured with both *voice coils driven. With one voice coil driven, as used in my servo design, the driver Q doubles, which in this case will make it 0.8. This means that when the driver is put in any normal-size box, the Q will already be well above 1. For example, a 3.0ft^3 box will result in a Q of 1.33. The box size that would simulate the low distortion conditions in the last experiment in my article would be about 1.3ft^3, corresponding to a Q of about 1.8.*

Before going any further, I'd like to address the point about "the smaller the box size, the better." It does appear that a stiffer air spring makes the driver cone behave more like a pure mass-spring system. However at this point, I don't really know what the optimum box size is.

After looking at the numbers, I'd have to say that the Adire DPL12 would probably work fine with one voice coil driven and the other used as a velocity sensor. If I were experimenting with this driver, I think I would start with a box size of about 2ft^3 and work my way down. Keep in mind that smaller box sizes equal reduced efficiencies, so more amplifier power is required to develop the same sound pressure level. For your application I would recommend a high–quality amplifier with a minimum output of 200W into 8Ω, which appears to be the impedance of one DPL12 voice coil.

Before closing, a word of caution is probably in order. You must carefully adjust the design I presented by trial and error to get the best combination of loop gain, current, and velocity feedback and stability. You will need some test equipment to do this. At a minimum, I recommend a signal generator, oscilloscope, and some type of SPL frequen-

cy response measurement equipment. Without these it would be very difficult to troubleshoot the system. The good news is that if the servo cannot be made to work to your satisfaction, with both voice coils driven, the DPL12 should make a state-of-the-art closed-box woofer.

Best of luck with your project. I would be greatly interested in hearing how it turns out.

Is this closed-loop concept described in your article applicable to other enclosures such as the bass–reflex or the bandpass, or only to that with closed enclosures?

Philippe Polyte
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Daniel Ferguson responds:

Since any servo can only control what it can measure, I can't envision an application for a servo where the woofer cone is not the sole sound source. For example, in a correctly constructed bass-reflex system, the woofer essentially stops moving at the box frequency. Distortion levels near the box frequency are then dependent on port design since the driver is essentially "out of the picture." This would also seem to be the case for bandpass systems, as port noise may be their biggest negative and it is clearly not influenced by the driver.

Thanks for your interest in my article.

RIAA CHARACTERISTICS

My interest was aroused by Joe Tritschler's query as to RIAA requiring two first-order transfer functions ("Borbely's RIAA Preamp with Tubes," *aX* 10/03, p. 36). I checked in *The Gramophone Handbook* published in 1956, two years after the RIAA standard was determined. An EMI advertisement in the book simply stated: "The replay characteristic for micro-grove recording is the sum of a 75 microsecond top fall, a 318 microsecond bass rise and a 3180 microsecond bass fall."

In the text no mention was made of a requirement for two turnover points. It noted, however, that Columbia, RCA, Decca, HMV, Nixa, and some national standards were already using, or calling for, turnover points at either 1.6kHz or 2.5kHz in the treble and 500+ or "100Hz in the bass." Playing existing LPs through hi-fi amplifiers with only RIAA compensation might have sounded odd without a flat midrange.

But the authors of the RIAA characteristic may have also been thinking about the majority of reproducers in use at that time. Few of these provided much information below 120Hz or above 5kHz, hi-fi was in its infancy, and 500 to 2500Hz was a much bigger chunk of the available spectrum than it seems today. Most people were still used to the sound of acoustic phonographs (down 15dB at 200Hz, down 5dB at 3kHz and with a hump of about 10db between 1 and 2kHz), so, even with rudimentary or no compensation, recordings made with RIAA characteristics should sound better.

On the other hand, almost all cheap reproducers at that time employed piezoelectric cartridges, which, with constant velocity recording, gave an electrical output falling, with increasing frequency, at 5dB per octave. Why a straight line characteristic with the inverse of this rate was not chosen is then difficult to answer.

