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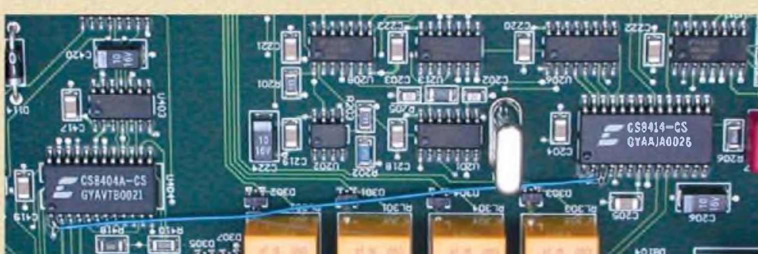

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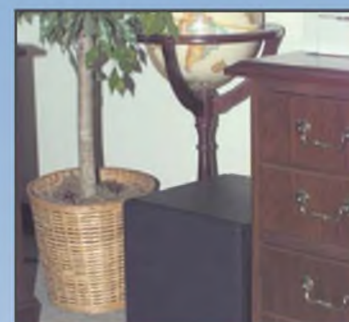

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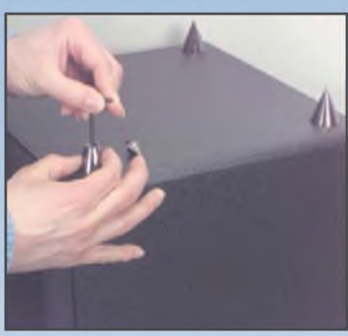

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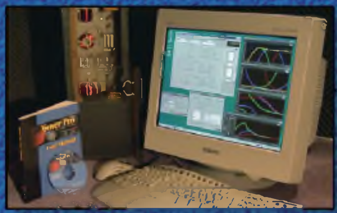

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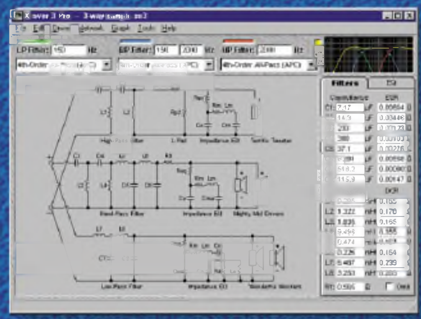

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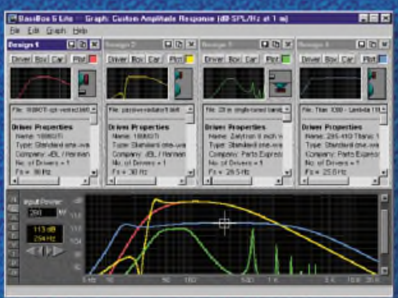

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

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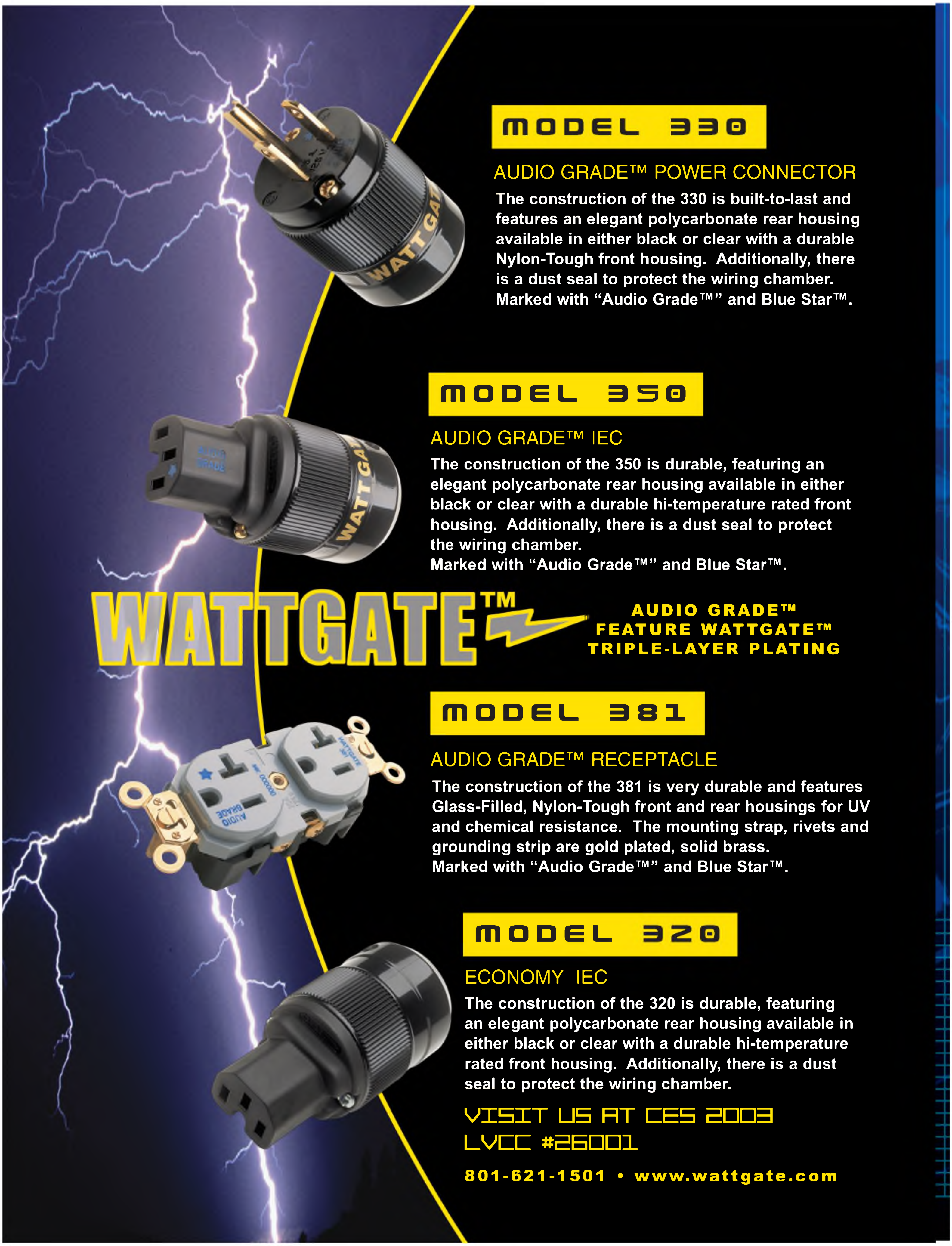
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EMINENCE CATALOG

Eminence recently introduced a new 46-page product catalog and redesigned website. Both resources include detailed information on the Genuine Eminence product line, as well as the new Eminence Magnum® and DeltaLite™ series. Not only has the entire product line received a new look; most of the products exhibit improved power-handling characteristics and expanded performance data. Eminence has developed sections entitled Anatomy of a Loudspeaker, Understanding Loudspeaker Data, Speaker Selection Guide, and Sensitivity Guide, all available in the catalog and on the website. The new website also includes downloadable .pdf files of each Eminence product as well as forms for setting up dealers, distributors, and manufacturers as Eminence customers. Eminence also added a Virtual Plant Tour, Product Reviews, Testimonials, and an Endorsing Artists Page. Finally, there is a complete section devoted to customers requiring custom designs and engineering services. For more, go to www.eminence.com.

The LP Terminator

This article presents the theoretical background and method to properly terminate a magnetic phonograph cartridge. **By Raymond A. Futrell**

A method for terminating a magnetic phonograph cartridge might seem irrelevant since the compact disk player has practically replaced the phonograph. However, some people have large record collections—with no intention of replacing them with CDs—and still use a phonograph. This article is for the people who still play records and want the best possible sound, and for those who like to discover a simple engineering explanation for a baffling problem.

BACKGROUND

In 1975 my audio system included a Thorens turntable with an Empire cartridge and a Revox integrated amplifier. When I upgraded to the highly recommended Shure V15® cartridge, the results were very disappointing. The Shure cartridge sounded terrible, with a dull midrange and shrill violins. Nothing that I tried improved the sound appreciably, and I finally gave up in frustration. It took me several years to realize that the same filter theory I used to design RF filters also applied to magnetic phonograph cartridges.

I assume the phonograph cartridge and preamp can be modeled as the simple lumped element circuit shown in *Fig. 1*[1]. Eg is the voltage generated in the cartridge, Rg is the cartridge DC resistance, L is the cartridge inductance, C is the total shunt capacity in the cartridge, tonearm, interconnects, and preamplifier. R_L is the load resistor in the preamp, nominally 47kΩ, and E_L is the voltage across R_L. The RIAA preamplifier has infinite input impedance and looks like an open circuit.

LOW-PASS FILTER

The two reactive branches of this circuit form an "L" section two-pole low-pass filter. Poles refer to the elements that cause phase lag, in this case the series L and shunt C. The analysis of this circuit isn't too complex if you consider it in two stages.

First, analyze it at DC. With a DC voltage, L is zero impedance, a short circuit, and C is infinite impedance, an open circuit. The circuit is simply a resistive voltage divider and has the following DC transfer function, which is the ratio of the output voltage to the input voltage.

$$\frac{E_L}{E_g} = \frac{R_L}{R_g + R_L}$$

Second, with AC, you must also take the reactive elements into account and include the series and shunt reactance with the resistors. You then have the following transfer function:

$$\frac{E_L}{E_g} = \frac{R_L//X_c}{R_g + X_L + R_L//X_c}$$

X_c is the capacitive reactance and X_L is the inductive reactance. $R_L//X_c$ means the parallel combination of R_L and X_c. The capacitive reactance is $X_c = 1/SC$, the inductive reactance is $X_L = SL$, and $S = j\omega$, the complex frequency in radians per second. After these substitu-

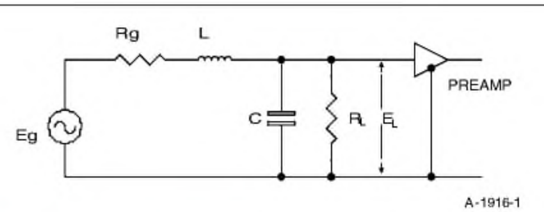

FIGURE 1: Phonograph cartridge model.

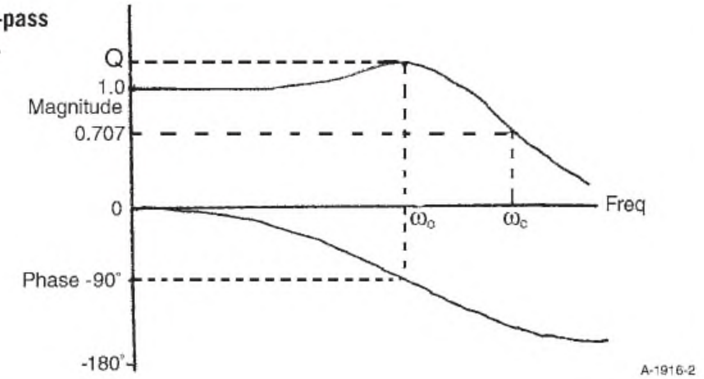

FIGURE 2: Low-pass filter response.

ABOUT THE AUTHOR

Raymond A. Futrell has been in engineering for about 40 years. He has an MSEE and works on military standards and specifications as a contractor to the Army Materiel Command. As system engineer he has designed electrical and communication systems for vehicles, aircraft, and ships, and also designed antennas, amplifiers, and power supplies. He worked world wide as a field engineer and troubleshooter. Futrell's hobbies include audio and reading.

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tions and some algebra, you derive the complete transfer function:

$$\frac{E_L}{E_g} = \frac{\frac{1}{LC}}{S^2 + S\left(\frac{1}{R_L C} + \frac{R_g}{L}\right) + \left(\frac{R_L + R_g}{R_L}\right)\frac{1}{LC}}$$

This is a low-pass filter transfer function that describes the filter magnitude and phase response with respect to frequency. This may not look too useful, but the transfer function and some basic filter theory are all that is needed to properly terminate a magnetic phonograph cartridge.

TABLE 1
Q VERSUS ω_C

Q	ω_C
0.533	$0.704\omega_0$
0.707	$1.000\omega_0$
1.000	$1.272\omega_0$
2.000	$1.498\omega_0$
4.000	$1.537\omega_0$
8.000	$1.549\omega_0$
100.000	$1.554\omega_0$
1000.000	$1.554\omega_0$

THEORY

Now, let's review some basic filter theory. The canonical (standard) form of a two-pole low-pass transfer function is as follows:

$$\frac{E_L}{E_g} = \frac{H_0}{S^2 + S\frac{\omega_0}{Q} + \omega_0^2}$$

The coefficients of the transfer function are gains, frequencies, and time constants (inverse of frequency). H_0 is a gain term, Q is the quality factor, and ω_0 is the undamped resonant frequency defined as $1/\sqrt{LC}$ radians per second. You can select values for these coefficients to obtain different magnitude and phase versus frequency responses.

There are, however, two things you cannot change. As the frequency becomes much greater than ω_0, the magnitude response will decrease at –40dB per decade and the phase lag will approach –180°.

To normalize the transfer function to have unity gain at DC ($\omega = 0$), set S equal to zero and evaluate the transfer function. You obtain this result:

$$\frac{E_L}{E_g} = \frac{H_0}{\omega_0^2}$$

For unity gain at DC, $H_0 = \omega_0^2$, so make that substitution in the transfer function. Now evaluate the transfer function at ω_0, the undamped resonant frequency. You just set $S = j\omega_0$ in the transfer function and obtain the following result:

$$\frac{E_L}{E_g} = \frac{\omega_0^2}{-\omega_0^2 + j\frac{\omega_0^2}{Q} + \omega_0^2} = -jQ = Q\angle -90°$$

This shows that at ω_0 the magnitude and phase of the transfer function is Q and –90°. In a low-pass filter, the peak in the magnitude response is directly proportional to the Q. This is shown graphically in *Fig. 2*.

The cutoff frequency (bandwidth) is indicated by ω_c. By definition, the cutoff frequency is where the magnitude re-

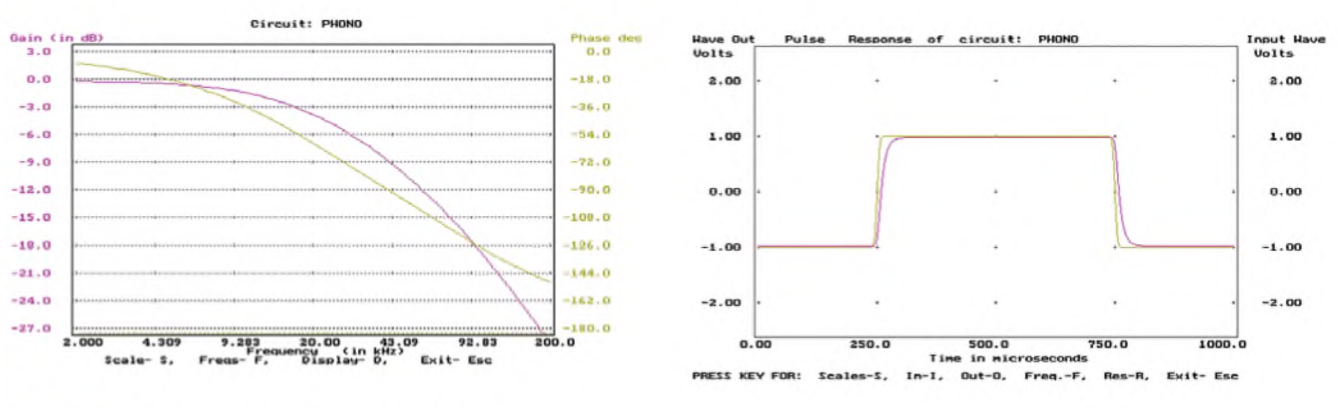

FIGURE 3: Simulation results for 31pF capacitor.

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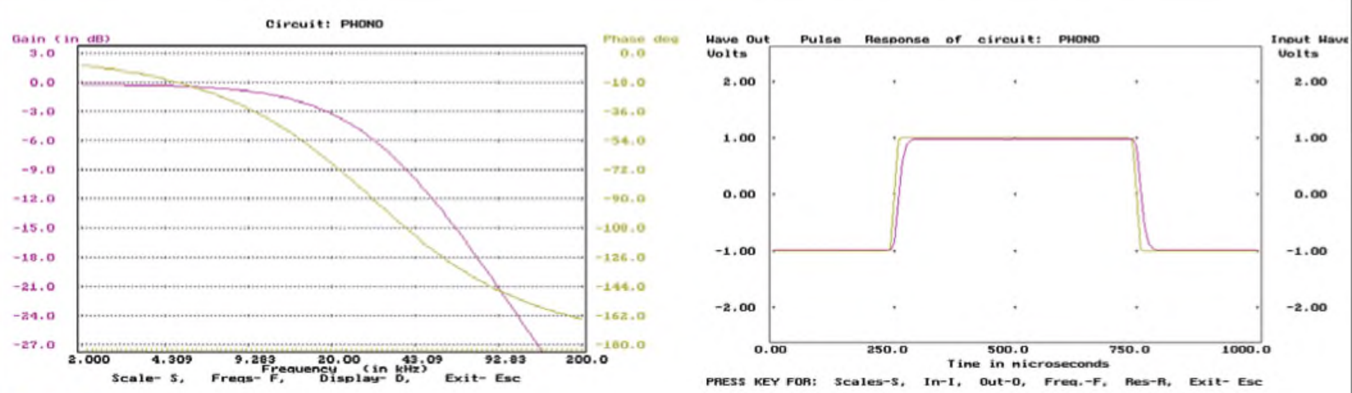

FIGURE 4: Simulation results for 62pF capacitor.

A-1916-4

sponse has decreased to 0.707 of the low-frequency response. In theory, finding the cutoff frequency is simple. You set the transfer function equal to 0.707, replace S with $j\omega_c$, and solve for ω_c.

In practice, the theory is wrong. Some tricky algebra is required to find the cutoff frequency, so I'll just present the first step of the derivation, and the formula for ω_c. The first step is to convert the transfer function to magnitude form.

$$0.707 = \frac{\omega_0^2}{\sqrt{\frac{\omega_0^2 \times \omega_c^2}{Q^2} + \left(\omega_0^2 - \omega_c^2\right)^2}}$$

After considerable algebra,[2] you derive the formula for ω_c.

$$\omega_c = \omega_0 \sqrt{\left(1 - \frac{1}{2Q^2}\right) + \sqrt{1 + \left(1 - \frac{1}{2Q^2}\right)^2}}$$

For your convenience, I have listed some calculated values in *Table 1*.

Note that large increases in Q do not result in large increases in ω_c (bandwidth). The Q does not determine the bandwidth of a low-pass filter; it determines the peak in the magnitude response.

From filter theory you know that the Q of the L-C section completely determines the magnitude and phase response of a single section filter. For a maximally flat phase response (Bessel filter) the Q = 0.533, and for a maximally flat magnitude response (Butterworth filter) the Q = 0.707. As the Q increases beyond 0.707 the magnitude response develops a peak (Chebyshev filter), and the phase response becomes more nonlinear.

The formula for Q is obtained by equating the second denominator coefficient of the canonical transfer function and the derived transfer function and solving for Q.

$$\frac{\omega_0}{Q} = \frac{1}{R_L C} + \frac{R_g}{L}$$

After some algebraic manipulation and substitution,[3] you obtain the formula for Q.

$$Q = \frac{R_L\sqrt{\frac{C}{L}}}{1.0 + R_g \times R_L \times \frac{C}{L}}$$

This formula seems too complicated to easily calculate the Q, but you can evaluate the denominator with R_g = 1k, R_L = 47k, L = 0H5, and C = 100P. With these values, the denominator is approximately 1.009, so round it to 1.0 and ignore it. That removes the complication and causes negligible error. You then have the approximate formula for Q.

$$Q \approx R_L\sqrt{\frac{C}{L}}$$

COMING TOGETHER

This is the formula for the Q of a parallel resonant R-L-C circuit. After all that algebra, you discover that the phono-

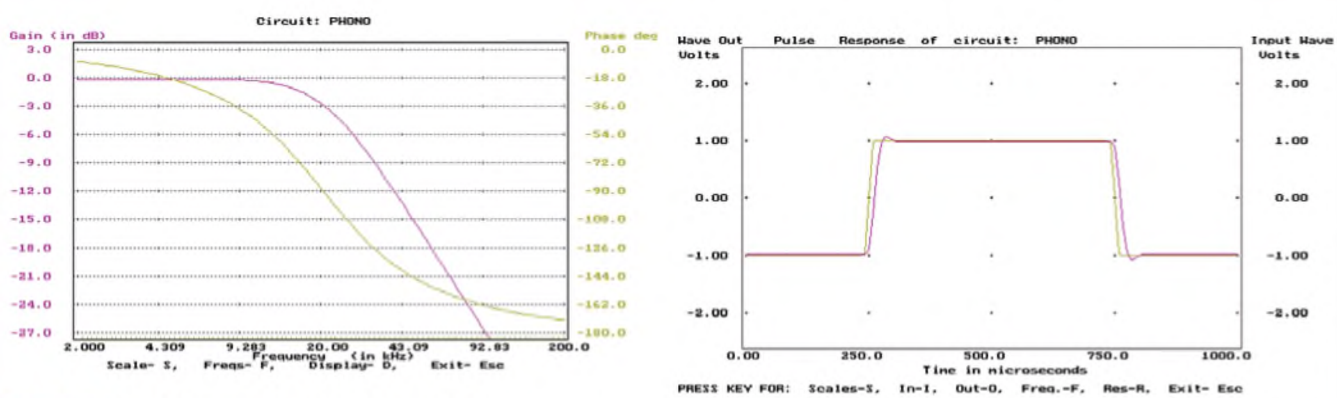

FIGURE 5: Simulation results for 125pF capacitor.

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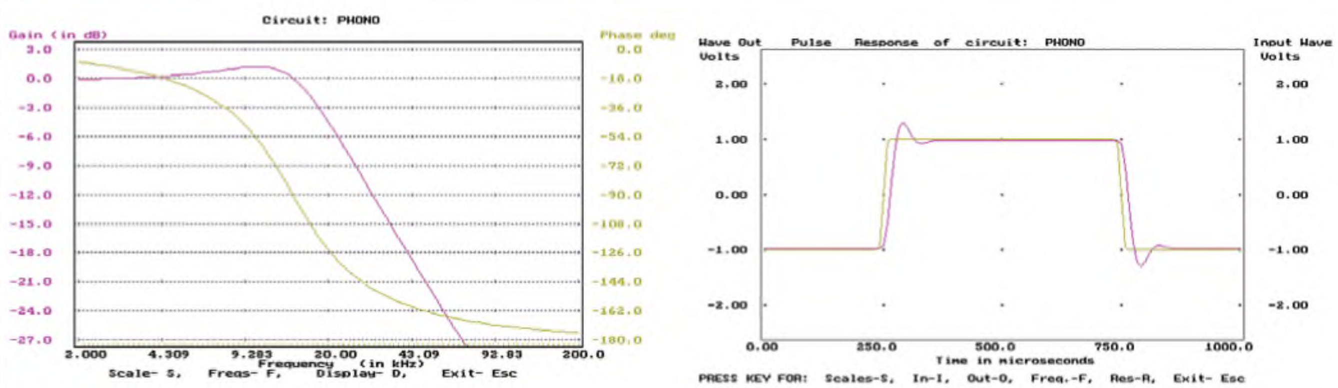

FIGURE 6: Simulation results for 250pF capacitor.

A-1916-6

graph cartridge is mathematically just a simple resonant circuit. Now you can find out what value of capacitor is required for a Q of 0.533 and 0.707 with a Shure V-15® cartridge using L = 0H5 and R_L = 47k. Rearrange the formula for Q to calculate capacitance.

$$C = L\left(\frac{Q}{R_L}\right)^2$$

The calculated value of C is 64pF for a Q of 0.533 and a Bessel filter (maximally flat phase), and 113pF for a Q of 0.707 and a Butterworth filter (maximally flat magnitude). Now that you have Q, L, and C, you can calculate ω_0 and ω_c. The calculated cutoff frequency is 19.8kHz for a Bessel filter and 21.2kHz for a Butterworth filter. These simple calculations are all that is required to properly terminate a magnetic phonograph cartridge.

When I first performed this calculation I was skeptical of these theoretical results. The values are less than half of the 250pF terminating capacity recommended by Shure, and I also believed that a few hundred pF of shunt capacity in a moderate impedance audio circuit had negligible effect. I didn't have a circuit analysis program to simulate the circuit, so I verified the theory with an experiment.

I bought a low capacity phono cable and a Leach JFET preamp[4], which has about 20pF of input capacity. I cut off a short piece of the phono cable and sol-

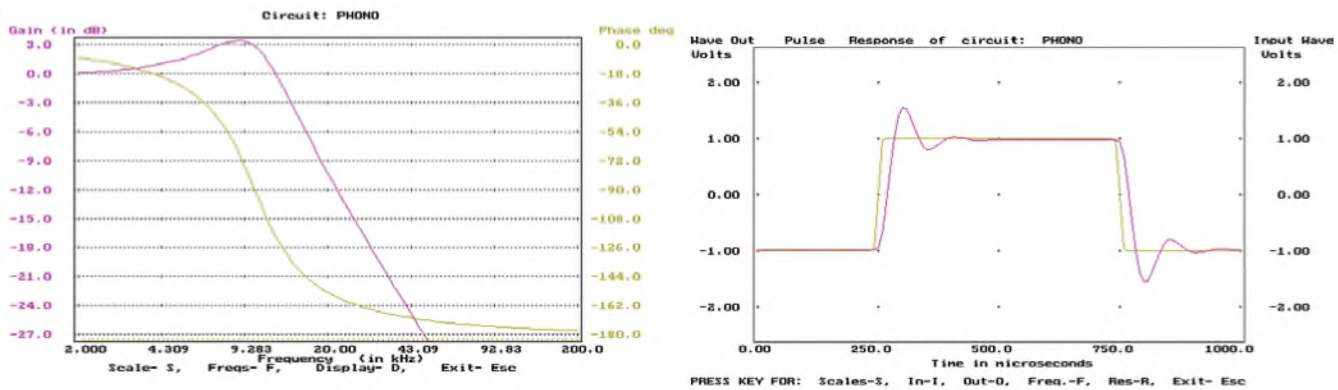

FIGURE 7: Simulation results for 500pF capacitor.

A-1916-7

dered the wire ends directly to the preamp inputs. The preamp came on a 3 × 3″ circuit card that I mounted in the base of the turntable under the tonearm. I estimated this configuration would result in about 100pF of shunt capacity.

When I did comparative listening tests, the improvement in the sound was dramatic, not subtle. The dull midrange and shrill violins went away. The Shure cartridge sounded fine. The improvement was so striking that one person commented, "You bought a new cartridge."

The bad sound was caused by too much shunt capacity. The phono cable had 250pF of capacity (manufacturer specification), and assuming 25pF winding capacity in the cartridge, 25pF in the tonearm wires, and 200pF in the Revox amplifier, that's 500pF of capacity. The Revox preamp inputs, of course, were shunted with 100pF capacitors to prevent radio frequency interference. That was standard practice[5]. I aggravated the problem by trying to correct it with more shunt capacitors. It's not surprising the sound never improved.

To demonstrate the effect of the shunt capacity, I ran the circuit in *Fig. 1* in a circuit analysis program with Rg = 1kΩ, R_L = 47kΩ, L = 0H5, and several values of capacity. Notice how the magnitude, phase, and transient response (1kHz square wave) worsen for the larger values of capacity. *Figures 3–7* show the results, as C is incremented from 31pF to 500pF.

Clearly a few hundred pF of shunt capacity in a moderate impedance audio circuit has a significant effect. This is consistent with filter theory. When the terminating capacitance (or resistance) changes, the filter response changes. I believe this is one reason for the audible difference between interconnects, preamps, and so on.

Filter theory shows that you can't pick the cartridge terminating resistance and capacitance arbitrarily. They must be commensurate with the Q and bandwidth. This applies to any inductive signal source such as a magnetic phonograph cartridge, dynamic microphone, or transformer. ❖

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1. National Semiconductor Corporation, *Audio/Radio Handbook*, 1980. Section A5.0, p. 6–13, "Magnetic Phono Cartridge Noise Analysis." The author, Mr. Maxwell, uses the same lumped element model of a phonograph cartridge. This noise analysis shows how he calculated the noise of a phonograph cartridge in his design of a phonograph preamp in *Electronic Design*, Feb. 16, 1976, p. 146, "Hold Noise Down With JFETs..."

2. Ibid. Section 2.15, p. 2–55, "Scratch, Rumble and Speech Filters." Basic information on audio filters.

3. F. Langford-Smith, *Radiotron Designers Handbook*, 1952, p. 411, section 4. See Fig. 9.4 and the formulas (22) for the Q of a tuned circuit. The Q formula in this article is mathematically identical to these formulas.

4. Professor W. Marshall Leach has posted the JFET preamp schematic on his WebPages at the Georgia Institute of Technology website, http://users.ece.gatech.edu/~mleach/. Dr. Leach has information on amplifiers and filters here.

5. National Semiconductor Corporation, *Audio/Radio Handbook*, 1980. Section 2.4, p. 2–11, "Audio Rectification."

The DR15a Horn

This series of Double Reverse (DR) horn designs now addresses the needs of owners of 15″ drivers.

By Bill Fitzmaurice

PHOTO 1: The DR15a horn.

It's finally here. Many readers have petitioned me for quite some time now for a Double Reverse (DR) horn to house a 15″ woofer, and the DR15a (*Photo 1*) is it. Designed as a backline bass or keyboard cab, or as a main PA in large (up to 1,000-seat) venues, it is more efficient with a wider bandwidth (*Fig. 1*) than any other single 15″ cabinet, flat out blowing away commercial cabs costing thousands of dollars. You can build it with a premium woofer for less than $400, or with a mid-line driver for less than $200. But all things considered—as with its cousin the DR12a (Oct. '02 *aX*)—I suggest that you don't build this box. As good as it is, you can do better.

HORN-LOADED VS. DIRECT RADIATOR

Let me explain. Those of you who have written me asking for a DR for a 15—as well as for an 18—have done so under the mistaken assumption that it would play louder and to lower frequencies than a DR loaded with a 10 or 12. That assumption is rooted in the obsolete technology of commercial speakers—for the most part either direct radiator cabinets or horns of obsolete design. To understand why this is a false assumption requires a review of basic speaker design concepts.

Two factors are paramount when discussing the low-frequency response and output capability of direct radiators:

1. *The limit of low-frequency reproduction by a direct radiator speaker is primarily determined by the Fs (resonance in free air) of the driver.* Ideally Fs is no higher than the lowest fundamental that you intend to reproduce, which for most popular music is represented by the low "E" note of the electric bass at about 40Hz. All other specs being equal, drivers with lower Fs will reproduce to lower frequencies than drivers with higher Fs when mounted in direct radiator cabinets. On average, the Fs for pro-sound woofers is about 60Hz for 10s, 50Hz for 12s, 40Hz for 15s, and 30Hz for 18s.
2. *The output capability of a direct radiator speaker is primarily determined by the product of cone area and excursion.* All other specs being equal, if two drivers of equal excursion have differing cone areas, the larger driver will be able to produce higher output when mounted in a direct radiator cabinet.

Different rules apply when discussing the passband and output of horn-loaded speakers:

1. *The limit of low-frequency reproduction by a horn-loaded speaker is primarily determined by the length of the horn.* While the Fs of the driver has bearing on system response, lowering driver Fs will not necessarily give better low-frequency response from a horn of a finite size. In fact, the opposite is possible.
2. *The output capability of a horn-loaded speaker is primarily determined by the area of the horn mouth.* Increasing the cone area of the driver in the horn will not necessarily increase either the horn's efficiency or output capability.

DRIVER/HORN RESONANCE

If you desire a low-frequency cutoff of 40Hz from a direct radiator, a driver Fs of 40Hz or less is usually required, explaining why 15s and 18s are preferred for electric bass and PA subwoofer duty. But if you desire a low-frequency cutoff of 40Hz from a horn, must a driver Fs of 40Hz be employed? The answer is no.

When a driver is mated to a horn, its Fs is lowered by the horn, resulting in a new parameter that I call Fs(h), the

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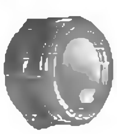

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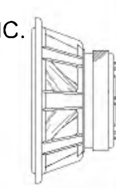

driver/horn system resonance. The smaller the throat and/or the longer the horn, the lower the Fs(h). With a horn of sufficient length and a small enough throat area, the Fs(h) may be half the Fs. A 10″ 60Hz Fs driver may exhibit an Fs(h) of 30Hz when horn-loaded, making it totally appropriate for subwoofer use.

It is possible to have an Fs(h) that is too low. In a horn with a vented rear chamber (as all DR horns are), the ratio of Fh (the horn cutoff frequency) to Fs(h) should be from 2:1 to 3:1; a higher ratio will cause a response dip in the second octave. You may raise the Fs(h) by either shortening the horn or enlarging the throat, but both of those actions rob SPL efficiency across the spectrum. On the other hand, a driver with a high Fs may have its Fs(h) lowered to the desired frequency by making the horn longer and/or the mouth smaller, enhancing both bandwidth and efficiency.

If you desire more output or lower frequency capability from a horn, is there an advantage to using a larger driver at the horn throat? The answer again is no. Even a 5″ driver may deliver high output at low frequencies with the right horn. On the other hand, there are distinct disadvantages to using large drivers in horn-loaded cabinets. This is best understood when you consider two paramount rules of horn design:

1. *Maximum output and bandwidth from a cabinet of a specified size is best obtained by minimizing the space taken by the driver and the rear chamber that houses it, devoting that space instead to the horn.* Whereas in a direct radiator you use a bigger driver if you want more power, in a horn you must make the horn bigger. Smaller drivers allow bigger horns in the same size box.
2. *Increasing the ratio of horn mouth to horn throat area will increase horn efficiency.* Maximum SPL from a horn of a given length and mouth area requires a small throat. Larger drivers require large throats; using them may well have the opposite of the intended effect.

A QUESTION OF EFFICIENCY

I've concluded that for horn-loaded applications, 10 and 12″ drivers make the most sense. As an example, look at *Fig. 2*. Here the DR15a is compared to the almost identically sized DR12. As good as the DR15a is, the DR12 outperforms it across the board, having a higher SPL at most of the measured frequencies from 32Hz to 8kHz.

The DR12 horn is more efficient, as the specified 50 to 60Hz driver Fs allows a 45 in^2 throat. To accommodate an Fs of 40Hz, and the added area of a 15's cone, the throat of the DR15a measures over 80 in^2. Efficiency suffers as a consequence.

Could you exceed the performance of the DR12 using a 15? Only by using a larger horn, requiring a larger cabinet. But you would get more out of a 12″ driver—or even a ten, for that matter—from a larger horn as well. The bottom line is that in horn-loaded cabinets small enough to be portable, a 15 is not the best choice, and an 18 is useless.

This is not borne out in the marketplace, where commercial horns loaded with 15s, 18s, and even 21s abound. An example of a commercial folded-horn cabinet loaded with an 18 is the Peavey DTH 118b™. Similar to subs from at least six other manufacturers, its sealed rear chamber and flat-panel geometry renders it an antique.

Comparing the performance of the 19ft^3 DTH 118b with the 13ft^3 DR12 in *Fig. 3*, the DR12 clearly outperforms the DTH 118b right down to 32Hz. Moreover, the Peavey™ rolls off high-frequency response above 250Hz, making a midbass cabinet, and another amp channel to drive it, a necessity. The DR12, like all DR horns, does not choke off the woofer's high end, requiring only the addition of tweeters to run full range.

Were I to build a cabinet as large as the Peavey, it would outperform 1950s vintage design by an even wider margin, and I would still use a 12 to do it. I would only recommend a 15 with a horn Fh of no more than 60Hz, which would require a box size no less than 30ft^3.

THE 15 CABINET

Now that I've convinced you, I hope, of why you should build a DR12 instead of a DR15a, why read any further? Simple. Because if you have one or two 15s already on hand sitting in obsolete commercial cabinets, and you want to give them a new home inside the best cabinet money can't buy, you can build this one. It's a whole lot cheaper than buying new drivers.

FIGURE 1: DR15a woofer only, model EVM-15B™. B-2187-1

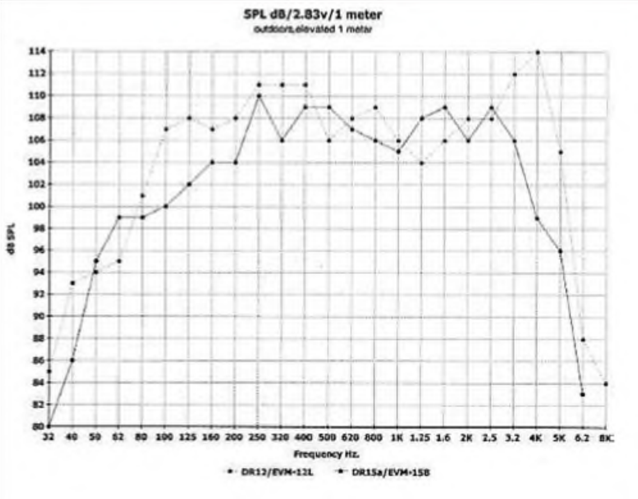

FIGURE 2: Comparing the DR12 and the DR15a. B-2187-2

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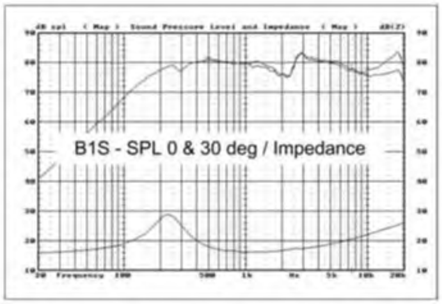

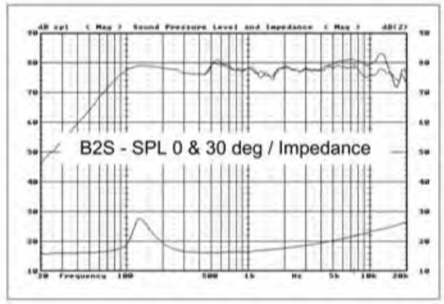

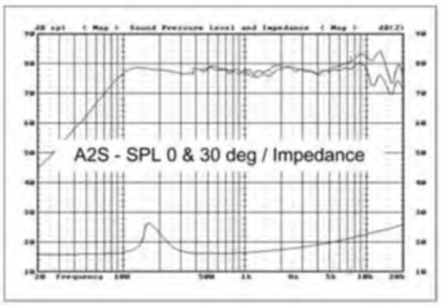

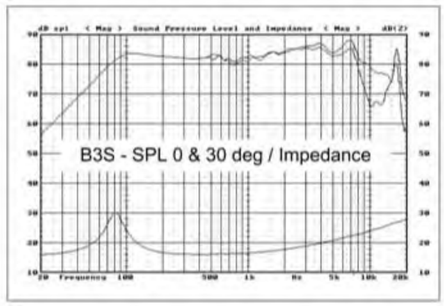


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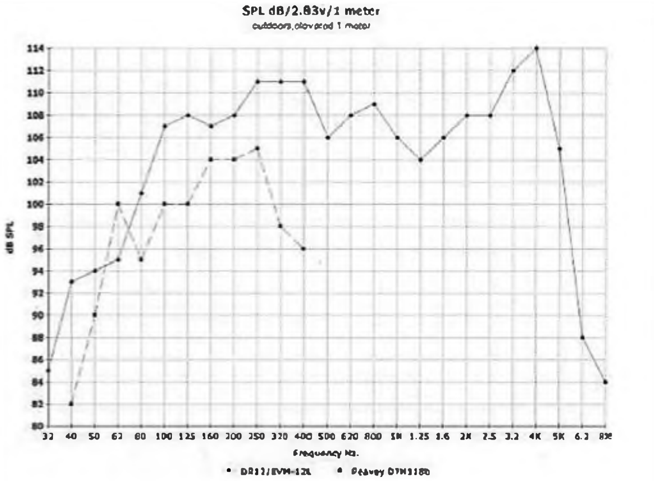

FIGURE 3: The DR12 versus the DTH 118b™: proof that you don't need big drivers for big sound. B-2187-3

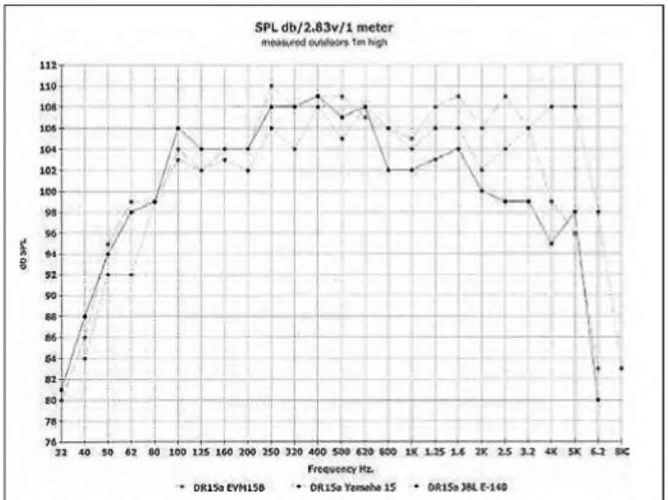

FIGURE 4: The DR15a with three different woofers. B-2187-4

Will your old driver work in the DR15a? Probably. When I built the prototype I tested it with three different drivers (I have a lot of old 15s about that I don't use anymore). The DR design is very tolerant of driver specs.

As for the V_{AS} and Q_{TS}, I have not found those specs to particularly affect performance in DRs, so I no longer consider them. The Fs is significant, but not critical. The drivers I tested were the JBL-E140™, the EVM-15B™, and an old Yamaha™ 15, model number unknown; their Fs/Fs(h) measured 32/23Hz, 39/25Hz, and 49/30Hz, respectively. In each case I tuned the cabinet Fb to the respective driver Fs(h). The results are in *Fig. 4*.

The JBL and EV are very close to each other up to 620Hz, above which the JBL rolls off a bit. When mounted as a direct radiator, it rolls off above 620Hz as well, so the performance difference between it and the EV is caused by the driver, not the cabinet. The Yamaha lags in the low end due to its higher Fs/Fs(h), and it runs about 2dB less in SPL broadband, but it runs 2dB lower in SPL on a baffle as well, so again the performance within the DR15a cabinet pretty much mirrors what it does in a direct radiator.

What is surprising with the Yamaha is its strong showing out to 5kHz. Who says that folded horns have no midrange? In any event, I'd say that just about any quality musical instrument 15 with Fs from 30 to 50Hz will work well in this box. Chances are what you have in your present cabinet falls within that range.

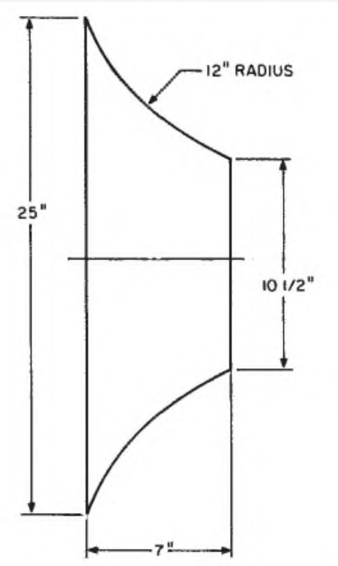

FIGURE 5: Side view of throat horn sides and partition. B-2187-5

There are a few structural differences between the DR15a and the DR12. I already noted that the throat area of the DR15a is larger, to keep the Fs(h) within a usable range. The cabinet does not have the trapezoidal shape of the DR12, to maximize the internal volume of the box within the footprint. To squeeze out a bit more high-end, the throat horn has flared sides to minimize out-of-phase reflections, and a partition in the middle to minimize the cross-section.

The ducts of the DR15a exit the rear chamber farther back in the horn for better low-frequency performance. And while I built the DR12 prototype with a woofer only, I built the DR15a prototype with tweeters, a pair of CTS KSN1177s™. Build it woofer only for electric bass if you prefer.

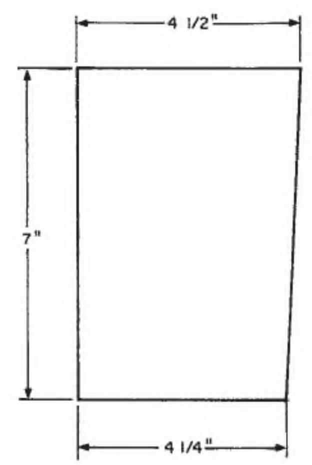

FIGURE 6: Throat horn divider (two required). B-2187-6

PRELIMINARIES

The prototype includes the thinnest materials possible for light weight, but if you're not a skilled woodworker you may opt to use thicker non-bending parts for ease of joinery. Most flat panels are of ½″ plywood, with ⅜″ and ¼″ plywood for bending parts. You secure most joints with drywall screws, holes drilled and countersunk, and construction adhesive, making sure that all joints are airtight. In places where fast setting is required, or screws won't work, you can use a hot melt glue-gun.

The sizes of all parts are nominal, as the true thickness of the materials you use will determine final sizing. Internal clearances are very tight, so if you use thicker materials than listed, increase the outside dimension of the cabinet to

compensate. In most cases cut the parts a bit oversize and sand-flush joints after joining them. You'll obtain the most accurate joints when first clamping the mating parts together, using a straight guideboard along the joint line where possible, and drilling, gluing, and screwing after proper alignment is achieved.

The driver chamber is fairly large, but reaching the woofer-attaching bolts is still a difficult proposition. Make life easier by using Allen-head bolts with T-nuts (available from www.partsexpress.com, as is most everything else you may need for hardware, including the recommended tweeters). A small right-angle ratchet driver is handy for driving woofer bolts and screws in tight areas where a screwgun won't reach. Sophisticated woodworking tools are not an absolute necessity—you could make do with a jigsaw, circular saw, and drill, but a table saw and sander are a great help, as are clamps—lots and lots of clamps.

If you have a table saw, make yourself a panel-cutting jig, which is simply a piece of plywood with two rails that slide through the slots in the table top. With this simple accessory you can easily cut large panels perfectly square. Never assume that plywood comes square from the lumberyard. I personally have never found a piece of Baltic birch that wasn't out of square. When it comes to accurately sizing parts, always cut in two steps. First rough-cut all your parts a bit oversize, and then trim them to square and true with the panel jig. Though this is not a project for beginners, extreme precision is not required. In fact, having things a tad out of whack can be a good thing. Perfect symmetry within any speaker cabinet can lead to resonant peaks and valleys, while some randomness tends to minimize spurious resonances.