G. Dunlop
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Joe Tritschler responds:

Thank you, Mr. Dunlop, for your extremely interesting letter. I guess I keep forgetting that ruler-flat amplitude and phase response from DC to light is a fairly modern infatuation. I've also heard that many 78-RPM cutting heads had an inherent attenuation of response below 500Hz, such as the ubiquitous Presto units. Furthermore, I seem to recall my 1980 National Semiconductor Audio/Radio Handbook (available from Old Colony Sound Lab, 1-888-924-9465, custserv@audioXpress.com, BKAA59, $14.95) pointing to the problem of managing the 60dB of dynamic range needed to cover a falling response from 20Hz to 20kHz. While this seems like no sweat today, I'm sure things were different in the mid '50s, especially with commercial electronics.

On a different note, several sharp-eyed readers have pointed out that the parts list given on page 39 is miserably inaccurate. I personally drafted and quadruple-checked the schematic, so it's correct, by golly, but the parts list is a holdover from an earlier version. My apologies if I unintentionally confused anybody (corrected version follows).

I would also like to offer my sincerest apologies to all the people patiently waiting on the status of the Tritschler Precision Audio TPA-308 phono preamp kit. Just as we were poised and ready for production, we got a double-whammy of snafu—first from our web server and then from our primary parts supplier.

We hope to be fully back in action in a few short weeks, so again, sorry for this rather embarrassing ordeal. On the plus side, apparently a big slew of 5842 preamps have already been built or are in the works. Thanks a million, folks.

TABLE 1
UPDATED TRITSCHLER PARTS LIST

COMPONENT	DESCRIPTION
R_1	See text
R_2, R_9	100Ω
R_3, R_{10}	22kΩ, 5W
R_4	22kΩ
R_5	1kΩ
R_6, R_8	220kΩ
R_7	2.2MΩ
R_{11}	10MΩ
C_1, C_5	2200µF, 10V
C_2	3.3nF
C_3	18nF
C_4	1.45nF
C_6	2.2µF
V_1, V_2	5842
K_1, K_2	RCA/phono

MORE TRANSFORMER USES

In response to Peter Buddee's article entitled "Audio Uses of Transformers" (May '03 *aX*, p. 50), we attempt to answer, add to, and explain the different types of transformers in use today. To respond to the list in order:

1. The moving-coil step-up transformer
The usual impedance of an MC-type cartridge is about 20Ω due to the small size and weight of the moving coil. Since the impedance is low, so is the voltage level. The MC step-up transforms the ratio at 20Ω to 20k to 47kΩ output, resulting in a higher voltage to a grid which is a voltage device. The tube now has a workable, more desirable voltage, so its resulting noise floor will be lower and output much higher. This type is not dependent on proper termination, because the grid voltage is more important.

2. Line-level preamp input transformer
This type is used when a signal source requires a balanced input to a single grid. Hum isolation is another use as well as a step-up to increase the input level if needed. This is easily done at this position without a Miller effect problem, but this position is prone to distortion if not terminated properly.

3. Preamp output transformer
This type is very important for properly distributing signal level to another device. The ratio for this position is important to keep high-frequency line capacity losses to a minimum. Isolation is offered as well. Termination is not necessary because line amp tube plate resistance will determine the output source Z. The low Z output voltage is important. Proper output voltage levels are needed to operate amplifiers; termination is only necessary for line matching devices or attenuation.

4. Line-level "problem solver" transformer
Ground loop eliminator.

5. Power amp input transformer
This type is not too common and could be troublesome. The usual isolation, balance/unbalanced, is accomplished, but termination is necessary here at the grid due to higher level, transformer capacity, and Miller effect to ensure low distortion.

6. Power amp interstage transformer
This type is useful for increased amplifier efficiency. A full-range interstage offers a low DCR path (secondary) for the output tube signal grid current. This shifts the operating point so that more power can be delivered in Class A from the output tube(s). This also increases the amplification factor of the driver tube eliminating the caloric load (plate resistor).