PHOTO 2: Cutting the throat horn sides and partition simultaneously.

PHOTO 3: Using picture frame clamps to align the partition and dividers for assembly.

CONSTRUCTION AND ASSEMBLY

The first step in construction is to cut out the throat horn sides and partition (*Fig. 5*) from ½″ plywood. These three parts should be identical, so you should rough-cut them to size, screw the three pieces together, and cut them to final size simultaneously (*Photo 2*). Next cut the throat dividers (*Fig. 6*), also from ½″ plywood.

Install these offset on the centerline between the horn sides and partition. To one side of the partition offset them above the centerline, to the other side below it (*Fig. 7*), so that the throat horn is divided into four slightly different cross-sections.

Attach both dividers to the partition (*Photo 3*), then each horn side to the assembly (*Photo 4*). Screw a couple of scrap plywood pieces to the front of the assembly to hold it square and true while attaching the throat horn sheaths, cut from ⅛″ Baltic birch plywood; flex the plywood before cutting to find the easier bending axis. Cut the sheaths a good inch oversize in each dimension. Attach these one at a time to the assembly (*Photo 5*).

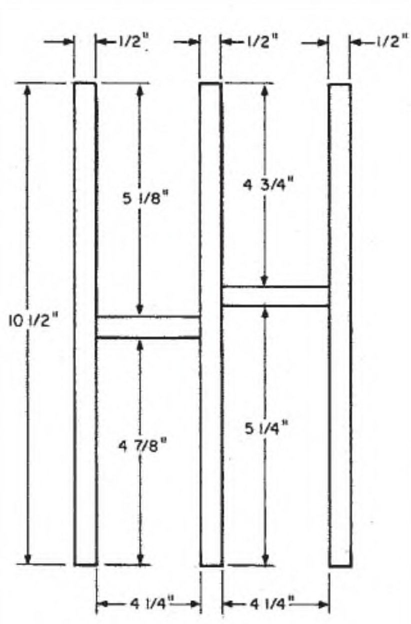

FIGURE 7: View of throat horn from baffle end, showing offset of throat dividers.
B-2187-7

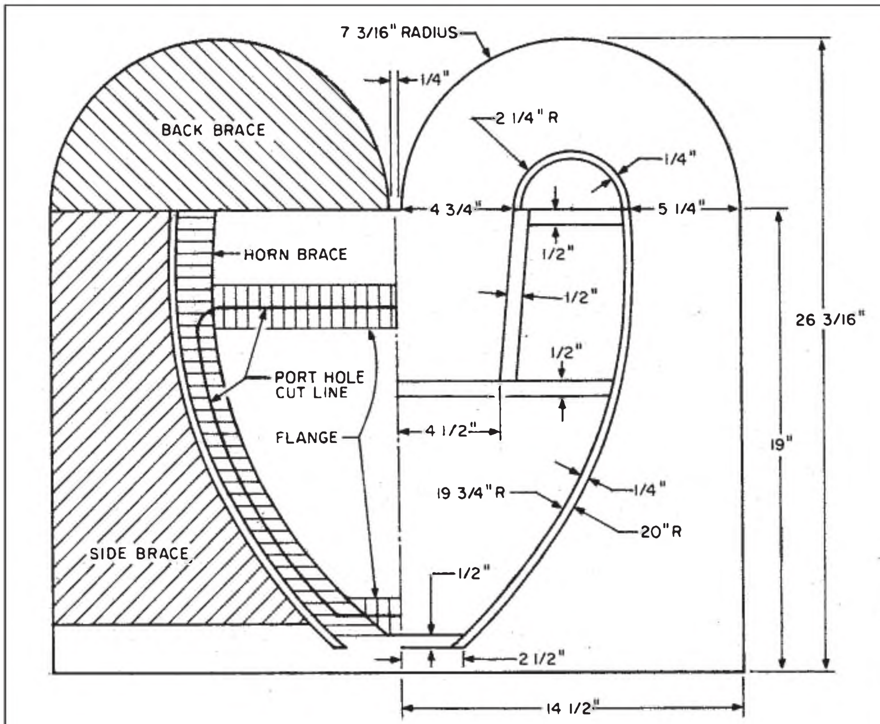

FIGURE 8: Top and bottom. Left side of view shows braces (shaded), right side shows parts locations.
B-2187-8

After the adhesive has set, use a jigsaw to trim most of the excess from the sides and throat, finishing the job with a sander; leave the excess material at the horn mouth for now. Cut the throat horn supports from ½″ plywood, making them about a half-inch longer than final size. Use a hole-saw to cut a few 2″ holes in them; these serve to minimize reflective surfaces within the cabinet. Mount them to the horn assembly (*Photo 6*). After the adhesive has set, run the assembly across a table saw on a panel-cutting jig to trim to final size and make it square and true (*Photo 7*).

Cut the top and bottom (*Fig. 8*) from

PHOTO 4: The throat horn sides/partition/divider assembly.

PHOTO 5: Sheathing the throat horn.

PHOTO 6: Using clamps and right-angle ratchet driver to attach horn support.

PHOTO 7: Truing the throat horn assembly on panel cutting jig.

PHOTO 8: Simul-cutting the top and bottom on panel cutting jig.

PHOTO 9: Using guide board and clamps to align throat horn on top.

½″ plywood. To make them identical, screw two rough-cut pieces of plywood together, final-cutting them simultaneously, using the panel-cutting jig (*Photo 8*). Draw onto them the locations of all mating parts, and the porthole cutout on the bottom only. Cut the porthole out. Attach the throat horn assembly to the top. Because the throat horn is slightly trapezoidal the supports don't follow a straight line, but the angle is slight enough to be easily pulled straight when clamped to a guideboard (*Photo 9*).

Cut the baffle from ½″ plywood. One end of the baffle attaches to the cabinet top, but make the other end only long enough to extend about an inch beyond the driver frame. Cut the baffle edges at a 10° angle to match the angle of the mouth horn panels where they will intersect it. Clamp the baffle into place against the throat horn and trace the horn opening onto it; remove the baffle and cut a hole through it on the tracing.

PHOTO 10: Aligning the baffle for attachment.

PHOTO 11: Aligning the tweeter baffle.

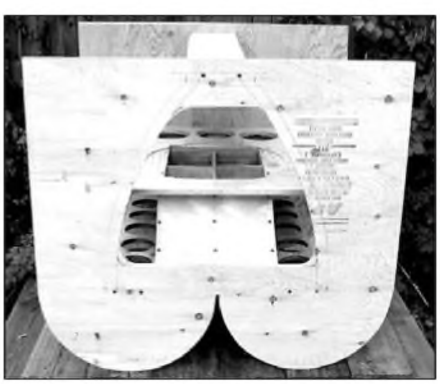

PHOTO 12: View through the porthole.

ALIGNMENT

Use the driver to mark locations for drilling the driver mounting-bolt holes, keeping them as far as possible from the baffle edge. Because the driver cone may hit the baffle in long excursions, either rout away about ⅛″ from the baffle where it would be hit or make a spacer from ⅛″ or ¼″ plywood, attaching it to the baffle. Drill holes in the baffle and drive into it 5⁄16″ T-nuts for the driver bolts. Attach the baffle to the assembly, using clamps and a T-square to true the assembly in the process (*Photo 10*).

PHOTO 13: Aligning a back support.

PHOTO 14: Make sure it fits! Note additional bracing.

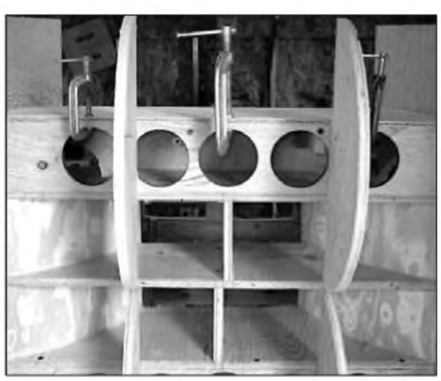

PHOTO 15: Attaching sheathing, step one.

Starting with a piece of ½″ plywood about 8″ wide, cut the tweeter baffle, with the edges at a 40° angle; save the scraps cut from either side. Attach the tweeter baffle to the assembly (*Photo 11*). Attach the bottom to the assembly

PHOTO 16: Attaching sheathing, step two.

PHOTO 17: Using cauls to align a side brace.

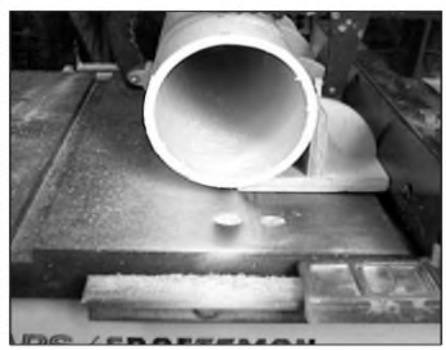

PHOTO 18: Halving PVC is easy with a home-made jig.

PHOTO 19: View with PVC reflectors in place.

and drill a few 2″ holes through the woofer baffle below where the driver mounts (*Photo 12*).

Cut the four back braces (*Fig. 8*) from ½″ plywood. Attach these to the throat horn supports, spaced not quite evenly apart to randomize the frequencies of internal reflections within the horn. Mount one pair 7½″ from the top, and one pair 8″ from the bottom, leaving the space between them at 8½″. Ensure that they are level by first clamping a guideboard to the horn supports, and then the back braces to the guideboard (*Photo 13*).

Cut four sets of horn braces (*Fig. 8*). Mount one set to the top, the other to the bottom to form the porthole flange. Cut the other two sets to size to fit between the baffle and the tweeter baffle, and from the throat horn supports to the baffle. Temporarily mount the driver on the baffle to make sure that these braces do not interfere with getting the driver in and out of the cabinet or accessing the driver mounting bolts. A couple of vertical braces fashioned from ½″ plywood 1½″ wide running from the bottom to the braces just above the driver are a good idea as well (*Photo 14*).

Use plywood scraps to complete the porthole flange; you may need to trim the flange a bit to allow clearance for the driver frame. Use the trimmings from the tweeter baffle to double its thickness at its edges. Cut holes for the tweeters in the tweeter baffle. Cut the mouth horn sheathing from ¼″ plywood. Attach these to the assembly first at the throat horn braces, butting them against the back braces, using clamps to pull them into place against the throat horn braces (*Photo 15*) for screwing. Gradually pull the sheaths into place, using the tweeter holes for clamp placements (*Photo 16*), gluing and screwing as you go. When the adhesive has set, sand the sheaths flush to the tweeter baffle.

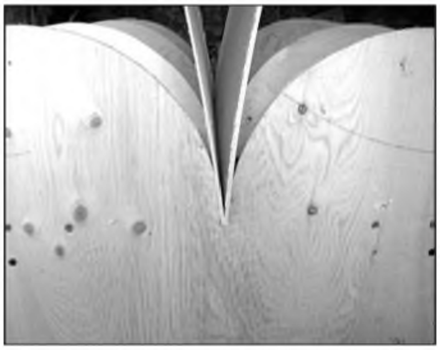

PHOTO 20: "Bookleaved" back sheaths.

PHOTO 21: A back sheath marked for drilling pilots.

PHOTO 22: Filling the back sheath seam.

Cut four side braces (*Fig. 8*). Attach them to the assembly, driving screws from inside the driver chamber, using a clamp and plywood cauls to align them with the back braces during the attachment process (*Photo 17*).

The back reflectors are 4″ Schedule 40 PVC pipe, halved lengthwise. You can easily accomplish this on a table saw using a fashioned plywood jig (*Photo 18*). The jig runs against the saw rip-fence, with the PVC securely clamped or screwed to it. You may use a toothed blade, but an abrasive blade is better.

PHOTO 23: Location of the ducts.

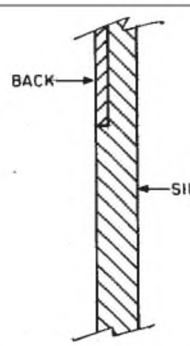

FIGURE 9: Detail of overlap of sides over back. B-2187-9

Don't cut quite all the way through the pipe, as it will tend to bind on the blade; go 95% through the pipe, leaving the final cut for a utility knife. After halving the pipe cut six lengths to fit the spaces between the top, back braces, and bottom. Fill each piece with poly-fill and glue it in place with hot-melt glue, being sure to make all joints airtight (*Photo 19*).

Cut the back sheaths from ⅛″ Baltic birch plywood. Glue them together at one edge with a bead of hot-melt, inserting the "book-leaved" pair into the cabinet rear (*Photo 20*). Bend the sheaths into place by hand, tracing the location of the joints with the braces for drilling pilot holes (*Photo 21*). Glue and screw in place first one sheath, then the other. Thoroughly fill the joint of the two sheaths with adhesive for an airtight fit (*Photo 22*).

You may choose to consider laminating a second layer of plywood on the back sheaths; they won't vibrate, but they may be less than roadworthy, especially if you entrust your gear to a crew of gorillas for transport. If so, allow the adhesive to set, pull the screws, and then laminate a second layer on with screws and plenty of adhesive.

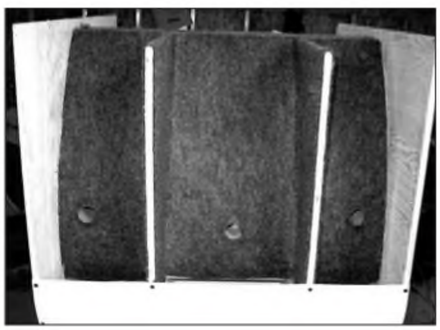

PHOTO 24: Applying finish before attaching sides.

FINISHING TOUCHES

Drill six holes to accept 1¼″ FVC conduit through the mouth horn sheaths for the ducts, three per side, spaced between the top, braces, and bottom (*Photo 23*). The two center ducts may be no more than 3″ long; install those permanently first. Locate the other four so that the ducts won't hit the throat horn inside the box. Make the top pair 7″ long and install them, leaving the final pair for the tuning process. Because the inner surfaces of the mouth horn are difficult to finish after the sides are in place, now is the best time to paint or carpet them as desired (*Photo 24*).

The sides overlap the back sheaths by a couple of inches, so you'll need to dado off ⅛″ (¼″ if you made the back double-thickness) from the sides where the overlap occurs (*Fig. 9*). You may do this with a router, or a table saw, using a finger-board to hold the sides tight against the rip-fence (*Photo 25*). Attach

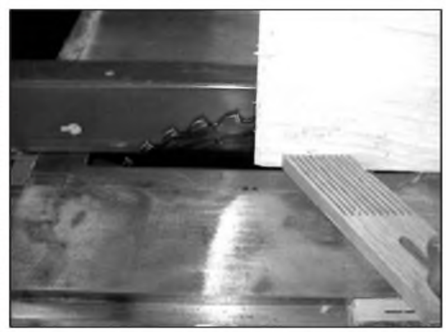

PHOTO 25: Using a featherboard with rip-guide to dado side edge.

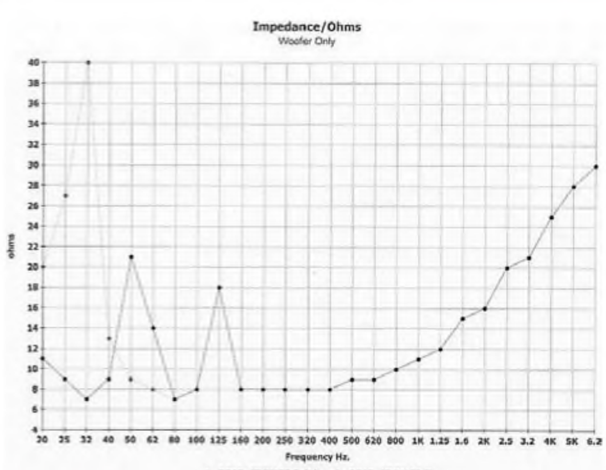

FIGURE 10: Fs(h)/Fb with the EVM-15B™. B-2187-10

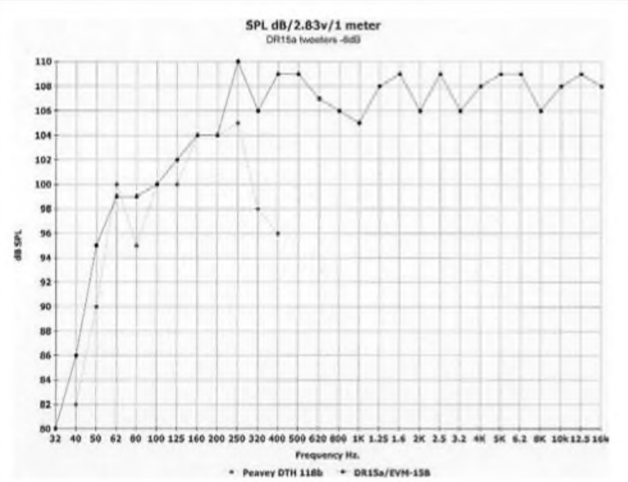

FIGURE 11: The DR15a (with tweeters) versus the DTH 118b™. B-2187-11

the sides to the assembly, sanding the joints flush after the adhesive has dried. Complete the application of your finish. Drill a hole as required through one of the sides to install your choice of jack and wire, along with a hole through the horn sheath for the wire to pass through, sealing with hot-melt.

Reflecting sound waves within the cabinet will cause serious peaks and valleys in response, so you must minimize reflective surfaces. Sound-absorbing pads are an effective, if pricey, option, and both they and acoustic foam can be difficult to install. I recommend pads on the driver frame and magnet, foam on the inside of the porthole cover, and poly-fill stuffing everywhere else. The key is to be sure that there are as few exposed reflective surfaces as possible without blocking either the rear of the woofer cone or the entrances to the ducts.

TUNING UP

The tuning process is next, assuming that you have the required equipment. If so, install the woofer, leaving the porthole off and the tweeter holes clear. Run an impedance plot, looking for a spike in the vicinity of 30Hz. This spike is the Fs(h).

Install the tweeters, weather-stripping their flanges, likewise with the porthole cover. Temporarily install the final two ducts, experimenting with their lengths until an impedance sweep shows a low, the Fb, at the Fs(h) frequency. For an EVM-15B with an Fs of 39Hz and Fs(h) of 25Hz making the last two ducts 5″ long gives the desired 25Hz Fb (*Fig. 10*).

If you cannot test your box, I suggest a good compromise for most drivers is to make the last two ducts 5″ long. By point of reference, loaded with the JBL-E140 the 23Hz Fs(h) required making the last two ducts 7″ long, while the tested Yamaha with Fs(h) of 30Hz required 3″ ducts for the final pair. Once you've decided on the length of the final pair of ducts, glue them in place from both the inside and outside of the cabinet.

You have two options when wiring the tweeters. If you wire them parallel with each other, you will get maximum sensitivity, giving a lot of high-end cutting power when needed; trimming amp EQ will flatten the response if necessary. On the other hand, if you prefer flat response, wire the tweeters in series with each other, cutting their SPL sensitivity by 6dB, while increasing their wattage to over 500. Wire the tweeters in-phase with the woofer and screw the porthole in place. With a cabinet of this size casters are a must, as are good handles; the bulk of the box, if not the weight, makes lifting it a two-person job.

If you question whether it's worthwhile to load an on-hand 15 into a DR15a, *Fig. 11* may help to make up your mind. Here the DR15a (with two CTS KSN1177s, series wired) is compared to our old friend, the Peavey DTH 118b. The Peavey is a lot bigger and heavier, but in the range where it is supposed to work best, low bass, it runs about 4dB less sensitive, while above 250Hz it doesn't work at all. I'd call that not much of a bargain when cabinets such as the DTH 118b go for around a grand. You can load your old 15 into a DR15a for about a hundred bucks—including the tweeters.

But please, if you don't intend your DR15a for an existing driver, don't build it! You can get more power from a DR12, and almost as much from the far smaller DR12a and DR10. And for those who want to see how small a folded horn can be, watch for the next in this series: the DR5. Your lunchbox is probably bigger.

PARTS LIST

All parts sizes are nominal, most somewhat oversized from final assembled dimension, all parts not listed (see text). Listed in general order of assembly, all dimensions in inches.

1. Throat horn sides and partition, 7 × 25
2. Throat dividers, 4½ × 7
3. Throat horn sheaths, 9 × 12
4. Throat horn supports, 25 × 3¾
5. Top, bottom, 29 × 26½
6. Baffle, 16 × 21
7. Horn braces, 4 × 20
8. Side braces, 12 × 18
9. Tweeter baffle, 5 × 25
10. Back braces, 7½ × 14½
11. Mouth horn sheaths, 20 × 25
12. Back halves, 24 × 26½
13. Sides, 19 × 26

Direct Heaters: A Guided Tour

This article will help you decide which scheme to use in applying power to vacuum-tube heaters. **By Mark Kelly**

I'm going to start this article with a long and rather involved technical discussion of the various types of heaters found in types of vacuum tubes (V/T), and then move on to the practicalities of different schemes for applying power to these heaters. If you find the technical stuff boring, or already know it, you can skip to the practical stuff and refer back to the first part as necessary.

Direct heaters are those in which the cathode and the heater are one unit, with a filament that may be:

- Pure tungsten–the so-called bright emitter, which runs at 2300C/2600K (dazzling white)
- Thoriated tungsten–the so-called dull emitter, which runs at 1600C/1900K (bright yellow)
- Oxide-coated tungsten or other metal–runs at 600–800/900–1100K (dull red to orange)

Indirect heaters use a tungsten or molybdenum alloy element fitted inside the cathode sleeve, which is coated with emissive oxides. These run at a temperature sufficient to raise the cathode to the same temperature as the oxide-coated direct-heated type. The object of this is to ensure thermionic emission—electrons are effectively "boiled off" the cathode.

This is a gross simplification; the actual mechanism involves a probability function of electron escape called the work function, which describes the amount of energy required for an electron to escape the surface of a given material. The lower the function the lower the energy—and therefore temperature, required for emission. There have been reports of a diamond film semiconductor with a work function so low that it provides high levels of electron emission at room temperature. One day we may get triodes that need no heater.

Tungsten is used for the very high temperature elements, simply because tungsten has the highest melting point of any metal, about 3750°C. Tungsten is a very dense (SG = 19), very hard but quite brittle metal, and like most metals displays a strongly positive temperature coefficient of resistance (a bit less than 1% per degree C, or 10,000 ppmK^{-1}). The other metals and alloys used have similar properties but lower melting points, so the following comments apply equally.

POTENTIAL PROBLEMS

The temperature coefficient of resistance means that the resistance offered by a piece of tungsten is very much higher when it's hot than when it's cold. The ratio of hot to cold resistance for heaters depends on how hot they become—usually about 7 to 1 for indirect types, 10 to 1 for dull emitters, and 15 to 1 for bright emitters (and for domestic light globes). There are two major effects associated with this. One is the problem of current surge at switch-on; the other is the tendency of the element to act as a current regulator when hot.

As an example of current surge, imagine an indirectly heated V/T fed from a constant voltage source. At switch-on the heater's resistance is 1/7 its hot value, causing the current (and power) drawn to be about seven times the heater rating. This effect is exacerbated in AC heaters, as the heater can be switched at any point on the AC waveform. If it happens to be switched at the peak, the voltage across the element will be 1.4 times its RMS value while the resistance is 1/7, so the power drawn is 14 times the rated value (V^2/R).

If the thermal mass of the heater is low enough, this pulse of power is sufficient to get the element to white heat. This causes the bright flash you sometimes observe at turn-on with AC heaters in some V/Ts—European-made small twin triodes (ECC8*/12A*7, and so on) seem particularly prone. This same effect also explains why your light globes just about always blow at turn-on.

The problem of turn-on current surge is worsened in bright and dull emitters with their higher ratios of hot to cold resistance. An additional effect that can occur in these larger V/Ts is that the pulse of current will induce its own

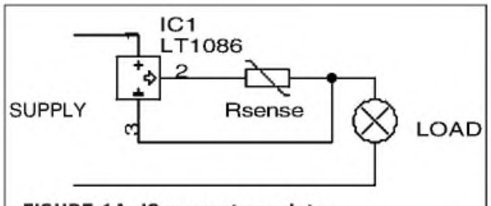

FIGURE 1A: IC current regulator.

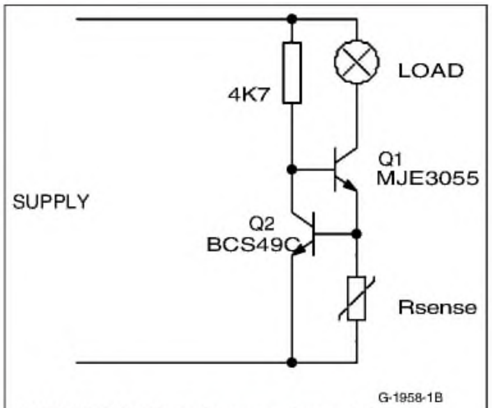

FIGURE 1B: Discrete current regulator.

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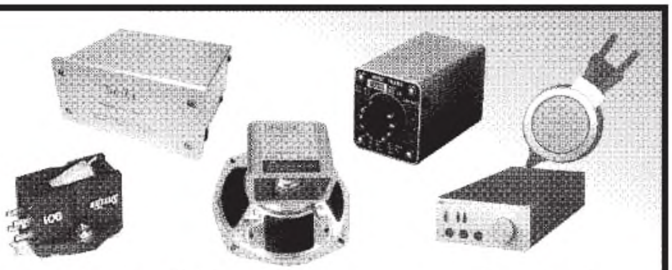

■Cartridge

Model	Price (US$)
Denon DL-102 (MONO)	150
Denon DL-103 (STEREO)	200
Denon DL-103R (STEREO)	250
Denon DL-103 PRO (STEREO)	350
Shelter Model 501 II (CROWN JEWEL REFERENCE)	750
Shelter Model 901 (CROWN JEWEL SE)	1,400

Postage (Air Economy)	
Area I $18	China, Korea Hong Kong Taiwan
Area II $22	Singapore Malaysia Indonesia
Area III $27	North America Oceania Europe
Area IV $34	Africa South America

These Area I ~ IV are for all products except book.

■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
Peerless 4722	38	50	20Hz～20kHz	300	Area III $40 Area IV $50

**Air Economy

■STAX

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

■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	Specifications				Price (US $)	Postage** (US $)				
	W	Pri.Imp(kΩ)	Freq Response	Application		I	II	III	IV	
XE-20S (SE OPT)	20	2.5 , 3.5 , 5	20Hz～90kHz	300B,50,2A3	396	47	56	84	113	Price is for a Pair
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	

** Air Economy

■TAMURA TRANS (All models are available)

F-7002 (Permalloy)	10	3.5	15Hz～50kHz	300B,50	836	60	70	110	145	Price is for a Pair
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	

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magnetic field, which will then induce motion of the filament across the field, the force of which can be enough to snap the filament (remember, tungsten is very brittle). Given that these V/Ts also tend to be expensive, current-limiting and/or slow-start techniques become very important in installations using them.

A different problem occurs if heaters are connected in series, where the increase in resistance will cause the first heater that reaches high temperature to hog all the voltage, thus raising its temperature and resistance, leading to a form of thermal runaway. For this reason the cheap mass-market V/Ts that were used in series heater strings—especially in TV sets—usually have controlled warm-up times, often about 20 seconds. If all the heaters in the string have the same warm-up time, thermal runaway will be avoided. This suggests that the US versions of the small twin triodes mentioned previously don't flash as much because they were manufactured with TV use in mind.

The other effect mentioned—that of the filament acting as a current regulator—is fairly self-explanatory. As the voltage across the heater increases, the resistance also increases, thus limiting the increase in current and therefore the increase in power dissipated. As an example, an increase of 10% in heater voltage gives a 5% increase in current rather than the 10% you might have expected. The increased temperature causes a 5% increase in the resistance of the filament.

This means a 10% voltage rise gives an increase in power dissipation by a tungsten filament of about 15% rather than the 21% which a linear resistance would. The safe operating limits for heaters are thus specified in terms of voltage, and current regulation of heater supplies is counterproductive.

TYPE OF HEATERS

Pure tungsten bright emitters run at a temperature so high that the metal evaporates, which eventually causes the filament to fail. Since these are never seen except in antiques and very high power transmitting V/Ts, we need not bother ourselves further with them, except to note that dull emitters are often called bright emitters, even by those who should know better.

Thoriated tungsten dull emitters are doped with low levels of thorium, which lowers the work function, enabling them to emit electrons at a temperature low enough for evaporation not to be a problem. The temperature is still high enough that the power consumption of a dull emitter is much higher than the equivalent oxide-coated type. An example is the WE212E/STC4212E, which is electrically very similar to the very old British DA250 triode. The 212E uses a 14V 6A (84W) dull emitter, while the DA250 uses a 10V 2A (20W) oxide-coated heater.

Dull emitters are not nearly as prone to cathode stripping as are oxide coatings, so they are often used in V/T designs where low levels of space charge make cathode stripping a problem, such as low-mu power triodes. Thoriated tungsten filaments are easily damaged by high levels of ion bombardment, so anything that causes the V/T to become gassy (such as outgassing from overheated electrode structures) will shorten life dramatically.

Direct oxide-coated heaters use a highly emissive oxide coating that allows low temperature operation offering the greatest thermal efficiency and are therefore used in situations such as battery V/Ts where power efficiency is very important. In some cases low-mu triodes also use oxide-coated direct filaments (such as 2A3 and 300B), but often these are designed with very high emission reserves to prevent cathode stripping and/or very high thermal mass to reduce hum. The very high emission from the oxide coating is due to it acting like a semiconductor; hence, even 300Bs are semiconductor devices—so much for the "no semiconductors" purists.

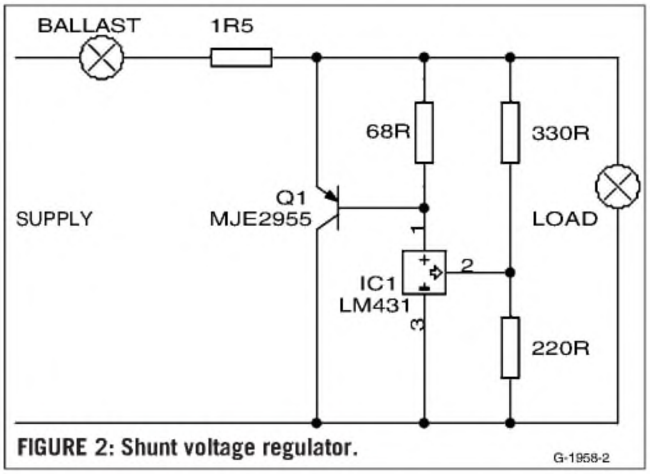


FIGURE 2: Shunt voltage regulator.

Most modern V/Ts use indirect oxide-coated heaters, which avoids the need to have separate heater supplies for each V/T and gives less heater-induced hum and noise. This system uses an oxide-coated sleeve to form the cathode, and the heater element needs to be very closely thermally coupled to, yet electrically insulated from, this sleeve.

One scheme to achieve this is to thread tiny ceramic beads on the element to fit inside the sleeve, but it's much cheaper to coat the element in a layer of insulating ceramic particles in a binder. For good thermal efficiency this layer should be as thin as possible, but it means that the electrical insulation it offers is easily broken down. This places a limit on the voltage difference between the heater element and the cathode sleeve, which can be tolerated without damage to the insulating layer.

Even if operated well within this rating, there will be slight leakage of electrons between the heater and the cathode, and this leakage current can cause noise if there is a high resistance between the two. For circuits in which the cathode is grounded for AC and the heaters are tied to ground, this causes no problem, but for circuits where the cathode floats, such as the long-tailed pair, SRPP, mu stages, and so forth, it is best to arrange for the minimum possible resistance between cathode of the upper V/T and its heater supply if low noise operation is important.

"SLEEPING SICKNESS"

Another effect that occurs with oxide-coated types is the growth of an interface layer between the sleeve and its oxide coating, which can effectively electrically insulate the coating, resulting in a loss of emission. This problem generally occurs when the cathode is kept hot while the V/T is biased at cutoff, hence its common name "sleeping sickness." Using V/Ts this way was a normal feature of industrial and computer use, and is approached in audio systems that keep the heaters lit in standby mode.

STC/Brimar (effectively the British arm of Western Electric) claimed to have solved the problem in their long-life series of V/Ts by using sleeve alloys and coating oxides that did not interact (retrofitting with these types was promoted as "Brimarising" your equipment). In the absence of this, it is prudent not to allow your equipment to operate with heaters lit but no B+ for longer than necessary. In normal conditions the flow of electrons from sleeve to coating keeps the interface in an electrochemically reduced state with respect to the coating, preventing the build-up.

Cathode stripping occurs when the outflow of electrons from the cathode exceeds—even momentarily—the available electrons in the space charge. This causes permanent electrochemical changes in the emissive coating, which reduces the emission available, thus reducing space charge, making further stripping more likely. Cathode stripping will occur when the potential difference between plate and cathode is established before the cathode has reached full emission, so some form of B+ delay is very important for all V/Ts using oxide-coated cathodes.

PRACTICAL STUFF

The various ways of applying power to the heaters can be classified as follows:

AC
mains frequency
RF and intermediate frequency

DC
battery supply
rectified AC, unregulated
- capacitor input
- choke input

rectified AC, regulated
- current regulator
- series voltage regulator
- shunt voltage regulator

switch mode derived DC

I have seen all of these applied to V/T circuits by different designers, with varying degrees of success. Any of these schemes may be applied to heaters individually, multiply in parallel, or multiply in series, although this last is now very rare, driven as it was by the need for utmost economy at the low end of the market.

The predominant design historically has been AC heaters in parallel, driven from one or more heater windings on a mains frequency transformer. Some believe this to be the best overall system, but it is not without its problems. Among these are mains frequency hum, noise and associated hash (especially in directly heated V/Ts), and lack of regulation of voltage and current.

Mains hum has three components—electrical, magnetic, and thermal. Thermal cycling of the heater element occurs because it gets hottest at the peak of each wave and coolest at the crossing point, giving 100 hot flushes per second. This is reduced by giving the element and sleeve a high thermal mass, ensuring that the heating and cooling are slow enough that the thermal cycling is minimal.

Electrical hum is caused by the modulation of the plate to cathode voltage by the heater voltage and is only a problem in directly heated V/Ts.

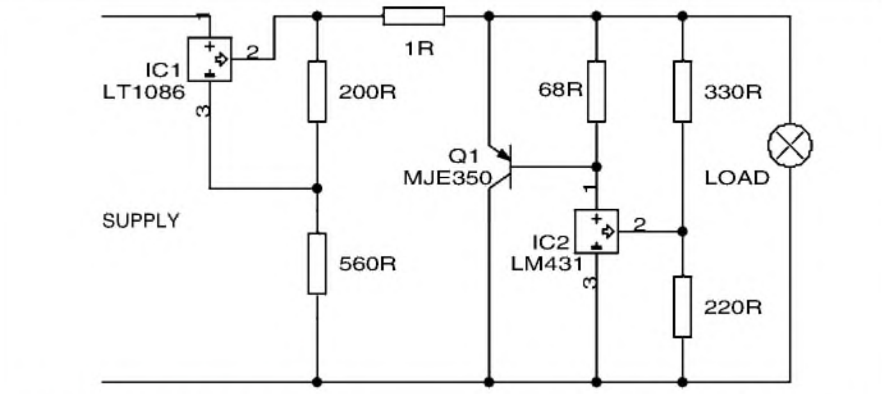


FIGURE 3: Series/shunt regulator.

Magnetic hum is caused by the mains frequency setting up a magnetic field, which induces eddy currents in any V/T structures, such as support rods, which are made of ferromagnetic materials. This is reduced by winding the element in a way to cause the field to self cancel (such as the bifilar winding in the EF86/6267) or making all structures non magnetic (such as stainless steel support rods in the RCA "red tubes").

In most cases mains frequency hum can be reduced by connecting the ground return of the V/T to a point midway in the heater voltage, by center tapped heater windings or a tapped resistor (adjustable is even better) across the heater supply. The problems with mains noise and hash are reduced by employing an electrostatic screen (Faraday screen) between primary and secondaries of the power transformer, and these should be specified in all power transformers for audio use. These problems are considerably worsened if the HT supply and the heater supply are on the same transformer, because the main culprit here is diode reverse switching transients causing ringing in the supply transformer, which will be quite audible in the heater.

The switching transients are much worse with capacitor input HT supplies because of the large current pulses as the capacitors recharge each cycle. This means that quiet diodes are very important—the quieter the better. I am firmly in the ultra fast soft recovery silicon diode camp, but I also use choke input HT supplies and DC heater supplies. Vacuum diodes have high series resistance and consequently a soft turn-off characteristic, so if you're not prepared to go to the expense of choke input supplies, then vacuum diode rectifiers are a reasonable compromise.

As to preference between different types of vacuum diode, all I can say is if you can hear your rectifiers your power supply design is inadequate. I have addressed the problems with poor regulation in the previous section about the positive temperature coefficient of tungsten.

SOLUTIONS

One method of reducing the problem with mains noise is to raise the frequency of the heater supply until it is outside the audio band and the thermal cycles are so short that they have almost no effect. I have seen designs calling for RF supplies. I know only one person who has built one of these, and the design was abandoned because all the heaters acted as aerials transmitting modulated RF signals to each other, causing huge problems with hash. This could be reduced by lowering the frequency to the point where structures the size of V/T heaters became inefficient aerials, but intermodulation products are likely to be a problem at these intermediate frequencies.

You can avoid all these problems by supplying the heaters with DC power, and this is as old as the V/T—the first domestic radios used battery power because the available rectifiers were too cumbersome and expensive. Obviously with DC there is no thermal cycling and the only signal modulation will be by residual ripple, but in directly heated V/Ts this comes at the cost of uneven grid to cathode voltage due to the heater potential across the filament. This does not, despite some claims to the contrary, compromise sonic performance—any sonic problems encountered with DC heaters are much more likely to be due to inadequate power supply design.

It does mean that one end of the filament will have slightly higher emission and so will theoretically "wear out" first. If this is a concern, it is a simple matter to arrange that the physical placement of the heater positive and negative are reversed for each member of a pair of V/Ts (e.g., each channel for single-ended or each phase for balanced) and swap the two V/Ts every so often. For high power transmitting V/Ts the standard recommendation was that heater polarity should be reversed every 500 hours of operation, but these were operated in deep class B or C, in which case even a small potential difference will result in the most negative part of the cathode leading and lagging the current pulse, greatly increasing the difference in emission.

BATTERY SUPPLY

As mentioned, the first applications used batteries for heater supplies, and battery supply remains the best all-round solution, as long as you can afford batteries to supply the required power—a 212E heater would be a real challenge for the Duracell bunny. The old radios usually had lead acid 6V batteries as the A supply for heaters (the B battery supplied HT, hence the term B+ for the HT supply, and the C battery provided bias). The voltage from a lead acid cell is 2.1V standard, so a three-cell battery nominally 6V will deliver 6.3V. This is the reason that 6.3V is the standard heater voltage.

The heater batteries will need to be recharged, and the standard recharging potential is 6.9V, which is why this is the upper spec for 6.3V heater ratings. The battery is considered fully discharged at 5.7V—the lower spec.

Are you beginning to see a pattern here? Lead acid batteries are hazardous due to their sulfuric acid content, and they give off hydrogen when cycled, a potential explosion hazard in a confined space. These days you should use a deep cycle sealed lead acid (SLA) battery for safety. These are available in a range of current capacities, specified in Amp hours (Ah). A 12Ah battery, however, cannot be expected to deliver 12A for one hour because this is too high a current drain. The Ah rating of a battery is normally specified at the 20 hour rate—the 12Ah battery will give 600mA for 20 hours, and it is best to use them at about this rate, so the 12Ah battery would suit a 600mA heater.

You will need to recharge this battery, and it is best not to have the charger running while the V/T is in use, which rather defeats the purpose. It is important not to allow the charging current to exceed the maximum current specified by the battery manufacturer; this will cause outgassing and reduce the life of the battery. A current-limiting resistor or, even better, a constant-current regulator in the circuit will prevent this. Your current regulator should be on the supply side of the voltage regulator so that the battery gets constant-current charge at the beginning of the charge cycle and constant-voltage charge at the end.

I use battery supply for the directly heated output V/Ts in my new preamp, and the regulated DC supply for the indirectly heated drivers is switched over to charge the batteries when the preamp is turned off (this requires supplies at 6.9V or higher, which then need to be reduced for the heaters, but refer to the series shunt regulation section later on). This arrangement allows a duty cycle of 16 hours in 24; if you need more than this, as might be required in pro installations, you will need back-up batteries.

RECTIFICATION

This brings us to rectifying an AC heater supply to DC. The most primitive scheme is a bridge rectifier with capacitor input, a voltage dropping resistor and capacitor for filtering, but you should avoid this design at all costs. The capacitor input produces the problem with large current surges and induced ringing in the supply transformer outlined previously, and this induced noise will ride through into the HT supply. With heater supplies there usually isn't the headroom to insert enough series resistance to slow it down, and you'll just go round in circles attempting to increase the filtering by increasing the value of the capacitors because the larger the cap the more current it takes to charge, hence the bigger the current surge.

A much better solution is the choke input, the main problem here being the difficulty and expense of obtaining the type of choke required. You will need one of a value well above the critical inductance ($R_1/6\pi f_{supply}$), where R_1 is the load presented by the heater. For example, a 211 or 845 presents a load of around 3R1, giving a critical inductance of 3.3mH for a 50Hz supply, so a choke of 6mH or more would be indicated. This choke will need to maintain this inductance with 3.25A DC through it and ripple current corresponding to 7V AC. At the same time the DCR will need to be very low to reduce copper losses. Such chokes are made by One Electron and Bartolucci, among others.

CURRENT-REGULATED SUPPLY

All these schemes share the problems with lack of regulation canvassed previously. The obvious solution is to regulate the heater supply, and here you have the choice of current or voltage regulation. As outlined previously, the positive temperature coefficient of tungsten means that heaters should always be run with constant voltage across them, so I believe that current-regulated supplies are counterproductive, apart from the benefit of total elimination of current surge at turn-on. You are quite entitled to disagree with me, so if you want to try a current-regulated supply, here's how.

The simplest method is to use one of the common three-terminal series voltage regulators and a current sense resistor (*Fig. 1a*). The resistor sets the output current to the value that gives a 1.25V drop across the sense resistor (for IC regulators with Vref = 1.25V, which is most of them). An output current of 600mA therefore requires a resistor of 2R1, a current of 1A requires a resistor of 1R25, and so on. The resistors will need to handle dissipating 0.75W and 1.25W, respectively—I would use a 5W wirewound.

Most common V/T heater currents don't happen to coincide with the E24 or even the E96 schedule of preferred values, so you will need to use series or parallel resistors to make up. The voltage drop across the regulator and sense resistor will be the regulators' dropout voltage plus 1.25V, giving a drop of greater than 4V with the common LM31x series of regulators. This means you need about 12V DC to feed a 6.3V heater after allowing for 10% line regulation, and the regulator will dissipate large amounts of power, which will make it run hot—for instance, a 600mA supply will mean the regulator will be dissipating 3W, so a heatsink rated at 10C per watt would be appropriate. You can reduce the dissipation by using a low dropout regulator such as the LT series, or if you don't mind the heater sitting slightly positive to ground, you can use the constant-current source of *Fig. 1b*.

This will operate quite happily with only 1.2V across it, as long as the tie point for the top BJT is higher than that. In this case the current is set by the resistor Rset in the emitter circuit of the lower BJT and is approximately 650mV/Rset.

VOLTAGE REGULATION

As mentioned previously I firmly believe that voltage regulation is preferable, so I'll go into much more detail about this. The usual method of voltage-regulating supplies is also to use one of the many available three-terminal IC voltage regulators. These are series regulators—regulation is achieved by inserting a variable resistance in series with the load.

Three-terminal regulators are available with output voltages up to 125V and with output currents up to 10A (not simultaneously), which covers all the heater supplies you are likely to need. The best designs offer high speed, low noise, low "dropout" (difference between supply and output voltage), and high regulation, as well as having inbuilt short circuit, over temperature protection, and current limiting—this last is a very useful feature. By choosing a regulator with a current limit less than twice the rated heater current, you can greatly reduce the switch-on surge. You can further improve this by inserting an NTC surge limiter in the primary circuit of the heater supply transformer, and this is as much surge protection as dull emitters or oxide-coated heaters need.

The small additional cost of the higher performance regulators over the 78xx and LM31x series will be amply repaid by their better performance. I have used the Linear Technology LT series with great satisfaction over the years. The output voltage of a variable regulator is set by a divider off the output back to the reference pin, again usually at 1.25V below the output voltage.

For very precise work you must take account of the input current to the reference pin in determining the resistor values for the voltage divider, and to lessen this effect it is a good idea to keep the values fairly low—generally the top resistor should be between 100 and 200R, meaning that the divider network will shunt about 6 to 12mA. For quietest operation, you should bypass the top resistor with a small fast capacitor of about 10 to 50μF—the Sanyo Osh-Con series is useful for this application.

SERIES SHUNT REGULATION

I have recently been playing with shunt-regulated supplies, however, and with appropriate design they measure and sound better. A shunt regulator uses a passive series resistance between the supply and the load and a variable resistance in parallel with (shunting) the load. Reducing the shunt resistance increases the current drawn through the series resistor, increasing the voltage drop across it and thus reducing the voltage across the load. Any resistance in the supply acts as part of the series resistance.

For this design to be short-circuit proof requires that the series resistor be large enough to dissipate the entire power drawn from the supply in the short-circuit condition, and that the supply be able to cope with this level of current draw. This is improved by using a large value of resistance as the series element and correspondingly higher supply voltage, but this arrangement becomes steadily less efficient as values are increased.

For it to be open-circuit proof, the shunt element needs to be able to dissipate the power generated when it shunts the load current in addition to the normal shunt current. The shunt element needs to be fast, quiet, and capable of sinking reasonable currents. None of the commercial IC shunt regulators (LM431 and similar) have the current capacity for these applications, so you will need to use either a discrete circuit using a voltage reference diode and a comparator, or the enhanced IC circuit of *Fig. 2*. I obtain best results with a high-performance audio transistor as the shunt element (e.g., Motorola MJE350).

The load resistor on the output of the 431 will need approximately 0.8V across it and pass enough current to supply both the IC and the base of the transistor. The IC performs best with about 5–10mA through it, and the base current in the transistor will be approximately the shunt current divided by the Hfe of the transistor. For example, with a 150mA shunt current and an Hfe of 30, the base current will be 5mA and the total current 10–15mA. So a 56R to 82R resistor is appropriate.