7. Power amp output transformer
The most important device in the tube amp, it is responsible for the work the amp must do, matching the load to the output tube(s) so that maximum power and bandwidth is transferred. It must accomplish this with the dynamics the tube(s) impose and the argument the speaker gives back. The output transformer must be terminated properly. The iron will do its job below 2kHz, and the coil design will do the rest of the spectrum.

Full bandwidth outputs are a discipline of many factors. Rolloff on the highs and the lows are the speed limits of a design. Proper application of output tube load impedance, primary current, and power capability are the main decisions.

8. The plate (anode) choke
This device is not a load but a reactor. A load is the working device; a plate choke or reactor is only for offering operating current and voltage for the tube and isolation for the load. A reactor offers an impedance over a given frequency so that the power supply will not see the signal.

Reactance is the expression of self inductance of the inductor. The signal rides on it to the load (grid, output trans grid resistor). This is like a transformer only as a low DCR path for operational current. You must calculate the application of this device for the proper amount of inductance to the given current flow through it and its added gradient capacity. These parameters must be met to offer a proper bandwidth result.

9. The grid choke
This device must be a very high inductance (1000H or so) due to little or no current through it so a resulting high winding capacity is developed and rolloff of the high frequencies will occur. Two or more series chokes would be a better choice, but this device is somewhat redundant when interstage transformers that will do the job much more efficiently are available.

10. Power supply filter chokes
The proper value filter choke stores current for a given time and hands it back with a resulting waveform that the filter capacitor then takes over. The filter choke in combination with the filter capacitors chops off the mountain to fill the valley. The result is no hills or a clean voltage. The filter choke inductance value is determined by the current through it, as with the reactor described previously. This value is based on 120Hz. The more current through the choke the less inductance needed. The DCR of this device is a result of the proper wind, wire gauge, and core area selection.

11. Power mains transformers
The power transformer supplies the power that moves the speaker with the

help of the output tubes. It must offer proper voltages and current to operate all that it's connected to. The main consideration of this device is to supply all power to each active device without heating up too much. It is the workhorse of the amp and care must be taken when selecting each winding.

Just like matching the impedance of an output winding to the speaker, the winding voltage/current must match the device it is powering. Five volts at 2A must go to a like filament, but if it is connected to a 5V at 3A device, the 5V will load down to 4V to deliver the 3A. This will increase the primary current (heat goes up) because it just lowered the primary inductance more than it was designed to do.

The primary must deliver the full VA or power for all the secondaries. Each secondary when loaded will lower the primary inductance by its ratio and the primary will draw current accordingly. When all secondaries are loaded properly, the primary should be at its maximum current flow with moderate heating.

Application for class A operation is 100% duty for a transformer. Older transformers had a 70% duty because class B or AB operation was popular at that time. The sound quality result of this device improperly applied would be a disaster.

Jack Elliano
Electra-Print Audio Co.

Peter Buddee responds:

Not being able to compete with the number of years Jack has spent dealing with audio transformers, I enjoy the fact that we seem to share an appreciation of what (correctly applied) components of this species can achieve. I can only hope that our wait for practical (application-oriented) feedback on our inquiries will not be too long, since I believe there is much to investigate and report on within this field. As mentioned previously, even a very busy group of audio amateurs can spend only a certain number of hours during a year investigating things, so involving a number of groups and individuals should result in more information being gathered.

I was reminded recently of the value of sharing the efforts of others when I spotted David Davenport's interesting article on a transformer-coupled preamp in audioXpress *("Odyssey of a Line Stage," Feb. and March '03). For a long, long time, I have believed that capacitors in the signal path were a no-no, but perhaps it is now time to reconsider this opinion.*

From these (Scandinavian) shores we can report on some recent gains regarding the use of moving-coil step-up devices. We have used this kind of device for decades, achieving some (very necessary) voltage gain, not thinking much about how this came about.

But now, we realized there is also an impedance transformation taking place, to the square of the turns ratio. With this influencing the frequency balance of the chain cartridge-transformer-phono preamp input, we had one of the members of our group using an MC step-up transformer with selectable turns ratios (the highly regarded big Audio Note unit made in Japan). Trying a number of turns ratios for a couple of different cartridges (with different source impedances) showed surprisingly large variations in tonal balance, with a brighter sound for a higher impedance load of a given cartridge, and a much darker sound for a lower impedance load of the same cartridge.

OK, so you have a clear difference in sound, depending on how you tweak the impedance matching. (The resulting difference in sound level needs to be compensated for, of course.) Now, how do you decide what is the "most correct" tonal balance? This is much more tricky, of course, but the preliminary tests showed an average preference for a load (at the phono preamp input) in the neighborhood of ten times the (transformed) source impedance of the cartridges used in these tests.

For example, think of a transformer turns ratio of 1:10, giving an impedance ratio of 1:100. Hence, a 50Ω (source impedance) cartridge would give a (transformed) source impedance of 5000Ω, being 1/10 of the standard 47kΩ phono preamp input. (Seeing things the other way, for the same chain, starting at the 47kΩ input, a 100 to 1 impedance ratio would make the cartridge see a load of approximately 500Ω, still giving the same 1 to 10 ratio.)

Even if this shows a surprisingly good correlation to the old electrical engineering "rule of thumb," be careful not to make any quick conclusions. The important issues here are that the impedance matching of (moving coil) cartridges influences the sound more than we ever expected, that testing your own preferences for tonal balance is an educational experience in itself, and that a transformer solution gives (presumably?) a valid opportunity to test impedance matching without changing any other parameters.

I invite any comments on these results, or on any other type of application involving transformers.

Peter Buddee
Stockholm, Sweden

CUSTOMER SERVICE

I read the letter from Robert K. LeBeck, Jr., in *aX* 10/03 with great sympathy. It becomes harder every day to get parts or information on equipment. He should try www.Tritronicsinc.com (orders 1-800-365-8030 or research 305-639-9991). They handle Philips manuals. They have done research for me, even when I just needed a few TV parts. Possibly, the SACD-1000 is "proprietary" or a very rare product and there is no service manual released past the service center, but it's worth a try.

John Drewski
Rochester, N.Y.

Robert K. LeBeck, Jr. responds:

Thank you. I immediately went to the Tritronics website (www.tritronicsinc.com) and did a search for items related to my subject of interest, the Philips SACD1000 player (SACD/DVD-VCC). The result was "item not found."

After examining the site a little more and remembering the "research" mentioned by Mr. Drewski in his letter, I discovered the research request option on the website and entered a query for a "service manual" for the SACD1000 unit. I received a reply the next day (or sooner?), which indicated a part number for the item I was requesting—Part # CD-SM1997, special order, $50.

I then exchanged a few more e-mails and discovered that the item was said to be a "service manual" on CD-ROM (part # CD-SM1997), with a nominal price of $50, so I thanked the researcher, saying that I would go ahead and order it.

I ordered the CD-ROM online, and it eventually arrived by Airborne Express. My initial examination of the contents indicate that it is, indeed, the information I'd been hoping for!

I thus owe reader John Drewski much thanks for his valuable information on this source of useful technical information. ❖

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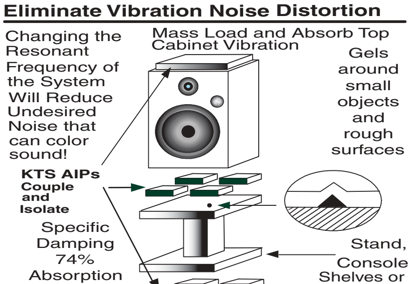

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