One development of this idea is to use a nonlinear resistance with a positive temperature coefficient as part of the series resistance, so that the increase in current draw increases the resistance of the element. The best nonlinear resistance is—you guessed it—a tungsten filament. You can use specially designed V/Ts called ballast tubes or barretters in this application if you can find one with the specs you need.

Because it is unnecessarily expensive to use a V/T for this, I used 6V motor cycle lamps for 6.3V supplies and 12V car lamps for higher voltage supplies. You may be able to find a lamp holder to solder directly into printed circuit boards. I used a plug with flying leads to a chassis-mounted lamp. Although the lamps are rated for 1000 hours service, this would only apply in the short-circuit condition in this application. In normal use the lamp sees about half rated voltage, so expected life will be 100,000 plus hours—that's over 30 years at eight hours a day.

As a practical example of a shunt regulator for a heater supply, you can use a common V/T such as a 6SN7GT with a heater rated at 6.3V 600mA. Good design practice calls for the regulator system to be able to cope with line voltage changes of ±10%, both in terms of maintaining regulation and having sufficient dissipation. A sensible minimum shunt current is 10% of load current, so the regulator system needs to be designed to provide this when line voltage is at −10% of nominal.

The other major constraint is the available transformer winding voltage. It makes sense to keep to reasonably easily available voltages.

Using a 9V winding with the previously mentioned constraints calls for a series resistance of 4R3. This can be made up of a 6V 12W lamp, which will have a resistance of less than 1Ω with a current of 1A through it, and a 3R3 15W wirewound resistor. In the short-circuit condition, with 12.6V across the resistor and the lamp, the lamp's resistance will increase to about 3Ω, limiting the current to 2A. This will eventually cause the transformer to overheat.

These calculations are at line voltage plus 10%, assuming zero effective impedance in the transformer windings. Real-world conditions will be less severe. The current draw at turn-on will also be limited to about this value (the lamp heats much more quickly than the V/T heater), offering very effective turn-on surge limitation. The major drawback of shunt-regulator systems is that they are wasteful of power—in the previous case a 6.3V 600mA (4W) heater supply is drawing about 10W from the transformer.

SERIES SHUNT REGULATION

One way around this—and the scheme I now use—is double regulation, or series shunt regulation. To do this you need an IC series regulator running at a higher voltage followed by a shunt regulator to give the required voltage. Because the input is already regulated, you needn't worry about supply side swings, so the voltage drop in the series element can be quite small, and the shunt current can also be a small fraction of the heater current. You can also take advantage of the current limiting in the series IC regulator to give short-circuit protection, and as long as the shunt transistor has enough heatsink, open-circuit protection is not a problem.

I use a 6.9V series regulator (so the output of the series regulator can be used to charge the battery supply as outlined previously) followed by a 6.3V shunt regulator based on the LM431 IC (*Fig. 3*). In this case the shunt current is about 100mA, so the series regulator needs to deliver 700mA at 6.9V, giving about 5W. The series regulator input needs to be 1.2V above this plus allow for diode drop, cap ripple, and 10% line variation, giving a minimum of 11.5V unregulated DC. But again, it makes sense to use readily available secondary voltages, so try for a 9V AC winding, giving a theoretical 12.6V. This means the series regulator will be dissipating about 3W, so it will need a heatsink.

An advantage of this scheme is that the series regulator circuit can be in the power supply and the shunt circuit wired close to the V/T socket, so that if you use an umbilical between them it is carrying DC, but any noise pickup in the wiring is shunted out.

If you have stayed with me throughout, you are possibly wondering whether all this effort is worth it. As I gain more experience, I become more convinced that the main influences on the sound of V/T amps are, in order of importance:

1. circuit design
2. transformer quality
3. power-supply design
4. parts quality

It is often much cheaper, and always more effective, to work on the first three. Heater design is an important part of power-supply design, and the cost difference between quality levels can be very small. ❖

Easy Layouts and Other CAD Tricks

Here are some tricks of the trade for building speakers and producing printed circuit boards. **By Patrick M. Brunner**

Having built an assortment of speaker projects (such as Mr. Edgar's tractrix horn) and electronic projects, I have found it much easier to design a project layout with the use of a CAD program. I use "Generic Cadd, Rev.6.0," which is cheap (under $100 the last time I saw it in a retail store) and easy to learn and use, plus it has some very convenient printout features. Any CAD program you may have available should suffice.

The big advantages with these types of programs are easy precision with digital coordinate readouts (up to six decimal places, although three is plenty), snap-to-grid points, easy repeated features with linear or radial copy, and editing by a couple of keystrokes at most.

The other desirable item for implementing these construction tricks is a wide carriage tractor feed printer (for larger projects), although a narrow carriage unit will suffice in most instances with a little added work. Even a 9-pin unit is adequate. I use an old, slow, reliable Epson Ex-1000 as well as a Stylus-1520.

SPEAKER CABINET LAYOUTS

1. Draw cabinet's panels actual size. CAD features allow easy, exact placement of evenly spaced assembly holes, accurate curves, and other details not easily done by hand.
2. Make accurate, actual-size drawings of each component (on a separate layer) to allow for easy component placement and to verify that the parts fit without interfering with each other or cabinet structure. Be sure to include mounting holes and any other pertinent details of the part. Selecting different colors makes it easy to differentiate various parts, layers, panels, and so on. You can view the whole project easily. Turning layers on and off reduces the confusion on more complex drawings.
3. An important "calibration" trick is to draw a rectangle 5″ × 10″ (use narrower dimensions on a narrow carriage printer) and print it out at a 1 to 1 scale. Measure the resulting printout and verify that the result is 5″ × 10″. Your printout's dimensions may be off. If they are off, re-scale your test drawing (write down the re-scale correction factor(s), possibly different for the X and Y axes) and make another test print. You should be able to get an exact 5″ × 10″ printed rectangle. If scaling is needed, re-scale during the printout configuration. Don't re-scale your drawings.
4. Select the layer(s) (and objects) of interest for printing. Using the printout features, print out the completed drawing to get a 1 to 1 print. If the panel (or other items) are greater than 13″ wide, make one or more printouts to cover the needed width. Few projects will need more than two printouts (26″ wide). With Generic Cadd R6.0 you can select any print length you like, so with tractor feed paper any length is easy. If you need two or more printouts, join them with transparent tape.
5. Spray (lightly) the back of the printout with adhesive and attach to the panel you are working with. This will make locating holes easy and cutting panels simple. Drill and cut through the paper. Peel the printout remains off when finished. Use the spray adhesive sparingly so as not to bond the paper permanently to your panel or leave excessive adhesive residue on your work piece.

CHASSIS AND CONTROL PANELS

1. Using the same methods just noted, you can rapidly lay out an aluminum panel or chassis, center-punched and drilled, filed, and so on.
2. In addition to the panel's components and mechanical layout, you can use a separate layer for the panel's text and any graphics you desire. You can then implement front-panel graphics using Brady Laser Plate® adhesive-backed laser printable plastic film. (A similar product may be available under other names.) It comes with a protective paper carrier and has a matte finish. A laser printer will give the best-looking "label" for your front panel, but you can copy an inkjet printout in a pinch with lesser quality.

The Brady Laser Plate comes clear as well as in colors. The clear film with black laser printing looks good on aluminum or white (and some colored) Plexiglas. The results are very good for prototypes and home projects, but not as nice as (or expensive as) silk-screened or engraved panels.

After using the paper drill and cut pattern, clean and de-burr holes and edges. Remove any adhesive residue. You can texture aluminum panels with Scotchbrite abrasive pads for a uniform finish. Clean the panel thoroughly. Carefully remove the Brady Laser Plate backing and position it on the panel. Smooth out any air bubbles and then burnish the film for a good bond. Trim out holes and edges with an X-acto knife.

CROSSOVERS AND OTHER SIMPLE PCBS

1. You can quickly implement crossovers on copper-clad PCBs stock by doing a 1 to 1 layout with the same methods previously described. Do the layout drawing as viewed from the component (non-copper) side. Add "cut" lines dividing the copper, producing large wide conductors.
2. Do a mirror image of your layout and then print it out. Using spray adhesive, mount the printout to the copper-clad side.
3. Drill all of the components' leads mounting holes. A small tungsten carbide drill bit works best but is very brittle. Take care if you use one.
4. Route along the "cut" lines with a Dremel tool with a round-ended bit set to cut ¼ to ⅓ (0.015″ to 0.020″ deep) the way through the board stock. A Dremel routing attachment makes the depth control easy. Tungsten carbide tools last longer when dealing with fiberglass boards.
5. Remove the paper; clean any adhesive residue and sand away any burrs with 240-grit silicon carbide paper wet or dry paper. This also makes soldering easy.
6. Mount the components on the non-copper-clad side and solder the leads. When finished, if desired, clean off the flux with alcohol or other suitable solvent. Spray the copper with clear Krylon acrylic spray if you want to keep the copper shiny.
7. Suggestion: Input and output terminals can be made with steel or brass screws soldered on the copper side. Use a nut to secure the screw while soldering.
8. Suggestion: Don't forget holes for cable ties to mount larger components, and holes to attach the crossover to your cabinet. ❖

The ZERO Autoformer

Discover the advantages of high impedance speakers with this autotransformer to improve your system's sound. **By Paul Speltz**

Low impedance speakers are not a good match for small amplifiers, especially vacuum tube output transformerless (OTL) designs, but that was the combination I found myself living with a few years ago. The basic problem was that the woofers were electrically underdamped, which caused a bloated-sounding bass. After considering and trying a few obvious solutions, I used an autoformer to transform my low impedance speakers to high impedance speakers. This increased the amp-to-speaker damping factor. In addition to the expected results, I found some unexpected benefits.

MISMATCH TROUBLES

As a veteran speaker builder, with the experience of roughly ten different speaker designs built from scratch, I had essentially settled into the speaker system of my choice. It was and still remains a 3-way system using Dynaudio and Peerless drivers, making a nominal 4.7Ω load speaker system. Being content with my speakers, I started exploring amplifiers, which led me to tube amps, and then OTL tube amps. By definition, my OTL amp's output tubes need to directly handle the low 4.7Ω impedance load of my speakers. I would have been set if I had previously designed and built a 16Ω speaker system that would have been more OTL friendly.

Isn't this just how it goes sometimes; I spent years designing, building, changing, tweaking, and then finally settling in and loving my DIY speakers. Then I borrowed a friend's OTL tube amps. Even though the OTLs didn't properly damp the 4Ω Peerless woofers, they brought life to the music in a way that there was just no going back to a more traditional/popular amplifier design. I knew I had to build myself a kit pair of Atma-Sphere M-60 OTL monoblocks (no longer available in kit form).

I considered another speaker project, this time a 16Ω design, but I was never able to build up enough steam to take it past the concept stage. I had already experienced the benefits of no feedback in tube amplifiers, so I was unwilling to increase the damping factor of the amps that way.

Instead I decided to address the overly blooming bass with a bigger OTL amp. This would lower the amp's output impedance and raise the damping factor. I did this by building my M-60 kit using larger output power supplies, and loading up each amp with eight 6336B tubes per channel in place of the stock 6AS7 tubes. Each 6336B tube is roughly equal to two stock 6AS7 tubes, so I had essentially doubled the amp.

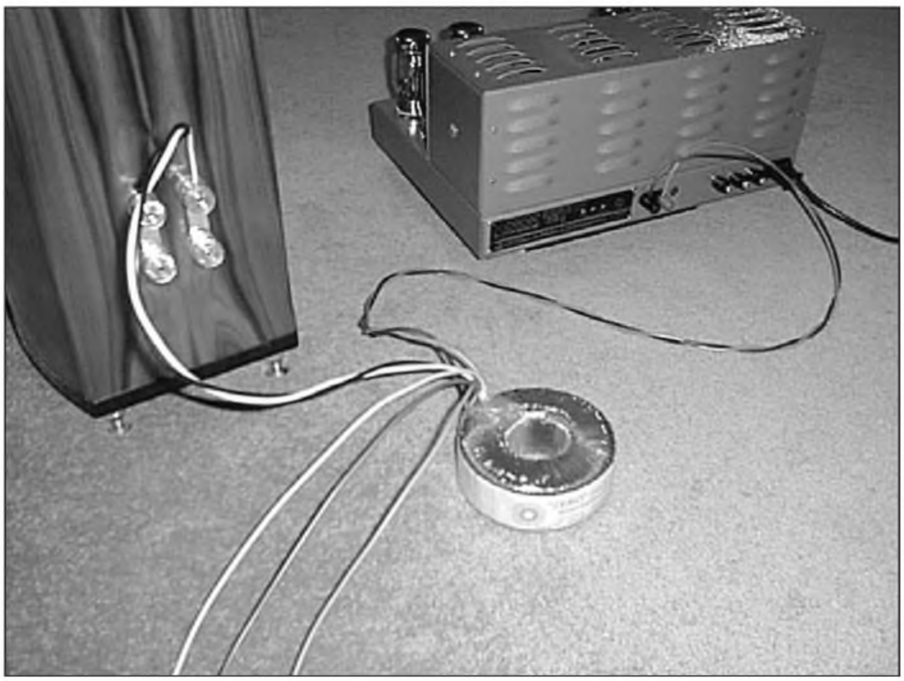


PHOTO 1: The ZERO speaker impedance multiplying autoformer used to multiply the impedance of any speaker so that it "feels" like the optimum load for the amplifier being used. For example, the ZERO can transform a 4 or 8Ω speaker into a 16Ω speaker.

ABOUT THE AUTHOR

Paul Speltz is a DIY two-channel audio enthusiast. He became a speaker builder in high school, and the hobby evolved to tube amp kits, DIY preamps, DIY interconnects, DIY speaker cables, refurbishing vintage tube gear, and turntables. Paul earned his BS degree in Industrial Technology, concentration in Electronics, at the U of Wisconsin-Stout in 1985. He is now the Senior Electronic Engineer for Viking Electronics, a telecommunications equipment manufacturer (www.vikingelectronics.com), where he has been employed for over 17 years. He is a huge OTL amp enthusiast, spending his free time contributing to the Atma-Sphere's Owners Group (ASOG) website (www.otlamp.com). He's also the President of the Audio Society of Minnesota (www.visi.com/~asm). In addition to audio and electronics, Paul enjoys snowboarding, windsurfing, and motorcycling. He has been married to his wife Judy for 15 years. They have a 12-year-old son, Thomas, who, he's thrilled to say, loves music.

With the "hot rod" amp biased at 1A, I achieved the woofer control as expected. The ability to flow twice as much current gave me four times more output power as well. This worked out well all winter, but with about 1200W of heat being dumped into the room, I wanted a different solution for the summertime.

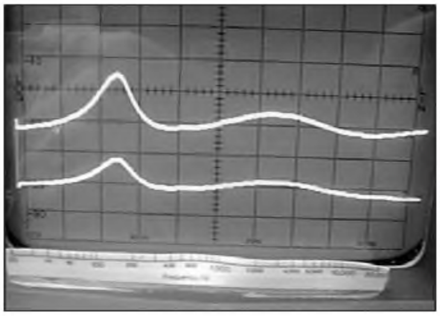


PHOTO 2: Frequency along the horizontal axis (log scale from 20 to 40kHz), and impedance along the vertical axis (2Ω/major line). Zero ohms is the bottom line, 2Ω on the next line above, 4Ω on the next line, and so on. The lower trace is of an actual 4Ω speaker. It exactly crosses 4Ω at the center line (1kHz). The upper trace shows the impedance of the same speaker after its impedance has been multiplied by two (2×) with the ZERO autoformer. It now crosses 8Ω at 1kHz.

AUTOFORMER IDEAS

After learning that autoformers had been used to drive extremely low impedance (1–2Ω) speakers, I tried a wild experiment. In my basement there was a pair of 300VA 50–60Hz toroidal power transformers, which I figured out how to configure into a 2:1 winding ratio autoformer. This would give me a 4:1 impedance ratio and enable me to transform my 4.7Ω speakers into 18.8Ω speakers. I was quite surprised at how well they worked.

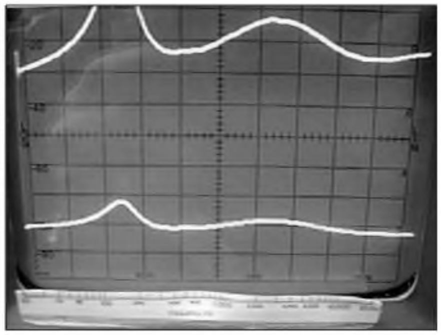


PHOTO 3: Same as *Photo 2*, except the upper trace shows the impedance of the same speaker after its impedance has been multiplied by four (4×) with the ZERO autoformer. It now crosses 16Ω at 1kHz.

Running them full range, I could hear an increase in bass control, and the high frequency was only a bit squirrely. I believed that the bass could become even better if I could get a lower DC resistance value on the autoformer, and the highs would clean up if the autoformer was actually designed for audio. I also tried the same experiment with a 50–60Hz EI-core power transformer, but the results were poor. The EI transformer had no frequency extension. It liked to run only in the 50–60Hz region that it was designed for.

Further pursuit of autoformers gained me a used pair of Atma-Sphere Z-Music (Auto) Transformers. New units were no longer available, so I was lucky to get them. These really worked out well. I used them to give my speakers high enough impedance so that during the summer months I could run the amps with half the stock tubes instead of my double-size version. One of the hardest pills to swallow for an OTL amplifier lover, like myself, is adding "iron" to the system. But since I liked them, I left them connected to my speakers through the summer.

AUTOFORMER BASICS

In general, a power transformer is used to transform voltage and current. An audio transformer is used to transform impedance, but both are really doing the same thing. The winding with more turns will have more voltage, less current, or higher impedance. The winding with fewer turns will have less voltage, more current, or lower impedance.

A typical audio transformer has a "primary" winding and a "secondary" winding. For example, a tube amp transformer may be used to convert a 10kΩ plate-to-plate impedance to a 4Ω impedance for driving a 4Ω speaker. To achieve this 10k:4 impedance ratio, a 50:1 winding ratio is required, due to a square root relationship between the impedance ratio and the winding ratio in a transformer. This typical transformer also provides electrical isolation from primary to secondary so that the output can be electrically floated.

An "auto-transformer" or "autoformer" is the simplest type of transformer. An autoformer has only one winding with multiple taps available. The impedance conversion is achieved by bringing the audio signal out on a different set of taps than the audio came in on. An autoformer can be used any time electrical isolation is not required.

The same impedance ratio to windings ratio rules apply; that is, the winding ratio is the square root of the impedance ratio required. So an impedance ratio of 4:1 (16Ω:4Ω) would be achieved with a winding ratio of the square root of four—a 2:1 winding ratio.

Another way to think of an autoformer is to imagine a typical tube amp output transformer with the primary windings removed. There is only the output secondary winding with 16Ω, 8Ω, 4Ω, and common taps. Now connect a 4Ω speaker on the 4Ω tap, and it will be reflected as a 16Ω load on the 16Ω tap. I tried using a typical tube amp transformer as an autoformer, but, as expected, there was a serious lack of low frequency energy transferred because the transformer was not designed to be used in this manner.

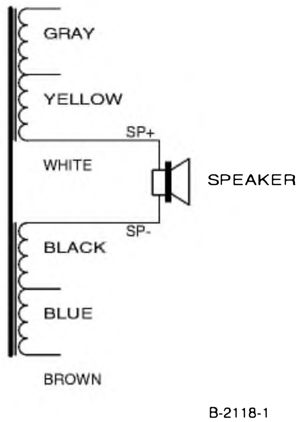
THE AUTOFORMER ADVANTAGE

There are a few general advantages an autoformer has over a typical transformer. First, I usually think that simpler is better, and I believe it applies in this case. With only one winding, the music has the advantage of coming out on the same winding that it goes in on, instead of having to totally pass from the primary winding to the secondary winding. I believe that removing the need for the entire audio signal to leap over the electrical isolation of a typical transformer allows for a much greater transfer of the low level infor-

PHOTO 4: A "typical" tube amp transformer from a Heathkit UA-1 monoblock. Its 57.7:1 winding ratio provides a 13.3kΩ (plate to plate) to 4Ω impedance conversion.

PHOTO 5: A 10kHz square wave signal passing through the properly loaded transformer. The non-vertical rising and falling edges and the complex ringing shows some of the inherent difficulties with this type of transformer.

FIGURE 1:
Speaker impedance multiplier.



MULTIPLIED BY	1.37	2	2.75	4	8	16
SP+	Yellow	Yellow	White	White	Green	Green
SP-	Brown	Blue	Blue	Black	Blue	Black

mation resolved in the music.

Second, the autoformer is "dry"; meaning it is not "soaked" with DC bias current. Accommodating DC current in general limits the frequency extremes. More "iron" needs to be added to keep the core from saturation and a transformer needs to be even larger to do bass well. This larger size tends to limit high frequency extension due to an in-

PHOTO 6: The ZERO autoformer. Using its 2:1 winding ratio leads, a 4:1 impedance conversion is achieved.

crease in inductance, so you need some middle-of-the-road compromise. I am not a transformer engineer, but I think I am safe in stating these generalities. As an example, a SE transformer will typically have a more limited frequency response than a P-P transformer, because it is dealing with greater DC current demands.

Third, the impedance ratio is very small. There is only a 16Ω to 4Ω conversion, not a 10,000Ω to 4Ω conversion. This makes the autoformer's job much simpler, making frequency extension, linearity, transparency, and so on, even easier.

THE ZEROS STORY

The summer went by and, still feeling good about the used autoformers, I responded to a call for help from an OTL'er with the same OTL amps connected to low impedance speakers. Since the Z-Music Transformers were still not available, I pitched the idea to the Atma-Sphere Owner's Group (ASOG, visit www.otlamp.com) that if a few guys are interested, I would "roll my own" autoformer for them. There was enough interest, so I took on the challenge. From my earlier experiments, I decided the autoformer would have a toroidal core. The ability for audio to pass through a toroidal transformer that was designed for 50–60Hz power impressed me. I also knew I was only capable of specifying the design requirements.

After two or three weeks of investigation, I found an audio transformer engineer willing and capable to do the task. I specified the frequency response, DC resistance, impedance of each tap, maximum power, and so forth. The transformer engineer needed to find the proper combination of core size, core material, wire size, number of windings, and so on, to fulfill my design requirements.

THE INDUSTRY MISTAKE

I believe the speaker industry has made a mistake migrating from the vintage norm 16Ω speaker to today's short circuit designs. (A couple of obvious offenders are the Apogee Scintilla 1Ω design and the Apogee Duettas Signatures with a stated manufacturer's impedance of ¾Ω.) I think this has happened because a 4Ω speaker will draw four times more current from a solid-state amplifier than from a 16Ω speaker at the same volume (voltage) setting, thus sucking four times more power from the amplifier (again at a fixed volume setting). This helps the speaker play louder in the showroom when it is going up against competitors. I think "perceived" efficiency helps sell speakers, and this is why speaker manufacturers have gone astray.

This has gotten so out of hand that the speaker industry stopped rating a speaker's "efficiency" and started rating it as speaker "sensitivity," which is how loud a speaker will play at a given voltage rating no matter how much current (and power) it is sucking out of the amplifier.

This aspect of the speaker industry has got to change back. After meeting with and then writing a letter to the editor of *Stereophile* on this issue, John Atkinson has noted in his "Measurements" sections of low impedance speakers that the speaker is actually drawing more power than the 1W standard. It's a start, so good for him. Three examples follow:

1. July '01, p. 65, Martin Logan Prodigy review:
   ". . . indicates it to be, as specified, basically a 4Ω load through much of the audio band. This will mean the speaker will actually draw 2W from the amplifier from that voltage level, not 1W."
2. August '01, p. 70, Krell LAT-1 review:
   "However, as shown by its impedance plot, it is a 4Ω design, drawing 2W rather than 1W from the partnering amplifier to reach this measured sensitivity."
3. August '01, p. 79, Audio Physic Avanti III review:
   ". . . revealed it to a 4Ω design; i.e., it actually draws 2W from the amplifier to raise the rated sound pressure level."

SPEAKER EFFICIENCY VERSUS SENSITIVITY

Speaker efficiency is not the same as speaker sensitivity. Years ago, all speakers were specified as having an efficiency rating, now speakers seem to be specified with a sensitivity rating. I think that the speaker industry needs to get back to rating a speaker's efficiency; or, at the very least, we as audiophiles need to understand the difference.

Efficiency is the true measure of speakers. It is like miles/gallon to the automobile, but speaker sensitivity is like saying how many miles an automobile can go on a tank of gas. If we don't know how many gallons of gas are in the tank, it is meaningless in terms of efficiency. This is the speaker industry's way of making 4Ω speakers with 2Ω impedance dips sound as though they are efficient.

Speaker efficiency is specified as sound pressure per watt (like miles per gallon of gas). In other words, how much acoustical power is put into the air, for how much electrical power is put into the speaker. The rating is typically given as dB per 1W measured at 1m distance.

Speaker sensitivity is specified as sound pressure per volts (like miles per tank of gas). In other words, how much acoustical power is put into the air, for how much voltage is put into the speaker, regardless of its impedance. The rating is typically given as dB per 2.83V measured at 1m distance.

It is true that 2.83V RMS into 8Ω is equal to 1W of power, but now the "sensitivity game" is played and a speaker is said to put out 90dB at 2.83V. The truth is that a 4Ω speaker at its 2Ω impedance dip is requiring 4W of power, not 1W like they would have you believe.

It is true that using an autoformer to transform a 4 or 8Ω speaker into a 16Ω speaker will decrease the speaker's sensitivity because it will take more voltage (but less current) to achieve 1W of power, though the speaker will still remain just as efficient. Actually some systems will have the ability to play louder if the amplifier being used has a maximum power band curve that peaks around 16Ω. Some OTL examples are the 60W Atma-Sphere M-60 that will do 80W into 16Ω, and the 25W Transcendent Audio Stereo T8 that should do 40W into 16Ω. Sure, most solid-state amplifiers can provide more power into a 4Ω load, but do they sound better that way?

My Sony ES series receiver sounds a lot better when I increase my speaker's impedance to 16Ω. I am also able to listen to the music at louder levels without feeling like I want to turn down the volume. Others have reported similar results with their solid-state amplifiers.

A WORD OF CAUTION

The ZERO autoformer holds the speaker's multiplied impedance charter down to 2Hz, yet the low 0.3Ω DC resistance may cause trouble if using a solid-state amplifier with excess offset voltage. As little as 300mV of amplifier offset voltage will cause 1A of DC current to flow. Also, if the solid-state amplifier's offset voltage changes dynamically with the music, a large amount of DC current can flow as well. Because of this, I recommend that a fast blow fuse (probably about 3A) be installed in series between any solid-state amplifier that is not fault protected, and the autoformer. The autoformer will protect the speaker from DC; the fuse is there to protect the amp from itself if it does try to push DC.

I was intuitive enough to set the requirements high so that there could be a large fluctuation in operation conditions and have it still function properly. The transformer engineer was smart enough to implement ideas such as using multiple smaller gauge wires wound together to gain the lower skin effect of the smaller wire, but still have the low DC resistance of a larger wire. A few weeks later I had my prototypes and nicknamed the pair of big round autoformers the "ZEROs (*Photo 1*)."

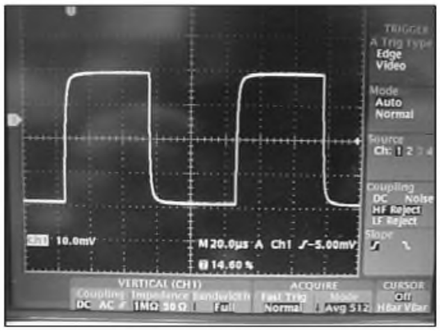


PHOTO 7: A 10kHz square wave signal passing through the properly loaded autoformer. The near vertical rising and falling edges along with the absence of any complex ringing shows one aspect of how much more transparent the ZERO autoformer is than a typical tube amp transformer.

EVALUATING THE PROTOTYPE ZEROS

I needed to make sure that, first, the prototypes achieve the sonic benefits of multiplying the impedance of the speaker load, and, second, they do this without showing any sonic signatures of their own. For three days I simply ran them on my system, played a lot of music, and listened for any sonic clues that told me that I added "iron" to the system.

I invited myself over to my friend Joe's house and ran them on his excellent system, which included an Atma-Sphere MP-3 preamp and MA-1 MkII amps, and a pair of top of the line Magnepan MG20 speakers. Joe was hoping for better bass from his Maggies. Using the ZEROs as an impedance doubler (2×) to make the 4Ω Maggies seem like 8Ω to the amps, it took about three seconds to hear that the bass had come out of the background and became a balanced part of the rest of the music. Joe was thrilled and commented, "I'm sold . . . count me in."

What we started noticing next was an overall improvement top to bottom. A nice drop in distortion and an increase in resolution made it easier to hear a lot more of everything that was happening in the recording. This was not subtle either. We both quite quickly noticed and described to each other the same things.

After some time, we tried using the ZEROs as an impedance quadrupler (4×) to make the 4Ω Maggies seem like 16Ω to the amps. This time the improvement was a little more of the same. If the 2× gave us five steps forward, then the 4× gave us just one step more beyond that. Yet it was enough for Joe to say that he plans on using them that way. In the three hours of listening with them on Joe's system, we were unable to hear any problems with the ZEROs.

In an attempt to find out whether the ZEROs were truly transparent, I listened to music on my system back and forth with them in and out, but compensating for the impedance change of the ZEROs by pulling output tubes. In other words, eight 6336B tubes per amp biased up to 1A directly connected to my 4.7Ω speakers, compared to two 6336B tubes per amp biased at 250mA with the ZEROs used as a 4× speaker impedance multiplier. The thought here is that one-fourth the tubes at one-fourth the bias current driving what seems like one-fourth the load should sound exactly the same if the autoformers are truly transparent. Well, not only were the ZEROs utterly transparent, I have to say the music with just two tubes running through the ZEROs sounded a little cleaner in the bass, and the highs were better focused. *Figure 1* details the speaker impedance multiplier.

I declared the prototypes a success, and had enough ZEROs made up for all the guys who wanted a pair.

THE ZEROS STRETCH THEIR LEGS

The first batch of ZEROs hit the streets, and everybody seemed to love them. Then a second wave of interest came, so I ordered a second batch. It seemed as though people liked them so much they told their friends about them and also tried them on many different types of amplifiers. In just 16 months since the prototypes, I have shipped six batches of ZEROs all over the United States and other countries, including Canada, England, Turkey, Sweden, Scotland, Australia, Portugal, Holland, Puerto Rico, Malaysia, Italy, and Thailand.

The benefits of driving a higher impedance load seem to go beyond OTL amplifiers. I have been told by ZEROs users that they gained sonic improvements on Atma-Sphere (OTL), Transcendent Sound (OTL and P-P), VAC (P-P), BAT (P-P), Audio Research (P-P), Bel Canto (SET and SS), LAMM (P-P), Joule-Electra (OTL), Meitner (solid state), Wright/Sound (SET), Cary (P-P and SET), Tenor (OTL), Carver (SS), Jolida (P-P), Sony (SS), NAIM (SS), Rogue (P-P), Futterman (OTL), Graaf (OTL), Zen (SET), and some DIY SET and OTL designs. It just seems as though most amplifiers sound better when they are driving a higher impedance load.

Although the ZEROs have been used with many amplifiers, I don't consider the ZEROs an amplifier tweak; I consider them a speaker tweak. The ZEROs have been used to increase the impedance of a large variety of manufactured speakers, including Magnepan, nOrh, B&W, Dynaudio, Carver, PBN, Merlin, Martin-Logan, QUAD, Klipsch, SoundLab, Silverline, Vandersteen, Audiostat, Soliloquy, Cabasse, N.E.A.R., Waveform, Shamrock Audio, Audio Physic, Spendor, JMLab, InnerSound, KEF, Medowlark, Alon, Galo, EV, and Wilson Audio.

Typical reported improvements from people who used the ZEROs to increase their speaker's impedance are:

A) Lower distortion
B) Firmer bass
C) Higher resolution
D) More extended and better-focused high frequency
E) Natural, effortless, and organic sound

The combination of lower distortion, higher resolution, and more extended/better-focused highs often resulted in a larger soundstage with better imaging. Greater dynamics and an increase in the instrument's tactile texture seemed to improve as well. Visit my hobby page at www.ZEROimpedance.com for a complete list of testimonial field reports. *Photos 2–7* show the results of impedance measurements.

WHY?

I have to ask myself why does increasing the impedance of a speaker lead to such globally common reported improvements. It goes against my audio philosophy that adding an extra component in the signal path can bring the listener closer to the original musical event. It must be the component added can provide greater benefits than "penalty." I believe the ZEROs are very transparent so the penalty is small.

Imagine the power of "magically" changing the impedance of your speakers. What's left are the gains the amplifier and wires provide when driving an easier load. I cannot prove anything, but I suspect it is simply easier for an amplifier and speaker wire to generate and transfer voltage than current. One watt into 4Ω requires 2V at ½A of current. One watt into 16Ω requires 4V at ¼A of current. That's twice the voltage and half the current for the same amount of power.

Running high voltage/low current lines are a necessity for both the power utilities (high voltage power lines) and commercial audio installers (70V paging systems) where high voltage/low current is generated, distributed, and then transformed to lower voltage/higher current when it has arrived to its destination. The advantages of running low current lines are well understood, so it makes sense to apply it to our hobby as well.

ADD A RESISTOR

If amplifiers and speaker cables like to drive higher impedance loads, why can't I simply add a large 10Ω power resistor in series with my 4Ω speaker? That approach will lighten the load on the amplifier, but bring back the original problem of an electrically underdamped woofer which caused a bloated-sounding bass, and over two-thirds of the audio power will be lost as heat into the resistor as well.

SOME FINAL THOUGHTS

I will never tire of hearing the enthusiasm of fellow audio hobbyists as they report the sonic improvements gained by simply increasing the impedance of their speakers using the ZEROs autoformers. Typically, I just made as many ZEROs as were prepaid by committed people. More recently, I have gone ahead and ordered a few extra pairs so I have some on hand for additional inquiries. Just e-mail me at PaulSpeltz@hotmail.com; with any luck, I will still have a pair available.

The original ZEROs (pictured) were $383/pair, but the added cost of some recent improvements has raised them 50 bucks. The ZEROs now have Teflon insulated silver coated copper leads that are five feet long for a 10′ total span, and now support bi-wiring from the autoformer to the speaker. The new cost of $433/pair includes shipping within the US. For additional information, visit my hobby page at: www.ZEROimpedance.com

I strongly suggest that you consider making your next speaker building project a 16Ω design, so that you get the luxury of high impedance for free. MTM configurations can easily be designed in series instead of parallel to get 16Ω instead of 4Ω. If you are not a speaker builder, I strongly suggest that you consider speaker systems from companies such as Coincident that offer 14Ω models, or use a high performance autoformer such as the ZEROs to do the trick to your existing speakers.

Even 8Ω loads are not optimal. Half of the ZEROs owners used them to increase the impedance of their 8Ω speakers. Multiplying the reported best-sounding ZEROs multiplication factor times the original speaker's impedance, and then averaging all the results, I found the average reported optimal impedance to be 14.6Ω. I will consider my efforts worthwhile when I have raised the awareness of the benefits of higher impedance speakers to the hobbyist, and the idea starts to trickle back to the speaker manufacturers. ❖

A De-emphasis Fix for the D2D-1

Install this quick and easy mod and rest assured your entire CD collection will be compatible with the D2D-1. **By Gary Galo**

The Assemblage D2D-1 Sample Rate Converter was one of the last digital audio products from The Parts Connection. Sadly, parent company Sonic Frontiers pulled the plug on TPC early in 2002 to concentrate on their Anthem line of home theater equipment (I had already written a complete review when we learned of TPC's demise). Chris Johnson, founder and former president of Sonic Frontiers, along with other former SFI/TPC personnel, has started a new company called Parts Connexion (www.partsconnexion.com) to continue their predecessor's audio parts business, but they do not have the rights to TPC's Assemblage line, and will not be continuing those products.

DESIGN OVERSIGHT

If you were among the many customers who purchased the D2D-1, consider yourself very fortunate (especially if you got it at the summer 2001 sale price of $399). The D2D-1 was the hands-down winner in the sample rate converter competition, superior to both the Perpetual Technologies P-1A (reviewed in April 2002) and GW Labs DSP (reviewed in May 2002). This remarkable product—based on Cirrus Logic's CS8420 sample rate converter chip—featured a dual phase-locked loop design, for superior jitter suppression. Its I^2S-enhanced output interface made it a synergetic companion to the DAC 3-series of D/A converters, but even with a standard S/PDIF connection, the D2D-1 sonically outperformed the other sample rate converters I've used.

I found only one quirk during my evaluations of the D2D-1: the unit does not pass the de-emphasis flag on to the D/A converter on CDs with pre-emphasis. Digging back into my own memory banks, I recalled a similar problem with the evaluation boards I received many years ago from Analog Devices for their AD1890-series sampling rate converter chips (see my column "Ask *TAA*" in *TAA* 4/94, pp. 42–44).

Those evaluation boards used the Crystal Semiconductor (now Cirrus Logic) CS8412 input receiver and CS8402 transmitter—these chips were the predecessors of the CS8414 and CS8404A, and have essentially the same pin configuration. The 8412 and 8414 input receivers do not automatically pass the de-emphasis flag on to the 8402 and 8404A transmitters. When these chips are set up in the consumer mode, an external connection must be made in order to pass the de-emphasis flag from the receiver to the transmitter.

Back when I was working with the Analog Devices boards, Walt Jung came up with the solution to the problem. A jumper had to be installed linking pin 3 of the CS8412 input receiver to pin 1 of the CS8402 transmitter. The data sheets for the CS8414 and CS8404A indicated that the fix should be identical for these new chips. Pin 1 of the CS8404A is normally high (+5V). When it is pulled low (to 0V) by pin 3 of the CS8414, the 8404 encodes the de-emphasis flag.

THE MOD

I suggested this to The Parts Connection's Glenn Dolick. He used my D2D-1 as a "guinea pig" and verified that it works. This is a relatively simple modification to perform.

Using a piece of 30AWG wire-wrapping wire (Radio Shack #278-503), solder a jumper from pin 3 of the CS8414 input receiver to pin 1 of the CS8404A transmitter. These chips are clearly identified on the PC board—the CS8414 is U204 and the CS8404A is U404. Both chips are surface-mount types, so you'll need a low-wattage soldering iron (Radio Shack's #64-2051, with its grounded fine pencil tip, is a good iron for surface-mount work) and some fine solder (Radio Shack's #64-035, a 0.015″ diameter silver-bearing type, is ideal).

Cut the wire-wrapping wire to length, tin both ends, and tack-solder each end to the correct pins. Take special care not to make any solder bridges between the pins of the 8414 and 8404A, or between the chips and nearby components. Remove only enough insulation from the wire to make the connection. *Photo 1* shows a close-up of the PC board with this mod installed.

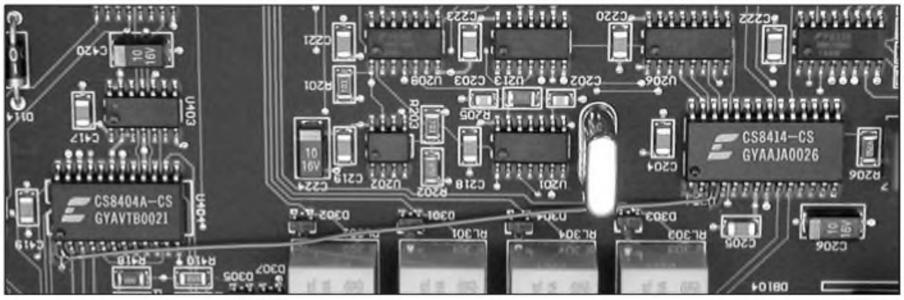


PHOTO 1: Close-up of the D2D-1 PC board with the de-emphasis mod installed. A piece of 30AWG wire-wrapping wire is tack-soldered between pin 1 of the CS8404A transmitter chip (left) and pin 3 of the CS8414 input receiver (right).

The Parts Connection's DAC 3.0 and 3.1 have a de-emphasis indicator LED on the front panel that should tell you whether the de-emphasis flag is being passed on to the D/A converter. The number of CDs recorded with pre-emphasis is relatively small, mostly Japanese discs manufactured by Denon. If you don't own any, you need not worry about the modification. With your CD transport connected directly to a DAC 3-series D/A converter, you can easily identify any pre-emphasized discs, and then decide for yourself if you need to do this mod.

NOT FOR EVERYONE

This modification has one idiosyncrasy. If the second phase-locked loop is not locked, the D2D-1 tells the D/A converter to go into the de-emphasis mode. This is true if the second PLL is disabled with the jumper on the PC board, or if playback of a CD is started before the second PLL has had a chance to lock (two LEDs on the front panel indicate a lock for each of the PLLs). Once the second PLL is locked, everything works fine.

This can admittedly be a nuisance if you are using a DVD player as your CD transport, since most DVD players don't output an S/PDIF signal until a disc has been inserted and recognized by the player. You'll need to insert the disc, wait for the player to recognize the disc, and then wait 15 seconds more until both PLLs are locked. If, for some reason, you need to disable the second PLL, this mod should not be performed.

This peculiarity also makes this mod incompatible with variable-pitch applications. Once the pitch control on a CD player has been activated, the second PLL unlocks, and the D/A converter is in a constant state of de-emphasis.

Proper decoding of CDs with pre-emphasis takes place only when the D2D-1's output mode is set to transparent or 48kHz. With an output sampling frequency of 96kHz, de-emphasis is not correct, even though my DAC 3.0 de-emphasis LED remains lit—there is a +3dB error at 16kHz using my *Hi-Fi News and Record Review* Test Disc II. This has nothing to do with this modification, or the D2D-1.

The Burr-Brown DF1704 digital filter was designed to perform de-emphasis at sampling frequencies of 32, 44.1, and 48kHz, but not 96kHz. The same error occurs when the GW Labs DSP is set for a 96kHz output. I checked the data posted on the Pacific Microsonics website for the PMD-200 digital filter (used in the DAC 3.1), and found a similar situation: de-emphasis is performed only at 44.1 and 48kHz. There's no pre-emphasis specification for sampling rates higher than 48kHz, so there's no point supporting de-emphasis at 96kHz.

Besides, in the 24-bit world there is no need for pre-emphasis. Since I don't have a 3.1 input board installed in my DAC 3.0, I don't know what the errors would be with the PMD-200. The bottom line: if the de-emphasis light stays on after both PLLs are locked, change the output mode to transparent or to 48kHz.

This modification should take only a few minutes to perform, and will make the D2D-1 compatible with your entire CD collection. This mod may also be applicable to other sample rate converter products using the Cirrus Logic receiver and transmitter chips. ❖

High g_m Smart Power Tubes, Part 3

In this last part the author presents five projects that you can duplicate, based on 6C45, 6H30, and 6C46 tubes. **By Stefano Perugini**

PHOTO 4: Top view of the 6C45-ΠE single tube amp.

THE SINGLE-TUBE, SINGLE-ENDED 6C45ΠE AMPLIFIER

This compact amplifier summarizes the enormous potential of the Russian triode 6C45-ΠE (*Photos 4* and *5*). I report the main specs in *Table 1*. The harmon-

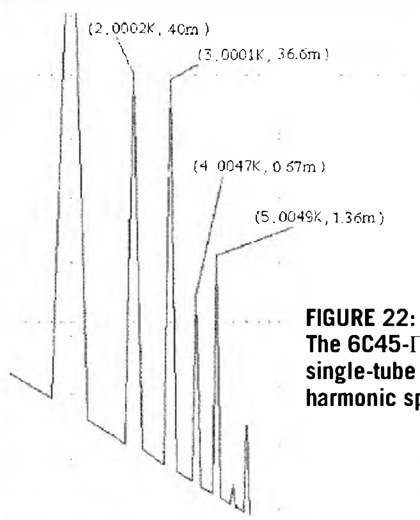


FIGURE 22: The 6C45-ΠE single-tube amp harmonic spectrum.

G-1833-22

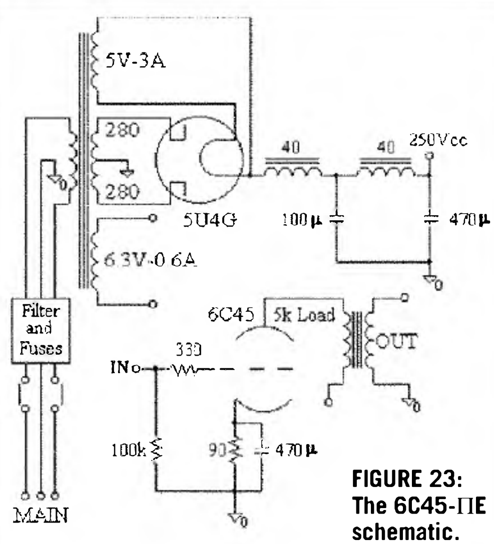


FIGURE 23: The 6C45-ΠE schematic.

G-1833-23

PHOTO 5: Front view of single tube amp.

ic spectrum is in *Fig. 22.*

Looking at the schematic in *Fig. 23*, you can observe a relatively high plate load (5kΩ)—sound quality and tonal imprint are more important than output (although in this project the theoretical maximum power is truly low). Low-power amplifiers tend to have abrupt saturation phenomena. Nevertheless, I have noticed an unforseen capacity of this amplifier to smooth the overload effects.

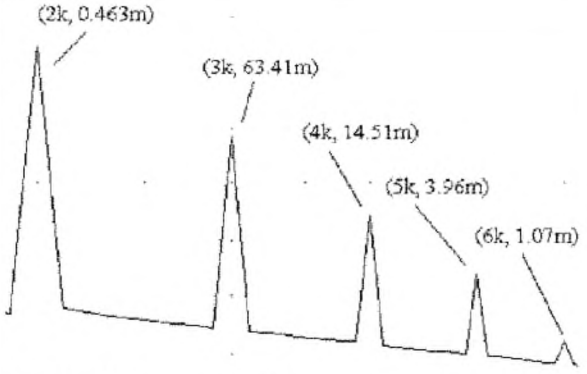


FIGURE 24: Harmonic spectrum. G-1833-24

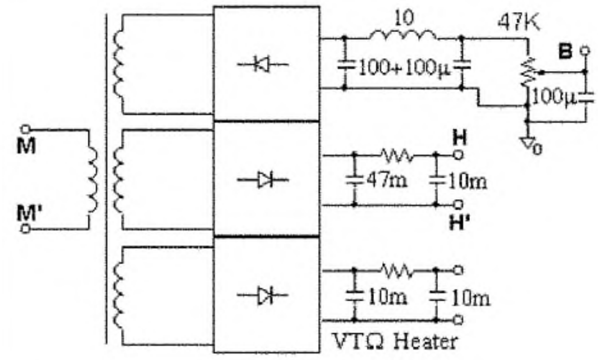


FIGURE 26: Low-voltage power supply schematic. G-1833-26

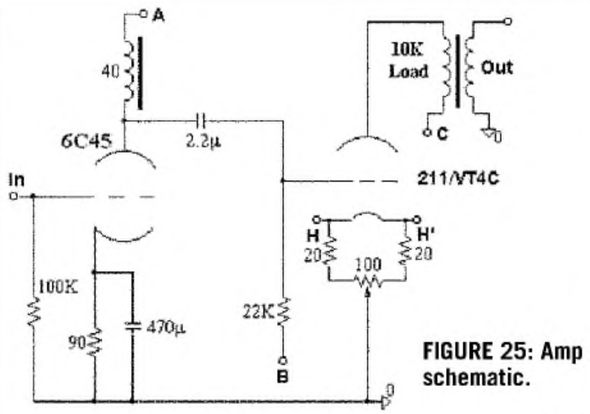


FIGURE 25: Amp schematic. G-1833-25

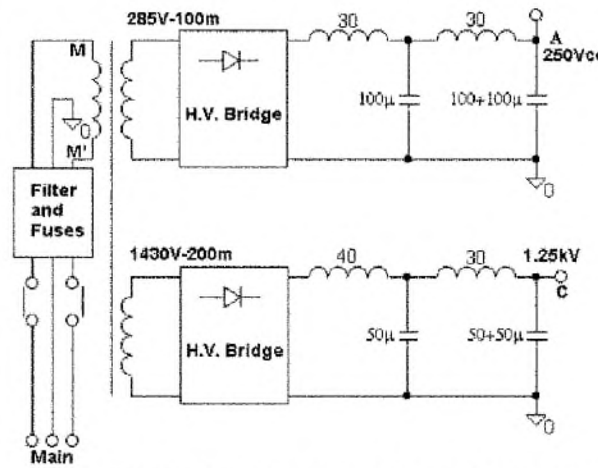


FIGURE 27: High-voltage power supply schematic. G-1833-27

THE SINGLE-ENDED 6C45-ПE/211 AMPLIFIER

The best use of the 6C45-П triode (*Photo 6*) is as a driver for power triodes with high voltage swing requirements as transmitting varieties but also 300B or the new Svetlana SV572-XX. High σ tubes remove the need for a three-stage architecture (gain stage + driver stage + output stage) allowing easy building of a better performing, better sounding two-stage amplifier. *Table 2* shows the unit's specs. The harmonic distortion measurements and schematic diagram—including those for the low- and high-voltage power supplies—are shown in *Figs. 24–7*, respectively.

A SRPP 6H30 LINE PREAMPLIFIER

I have tried the wonderful 6H30 (*Photos 7* and *8*) in a classic context, with great results. The more impressive quality of this shunt-regulated push-pull preamp is its easy driving of any complex load. The schematic (*Fig. 28*) reveals a very simple project in which you can experiment with different topologies as mu-follower and balanced structures.

Due to the low voltage requirement for the 6H30 (plates of upper triodes require only 120V), I use toroidal units as power transformers. I don't show the power supply, but any HV unit with 80–120V, 60–70mA well filtered is OK. Heaters require 6.3V, 1.8–2A.

A 6H30 LOW VOLTAGE OTL SEPP (SINGLE ENDED PUSH-PULL) STAGE

In the recent past, I have designed a Class AB_2 SEPP stage with a low-voltage power supply (*Fig. 29*). The fun-

TABLE 1
6C45-ПE SINGLE-TUBE SINGLE-ENDED AMP TECHNICAL CHARACTERISTICS

Class A1
Power 1.45W RMS
THD 1.25%
Damping factor 4.46

TABLE 2
6C45-ПE/VT-4C SINGLE-ENDED AMP TECHNICAL CHARACTERISTICS

Class A1
Power 19W RMS
THD 3%
Damping factor 3

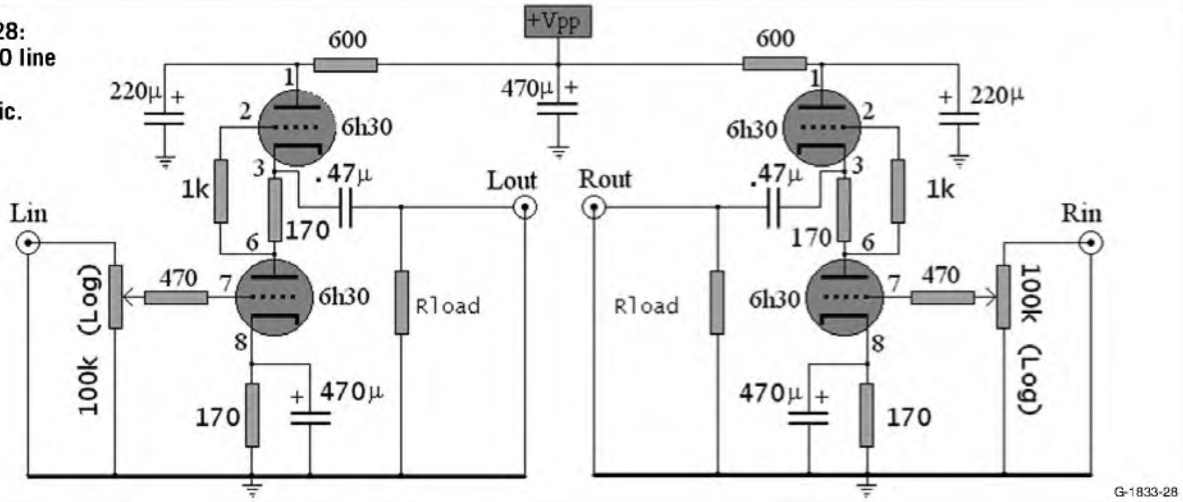


FIGURE 28: The 6H30 line preamp schematic.

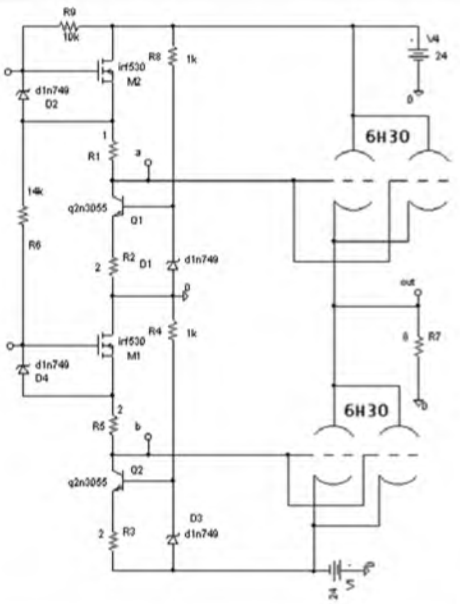


FIGURE 29: The Class AB_2 SEPP low-power output stage.

damental driver stage is designed around a compound N-MOS source follower (*Fig. 30*).

THE 6C46 INTEGRATED PSE AMPLIFIER

At the moment this amp (*Fig. 31*) lies inside a piece of PSpice code, but the results are predictable: A low-voltage integrated PSE amplifier (only 100V) with 2W of maximum power.

A plate load of 1000–2000Ω is OK. The output transformer can benefit from a low cumulative plate resistance rp, but its construction is slightly complicated by the current requirements of the output stage (120mA). The low load for the SRPP reveals the application in this design of the THD cancellation technique.

CONCLUSION

High g_m vacuum tubes have always divided the audio experimenter camp in two factions:

a) Enthusiasts who are pleased by these tubes' unique electrical specifications;
b) Opponents, thwarted by a supposed solid-state-like sound, costs, and apparent unreliability.

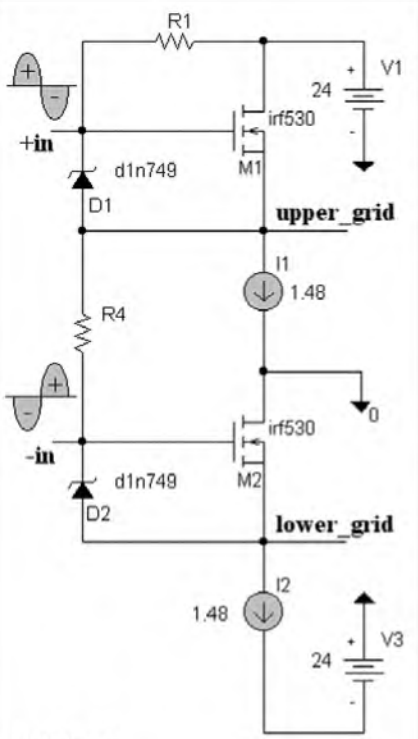


FIGURE 30:
The N-MOS source follower.

G-1833-30

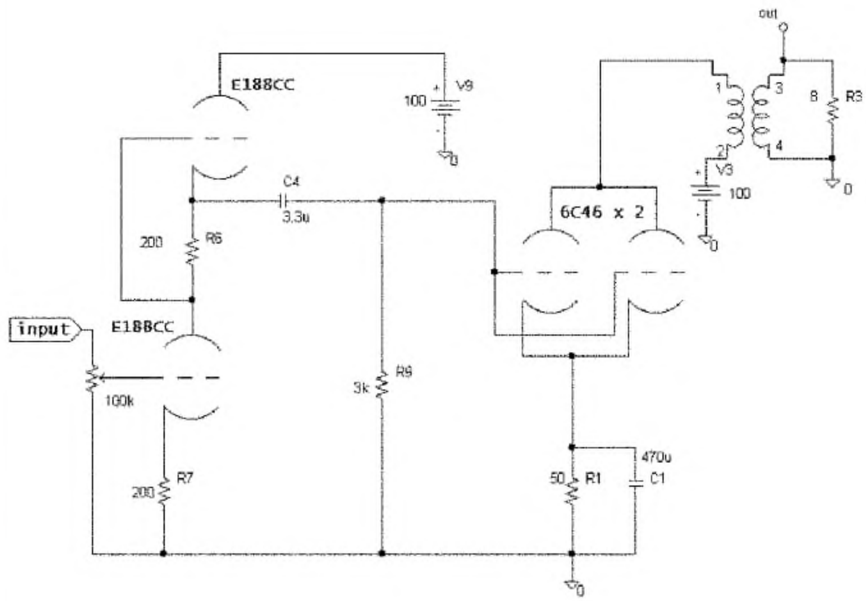


FIGURE 31: The 6C46 amplifier.

G-1833-31


PHOTO 6: Front view of single-ended amp.

PHOTO 7: The 6H30 line preamp (front).

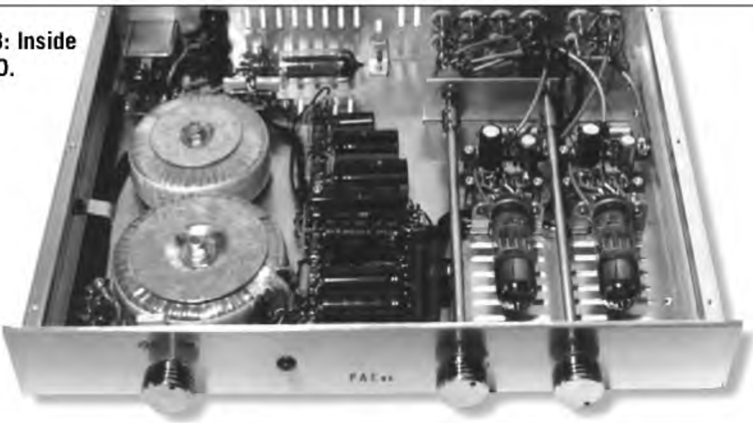


PHOTO 8: Inside the 6H30.

As with any other product of technology, these tubes have their pros and cons. Designing with this tube family requires a lot of caution, but the rewards can be great. ❖

Product Review

Monarchy DIP Upsampler

Reviewed by Gary Galo

Monarchy DIP Upsampler. Monarchy Audio, 380 Swift Ave. #21, South San Francisco, CA 94080, 650-873-3055, FAX 650-588-0335, monarchy@earthlink.net, www.monarchyaudio.com. Price: $299.

Monarchy Audio's DIP Upsampler is the most affordable upsampler yet (*Photo 1*). At $299 factory-direct, it is priced $100 lower than the GW Labs Digital Signal Processor I reviewed in May 2002 (p. 63). The DIP Upsampler converts incoming digital data—whether 44.1kHz, 48kHz, or 96kHz—to a user-selectable output sampling frequency of either 48kHz or 96kHz. The 48kHz output makes the DIP Upsampler compatible with nearly all of the older outboard D/A converters, and the 96kHz output takes advantage of the performance capabilities of more current DACs (see the sidebar, "Upsampling Demystified," accompanying my review of Perpetual Technologies' P-1A in April 2002, p. 52).

In addition to upsampling, this new DIP performs clock jitter suppression,

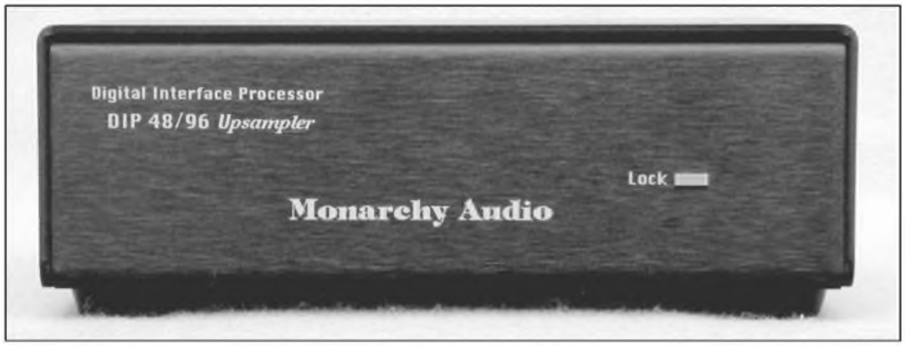


PHOTO 1: Monarchy's DIP Upsampler has no front-panel switches or controls. You can see the digital lock LED on the right.

PHOTO 2: Rear panel of the DIP Upsampler. S/PDIF and Toslink inputs are included, plus S/PDIF and AES/EBU outputs. Input switching is done mechanically, with the rear panel selector switch in the center.

boosting the S/PDIF digital signal, and input-to-output ground isolation. S/PDIF coax and Toslink optical inputs are included, along with S/PDIF and AES/EBU digital outputs. The S/PDIF connections are RCA jacks (*Photo 2*).

Two precision oscillators are used to generate the 48kHz and 96kHz output sampling frequencies. There may be a handful of outboard DACs still in use that will not accept 48kHz data. For such cases, Monarchy offers a high-precision oscillator for 44.1kHz to replace the 48kHz oscillator. The cost is $10 including shipping.

Their one-page sheet of installation instructions doesn't really tell you what you are getting into, however. You must remove the PC board, and the plated-through holes will make removal of the 48kHz oscillator a challenge for novices. If you are still using a DAC that accepts only 44.1kHz inputs, it's probably time for a new DAC!

DESIGN PARTICULARS

Photo 3 shows the inside of the DIP Upsampler. Monarchy's Upsampler is similar to the GW Labs DSP in several respects:

- It uses the Cirrus Logic CS8420 Sample Rate Converter chip as a stand-alone device.
- The common-mode AC line filter is on the main PC board.
- The power transformer is a dual-bobbin type, which helps attenuate power-line noise.
- The two secondary windings on the power transformer feed a pair of Shindengen LN2SP low-noise rectifier bridges.
- Raw DC supplies are L/C-filtered.
- Regulation consists of four 7805-type three-terminal regulators, including one dedicated to the critical PLL supply pin on the CS8420.
- Ferrite beads are used for additional attenuation of high-frequency supply noise.
- Monarchy's "pull-up" resistor technique is used on the Toslink optical receiver.
- Monarchy uses the same high-quality PC-mount RCA jacks as GW Labs.

There are also a few differences between these products:

- Monarchy uses a pulse transformer only on the outputs. GW Labs transformer-couples the inputs and outputs.
- There are no additional ICs—beyond the CS8420—in the DIP Upsampler. The GW Labs DSP has four other chips, which are unnecessary in the DIP Upsampler, since Monarchy does the input switching with a mechanical switch on the rear panel, and the output sampling rate is set with a jumper on the PC board. There are no front-panel switches on the Monarchy.

Monarchy claims signal boosting by a factor of ten. As with the GW Labs DSP, I found the signal boost to be a factor of six, with the DIP Upsampler producing a 3V peak-to-peak output regardless of the S/PDIF input level.

Monarchy makes one claim for the DIP Upsampler that is a bit misleading. They state that the Upsampler "Down samples 96kHz (DVD Audio) for use with all *conventional* DACs (so all DVD, or 24/96 discs can be played on them." The DIP Upsampler is *not* compatible with DVD Audio, however, at least not the DVD Audio standard using Meridian Lossless Compression, and supporting sampling rates up to 192kHz. The DIP Upsampler is compatible with all DVDs using PCM audio with sampling rates as high as 96kHz.

THE SOUND

The DIP Upsampler performs very well, especially considering its bargain price. When connected to my Parts Connection DAC 3.0, the DIP Upsampler offered clear improvements in performance over the DAC 3.0 used as a stand-alone device. It offers a good taste of the benefits of upsampling: improved spatial characteristics, including a wider and deeper soundstage, more precise localization, smoother treble, and greater inner detail.

In my review of Monarchy's DIP 24/96 (Sept. 2001, p. 66), I noted that it was the first outboard processor that actually improved the performance of my DAC 3.0. The DIP Upsampler is even better, and the performance improvement it offers over the DIP 24/96 is easily greater than the $50 price difference between the two products.

I spent a great deal of time comparing the DIP Upsampler to the GW Labs DSP, and found the GW Labs to be the superior performer. It offers a more spacious sonic picture, with more precise localization. The treble is smoother and sweeter, especially evident on cymbals and massed violins. The DSP is also more articulate and less muddy, with greater sense of air in the treble.

Monarchy notes that the jitter attenuation characteristics of the DIP Upsampler complement those of the DIP 24/96 jitter suppressor (reviewed in Sept. 2001), and recommends using the two in tandem—DIP 24/96 followed by the DIP Upsampler—for best performance. I still had the DIP 24/96 on hand, so I tried this scheme and found that, overall, the improvement was readily audible. Compared to the DIP Upsampler used alone, the two in tandem produced a warmer sound, particularly in the midrange and lower midrange. This was most evident on massed strings. Spatial characteristics also improved.

I did note one side-effect of this combo: high-frequency transients are smeared a bit. This is especially evident on cymbal crashes, which lose a bit of their crispness. The GW Labs still gives the tandem arrangement a run for its money, and at a cost of about $150 less.

Here's how I rate the performance of the upsamplers I have evaluated thus far:

1. Parts Connection D2D-1 (my reference upsampler, easily the best I've heard, but unfortunately discontinued)
2. GW Labs DSP
3. Monarchy DIP Upsampler
4. Perpetual Technologies P-1A

The superiority of the GW Labs DSP over the Monarchy may appear mysterious, since the working portions of both products appear—superficially, at least—to be identical. Jon Paul, Vice President and founder of Scientific Conversion, manufacturer of some of the best pulse transformers for digital audio, gave an AES paper that made a strong case for transformer-coupling of all digital audio interfaces. You can download the paper from their website at: http://www.scientificonversion.com/.

One of the key points in the paper is the superior rejection of common-mode signals and EMI suppression with transformer coupling, and a corresponding reduction of jitter. I believe the inclusion of an input coupling transformer in the GW Labs DSP may account for its superior performance.

The Monarchy DIP Upsampler is a fine product, nonetheless, and the clear choice for audiophiles on a budget. At $299, it continues the "best-buy" tradition of the Monarchy DIP products.

Manufacturer response:

As usual, Gary's reviews and comments are accurate and fair.

It's true that the Upsampler's performance can be further enhanced by using a wide band coupling transformer at its input, but this would add extra cost and is actually available from us as an add-on adapter for $49. This is a small [RCA female—transformer—RCA male] plug that inserts into the Upsampler's input as an add-on device.

For best results, just as Gary observed, the Upsampler should be used in tandem with our DIP 24/96 which boosts the signal to feed the Upsampler, in the following configuration:

CD/DVD player—DIP 2496—DIP Upsampler—D/A converter—preamp, and so on.

The DIP 2496's output is transformer coupled. So there is no need to install another transformer at the input of the Upsampler.

However, if the Upsampler is used without the DIP 2496, and if there is no transformer at the output of the CD/DVD transport, our add-on adapter is recommended.

It seems that in order to save costs the new crop of DVD/CD players do not use transformer coupling at its digital audio out any more. A good example is Denon's older model, 3520, which had two separate coupling transformers at its digital outputs. Only a capacitor was found in the newer Denon models.

There is another major difference between the GW Labs and the DIP Upsampler that Gary did not elaborate on: The GW Labs retains the original Red Book standard frequency of 44.1kHz. The DIP Upsampler "upsamples" this frequency to 48kHz (hence the name). On DACs that generate their own 44.1kHz reference clock instead of recovering the clock from the data stream, the DIP Upsampler obviously will not be as compatible as the GW Labs, due to a discrepancy in clock frequencies. A user should base his purchase decision on whether he wants a little "upsampling" (from the DIP Upsampler), or whether he wants a precision clock to replace the re-covered clock (as provided by the GW Labs' DSP). Either way he can expect a significant sonic improvement. ❖

C.C.Poon
Monarchy Audio

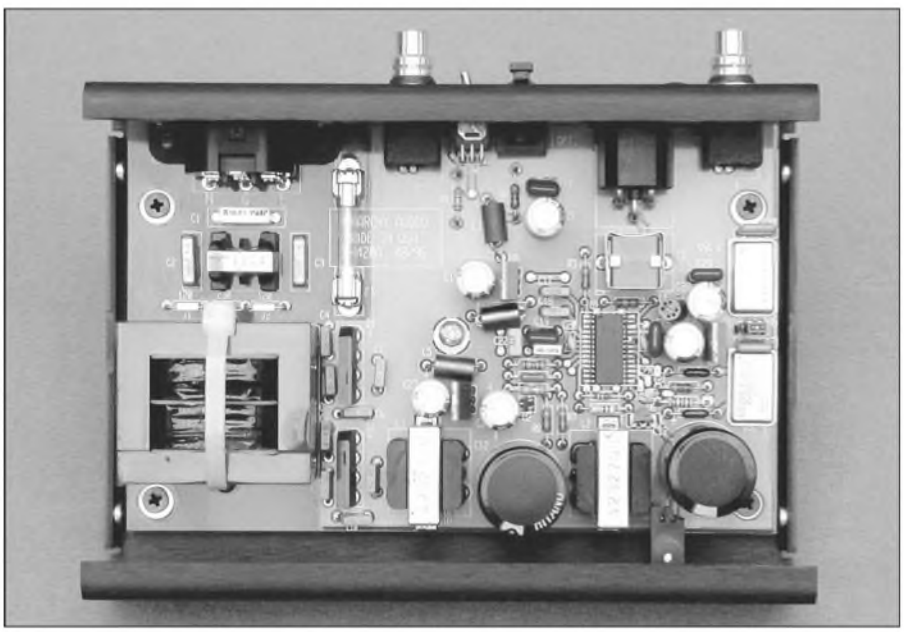


PHOTO 3: Inside the DIP Upsampler. You can see the industry-standard CS8420 Sample Rate Converter chip in its surface-mount package on the right. Four three-terminal supply regulators are included.

Xpress Mail

CORRECTION

Due to a printer's error, we are republishing Charles Hansen's measurements of the S-5 Electronics K-12M power amp from the Nov. '02 issue.

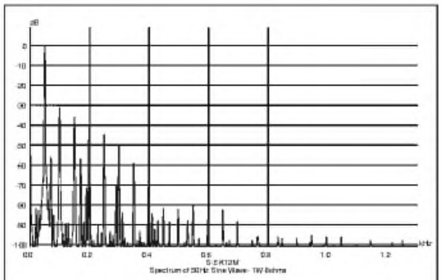


FIGURE 4: Spectrum of 50Hz sine wave.

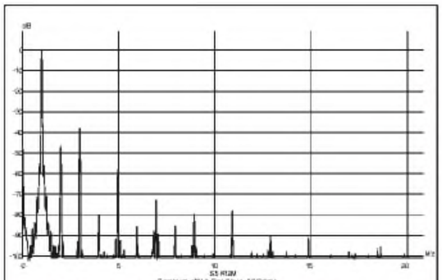


FIGURE 5: Spectrum of 1kHz sine wave.

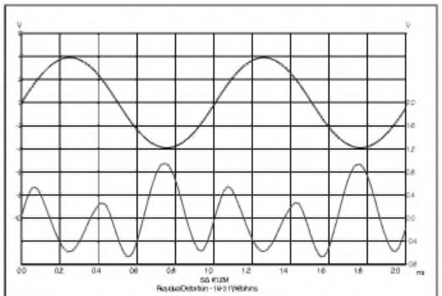


FIGURE 6: Residual distortion, 1kHz 1W 8Ω.

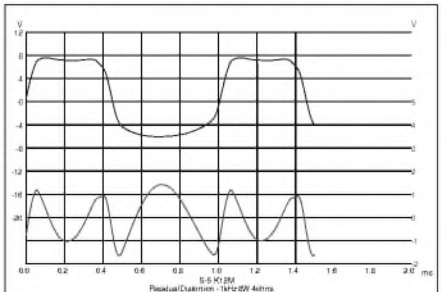


FIGURE 7: Residual distortion, 1kHz 8W 4Ω.

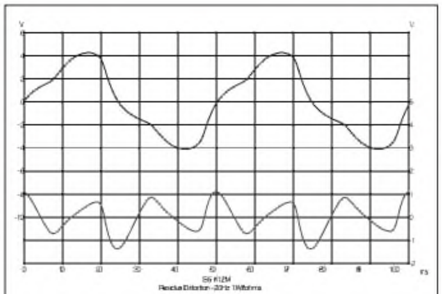


FIGURE 8: Residual distortion, 20Hz 1W 8Ω.

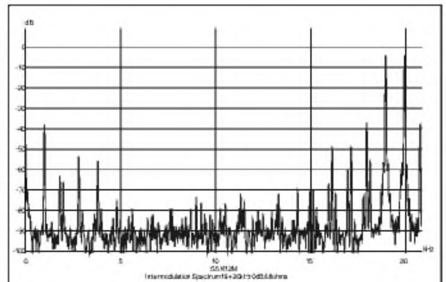


FIGURE 9: Spectrum of 19kHz + 20kHz intermodulation signal.

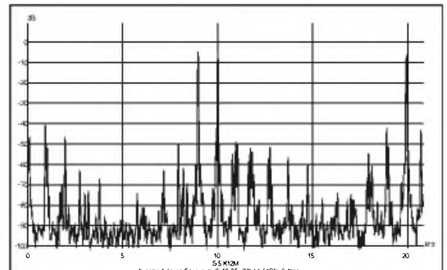


FIGURE 10: Spectrum of 9kHz + 10.05kHz + 20kHz intermodulation signal.

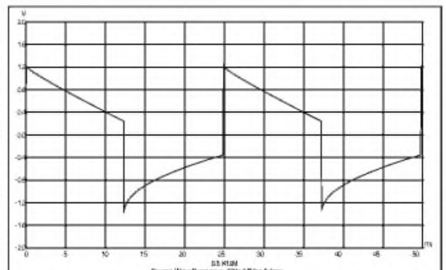


FIGURE 11: Square wave response, 40Hz 2.5Vpp 8Ω.

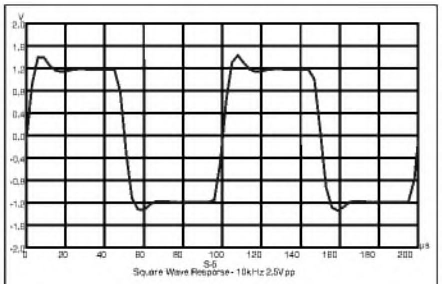


FIGURE 12: Square wave response, 10kHz 2.5Vpp 8Ω.

CORRECTION

In my article "Damping Loudspeakers in Series," published in the Nov. '02 issue of *audioXpress*, there is a minor but unfortunate typographical error that crept in along the way. Reader Ray Segurar was kind enough to bring it to my attention. I thought it might be confusing to some readers.

The fourth equation on page 40 is:

$$Q_E = 2\pi F \frac{2M_M}{B^2 4l^2} 2R_E$$

which is to be simplified by combining terms. In doing so, the fifth equation ended up with an erroneous extra "4" in the denominator. The correct equation is:

$$Q_E = 2\pi F \frac{4}{4} \frac{M_M}{(Bl)^2} = R_E$$

I apologize for the confusion this error may have caused.

Richard Pierce
Hanover, Mass.

MOSFETS VS BIPOLARS

I enjoyed reading Doug Self's latest installment on Load Invariant power amplifiers (March and April '02 *aX*). Doug is diligent in taking many detailed measurements and very good at explaining his findings and theories to the *audioXpress* audience. I was unaware of the MJL3281 and MJL1302 transistors (now available through ON Semiconductor, which spun out of Motorola—www.onsemi.com). Thanks to Doug for bringing them to our attention, with their outstanding f_t of 30MHz.

While Doug is quite right about beta droop being a problem with bipolar power transistors—especially the old ones—there is more to the story. I would like to mention some factors not mentioned by Doug which are equally or more important than beta droop. Bipolar power transistors have two additional major shortcomings: 1) limited speed, which contributes to dynamic crossover distortion at high frequencies; 2) limited safe operating area (SOA), which leads to the need for paralleling output devices and/or the use of output protection circuits.

The distortion improvement which Doug shows, which is most evident at high frequencies and is probably at least as much due to the greater f_t of the devices as it is to reduced beta droop. The old 3055 has an f_t of 1MHz, the better 15024 has an f_t of 4MHz, and the 3281 has an f_t of 30MHz. Even if they all had the same good beta characteristic as the 3281, there would be a dramatic improvement at high frequencies as you progressed from the 3055 to the 15024 to the 3281.

Bipolars are hard to turn off at high frequencies, and this leads directly to what is called secondary or dynamic crossover distortion. It also leads to higher output stage current draw at high frequencies and high levels, and sometimes even to destruction. The output current slew rate of a bipolar is $(2\pi) \times Ib \times f_t$, where Ib is the available reverse turn-off base current. Many designs, including Doug's, do not provide enough reverse base current for output devices with f_t of only a few MHz.

A 5V/μs output into a 4Ω load (corresponding to 40V peak or 200W at 20kHz) requires a current slew rate of 1.3 A/μs. This must be provided by the rate of turn-off of the upper NPN transistor in an emitter-follower design as the signal goes from a positive value toward zero if the stage is to remain in normal Class-AB operation without totem-pole current flow. Even a 20mA net turn-off current with an f_t of only 4MHz will support a turn-off current slew rate of 0.5A/μs.

The limited SOA of bipolars is the biggest reason commercial amplifier designers routinely parallel two to four or more output devices of each polarity. The beta droop issue then becomes moot. If the SOA is not multiplied by paralleling multiple devices, then VI limiting protection circuits (like the one Doug shows) must be used. These trigger and send the output stage into current limiting for many practical speaker loads at modest to high power levels.

They are especially prone to this when the phase angle of the speaker load goes reactive. An inductive "kick" can then result, sometimes destroying a tweeter. In the '70s, when such protection circuits were popular, we often referred to such amplifiers as "tweeter eaters." Given the fairly low IV protection threshold in Doug's circuit, I would hesitate to describe the design as load invariant.

Although MOSFETs are also not perfect devices, they do not suffer from the previously-mentioned beta droop, speed, and SOA problems. Their equivalent f_t is in the hundreds of MHz, and their susceptibility to destruction is almost purely thermal. They do not need SOA protection circuits (this does not obviate the need for some sort of short circuit protection). Bias current stability is about ten times better than that of bipolars.

On the other hand, they tend to be more expensive than bipolars and are not quite as efficient in making use of the power-supply voltage. Their transconductance droops through crossover, leading to some soft static crossover distortion unless they are biased hotter than a bipolar or error correction is employed. On balance, MOSFETs are far superior to bipolars. I've been designing MOSFET power amplifiers since the early '80s, and believe that once you've had MOSFETs, you'll never go back to bipolars.

Thanks again to Doug for a fine article, even if we agree to disagree on a few points.

Bob Cordell
Holmdel, N.J.

Douglas Self responds:

Many thanks to Bob Cordell for his kind words about my articles.

The circuitry I used explored both the EF (emitter-follower) and CFP (complementary feedback pair) output configurations, but concentrated on the former. It is not as widely known as perhaps it ought to be that the EF circuit comes in at least two versions. In the most common, the driver emitter resistors, through which base turn-off currents must flow, are connected to the output rail. Obviously, there can never be more than 0.6V available to drive turn-off current through the resistor.

The less common—and, in my view, superior—version has a shared drive emitter resistor with no output rail connection; I called this Type II in my book. The Vbe and power-device emitter-resistor voltage drop of the side of the output stage that is turning on is added to the voltage pulling current from the base of the transistor turning off, to the point that the off transistor can have its base reverse-biased by a couple of volts. I therefore always use this configuration for an EF output stage, and this is one reason why I believe that switchoff distortion does not intrude significantly into the results I obtained.

Another reason is direct visual evidence; I always keep a wary eye on the distortion residual when amplifier-fighting. Beta-droop effects always give low-order distortion, whereas in my experience switchoff distortion looks very spiky and is more likely to be confused with "normal" crossover distortion.

It is true that bipolar transistors have a more restricted SOA in the sense that they have a second-breakdown region that inconveniently lops a corner off the SOA in the high-voltage, low-current region. This is certainly a nuisance, especially with reactive loads that put more voltage stress on the output devices.

I quite agree that real-time VI limiting can give rise to some horrendous cracking noises if it is activated by an inductive load characteristic. The output voltage stops dead and heads briefly for the opposite rail, where (you hope) the output clamp diodes arrest it before the output devices complain. I have never known this to pop a tweeter myself, but given a powerful enough amplifier in relation to the speaker, it could certainly happen. (My own preferred method of blowing tweeters is sustained sine-wave testing.) It seems, however, that some form of VI limiting is often a commercial necessity to get the maximum power out of a given investment in output silicon.

The need to parallel output devices for high power is true of both device technologies. With bipolars I have always found this straightforward, because of the predictable Vbe characteristics. MOSFETs, however, have serious variations in the Vgs required for a given current, and in my experience—which I accept may not be definitive—some really tedious device matching is required if device currents are to be even roughly equal. I am aware of one design with 16 paralleled MOSFETs, which brought its manufacturer to bankruptcy because of the near-impossibility of matching 16 MOSFETs with sufficient accuracy.

However, in my opinion the worst aspect of the MOSFET is the highly non-conjugate conduction law, which, to put it roughly, is a square-law with a sudden start that blends into a linear characteristic. In other words, if you butt two of these together in a push-pull output stage, there is no bias setting that gives anything like constant gain versus output swing.

The bipolar, on the other hand, has an exponential conduction law with a very, very smooth start. Exponentials are not quite conjugate, but the modest size of the "gain-wobbles" diagrammed in the articles shows that there is an optimal bias setting where you can get very close to constant gain. The bipolar also has stacks more transconductance, which can be pressed into service giving linearizing local feedback.

Having designed amplifiers using both types of devices, I did go back to bipolar country, and I find it works for me!

THOR MEASUREMENTS

I'm currently building a professional monitor utilizing the superb SEAS Excel tweeter (with ATC "Super-Dome" midrange driver and ATC 12″ under-hung woofer). So, my jaw dropped when I saw Dr. D'Appolito's measured frequency response for that tweeter by itself, and then of the final THOR system (May '02).

My tweeters measure almost exactly as specified on the SEAS data sheet. They have a −3dB linear downward slope as frequency increases from a peak around 4kHz up to 11kHz, then a broad, smooth +2.5dB bump centered around 15kHz. D'Appolito measurements don't show these characteristics at all. In fact, he shows a shocking, ruler flat response across the entire bandwidth of his system. (The B&W 800/801/802 series and Dunlavey V and VII are the only passive professional monitors I've ever heard with response that flat!)

The SEAS "kits" page on their website (http://www.seas.no/thor.htm) indeed shows a much less flat response curve for the THOR system than that in the *audioXpress* article—one that is exactly what I would predict from my own measurements of the tweeter (and SEAS data sheet for it), and the crossover Dr. D'Appolito has designed. Can you please reconcile the difference for me? As an engineer who does CD mastering, I can tell you the difference in sound between D'Appolito's measured response and that published by SEAS would be huge.

A note to potential kit builders: The Excel tweeter is spectacular, as are the SEAS magnesium woofers, which I've heard in a commercial speaker. Crossover integration shown on the website is just as excellent as the article notes. So, I strongly recommend you build this design. You can compensate for the downward slope of the tweeter response by placing a resistor in parallel with one of the series caps in the tweeter crossover, as I do in my ATC design. (It's a second-order L-R filter at 3.5kHz, however, so I can't suggest values to try for THOR.)

Bob Sykes
San Francisco, Calif.

Joe D'Appolito responds:

I received four early-production samples of the T25CF002 tweeter. I tested all four and selected two that matched closely and also happened to have the flattest response. Having said that, you must always be careful when comparing frequency response data taken at different facilities. Not only do equipment differences come into play, but also the test environment and tweeter mounting have a strong effect.

I measured tweeter response with an ACO 7016 ¼″ mike. The calibration chart for this mike shows it to be flat within ±0.25dB out to 20kHz. I used the mike with a custom preamp to feed the MLSSA acoustic measurement system, and placed it on the tweeter axis at 1m and took a quasi-anechoic response that is valid down to 200Hz. I am very confident that what I measured is correct. So where does the difference arise?

I am not familiar with the equipment SEAS or Mr. Sykes used to get their data, but I do know that SEAS measures their tweeters on a large baffle. This effectively eliminates edge diffraction effects.

For the THOR design I measured the tweeters in the THOR enclosure which has a 9″ wide baffle. You can clearly see the effect of *the narrow baffle in my Fig 9. There is a 3dB dip at 3kHz and milder dip at 4kHz. These dips occur just where Mr. Sykes and SEAS measure a peak. This effectively eliminates the peaks and flattens the tweeter response.*

I read with great interest the article about the first scientifically designed transmission line speaker. I really believe you've done a great job producing the THOR speaker. However, I have some doubt about the low-frequency capability, although it was rated 9 (10) in the sonic summary.

You say that the output from the cone is approximately equal to, as if the driver were mounted in a closed box with the same volume. (I understand that there is an additional boost from the rear output.) The volume of the box is 41 ltr per driver. When examining the driver in a speaker simulation software, you will find that the cone excursion is very high at low frequencies for such a big closed box, severely limiting the maximum output.

There is still an open question whether the transmission line will limit the excursion. I haven't found any investigations on this topic for transmission lines, not even in the references pointing to G. L. Augspurger.

Mats Blomberg
Taby, Sweden

Joe D'Appolito responds:

Excursion does increase below resonance in much the same manner as with a vented system, but it is quite a bit better in that the driver is very heavily loaded with the resistive impedance of the line. Your simulation must mimic a very lossy line. To get the same effect with a closed box simulation, you would need to reduce box Q to about 0.5. This would come close to simulating a lossy nonresonant TLine. Unfortunately, many simulation programs do not allow you to enter box as an independent parameter.

Editor's comment: Correspondents should please note that until a simulation is built with real, physical materials, the results are no more reliable than the quality and accuracy of the software and the user.

With reference to Joe D'Appolito's THOR article, I have a query regarding the calculated volume Vp using the internal measurements in the drawing provided on page 10 and that calculated in column 3 at the top of page 15. With the measurements shown in the drawing I calculate the internal volume as 3567 in^3 {(11.75″ × 7.5″ × 41.25″) − (7.5″ × 3″ × 3″)}—this allows for the two 45° filleted corners but not the volume of the divider panel—whereas the calculation $V_{AS}/0.9$ (74/0.9) produces a volume of 5021 in^3.

Am I missing something or is there an error in the article?

Ross Herbert
Carine WA, Australia

Joe D'Appolito responds:

I am afraid you did miss something. If you read the last paragraph on page 16 and the following paragraph on page 17 you will see that the calculated volume was rejected as too large for a practical system and a much smaller volume chosen. Fortunately, the compromise had little effect on performance.

HOW DOES IT SOUND?

I love your magazine in many ways, but am becoming rather disillusioned about some of your articles. It seems more and more do not have any comments on how the designs compare sound-wise with what is "state of the art." In other words, how the project sounds, including comparisons to other acknowledged "great" equipment or in several people's opinions. This may be subjective, but it is needed info for readers looking to build something.

Most people who build amps themselves are looking for a great-sounding design that will be worth the effort and expense. Many cannot afford to build repeatedly, so the first shot needs to be good. Tube designs seem to be the articles most lacking in this aspect.

For example, "A Practical Circlotron You Can Build" (May '02, p. 34) contains no comments at all about the sound of the various designs given—not even the ones the author says he has built. I have no idea about which one I would want to build, since I have no information about the sound of the variations. Must I assume that the overall design has good sound, or the author would not have bothered to publish it?

Greg Starkey
Waukesha, Wis.

P.S.: I have both solid-state and tube hardware I have built from schematics and realize amps can sound and perform quite differently.

John Stewart responds:

Thank you for your response to my recent article covering a Circlotron Amplifier. Some of my comments in regard to your concerns may surprise you and others. I have some very strong feelings about the way this or that amplifier may "sound."

First of all, let me preface my comments as follows. I don't think I am qualified to tell you how any of my amplifiers "sound." I would probably tend to be biased in a direction that might make them sound better than others would say. I don't have any friends or acquaintances who are professionals in the audio field who might pass judgment. However, those who have listened have had positive comments.

As well, I can think of at least three filters we often tend to ignore when evaluating the "sound" of an amplifier, even though they are with us at all times. Between the output terminals of every amplifier and our ears—whether it be Circlotron, McIntosh, or our kitchen radios—we have a loudspeaker, its enclosure, and the listening room. Each of these has a profound effect on what we hear, in particular the speaker. At the present state of the art the loudspeaker appears to be the weakest link in the reproduction of sound.

For example, I have an old (50 years?) AC/DC, AM kitchen radio connected to a good 12″ speaker, properly baffled in my garage. The radio has an RF stage, and we are far away from electrical disturbances, so that the program material is reasonably good by the time it reaches the power stage.

As well, the power stage has been "smiled upon." I have replaced the usual 35L6GT output tube which provides no speaker damping at all with a feedback pair consisting of a 12J5 and 50L6GT. The 12J5 plugs into the rewired socket where the 35Z5GT rectifier used to live. The silicon diode replacing that results in a higher B+ so that the output power is about doubled.

I've done several of these in the past, all with considerable improvement in the sound, but only when connected to a good loudspeaker in a proper enclosure. To me these all sound good, but I and my ears are becoming old, so you can draw your own conclusions!!

I built only one Circlotron amplifier, but in such a way that you could easily try many combinations of the various topologies and components available. I wanted to offer something to the experimenter that would allow a wide choice of tubes as well as the ability to connect for triode or pentode operation. You can also operate with or without feedback. Choose which combination suits your taste best by referring to the comprehensive set of performance result tables.

Now, how did the Circlotron "sound"? Again, to me very good. It may sound even better to you, but of course that will depend very much on your reproducing equipment and listening room.

My speakers are by no means even close to the best, but they are nevertheless quite reasonable in my opinion. They are each about 3′ high and about 1.5′ square. Inside each is a 12″ woofer (with a large magnet), a 4″ midrange and a 2″ tweeter. The crossovers are adjustable. They are now about 25 years old, but soldiering on.

The listening room measures about 16 × 36 feet, so low frequencies are easily repro-

duced. I have a good music library to compare with (about 1000 vinyl discs and even some 78s). I also have four large wide-range speakers outside aimed into the back "40" here. Luckily, my closest neighbor in that direction is about one mile away.

There are probably a few out there who can actually hear the difference between this or that amplifier through all of the filters I've mentioned above. That would include professional musicians whose living depends upon detailed recognition of various sounds. Technicians and those in the business of manufacturing and repairing amplifiers as a living would have the opportunity to hear many amplifiers and be in a good position to make valid comparisons. My background is hi-tech, but not in the sound business.

Having said all of that, for me design of the various circuits involved is my primary interest. Many years ago I started out as a circuit designer in the realm of tubes and now find considerable satisfaction in pursuing that activity again. Several amplifiers have resulted from that activity, some of which you may have read about in past issues of these pages.

What happens to the amplifiers when the article is finished? Sometimes they are cannibalized for the next project! Others are sitting here on shelves in the workshop.

There are many trade-offs in any system. It is the designer's task to separate what is important from what is not. I recommend anyone putting together a sound system to pay special attention to the loudspeakers, their enclosures, and the listening room. The reward will be a good performance whether you choose a single-ended triode, ultralinear, McIntosh, or any of several other amplifier topologies.

SATISFIED CUSTOMERS

I just bought the assembled and tested version of the 12M tube integrated amplifier designed and built by George Fathauer and S-5 Electronics and advertised on page 45 of the June '02 issue. I was compelled to write this letter because this tiny 5 lb, 8W per channel miracle is nothing less than that—an absolute miracle. How this kind of amp can be designed, built, and marketed for this kind of money in today's audio marketplace is another miracle altogether. This amp must be heard to be believed.

For those who have not taken advantage of the opportunity to see and hear the 12M, it is based on the 11MS8 output tube, which I am told is normally used as a television tube. There are four 11MS8s used in the 12M and the circuit looks about as simple as it can be, which I am also told was the designer's intent. I am far from expert in the field of audio, but my ears have become familiar over the years with what quality amplification sounds like, and the 12M astonishes me more every time I listen to it.

I hope the readers of *audioXpress* and fans of genuine audio bargains will consider giving the 12M an audition. It can be ordered in kit form or assembled and tested. I also just subscribed to your magazine. Any publication that brings me news of products like the ones being built by S-5 Electronics is a publication I want to get regularly.

Jim Eatherton
Sutherland, Oreg.

Note: Tests of the S-5 were published in the Nov. '02 issue of aX (p. 47).*—Ed.*

EDITORIAL RESPONSE

Thank you very much for the nice presentation of my five-channel amp (June '02). It was very well done and I'm sure that anyone who tries to duplicate it should have no problems. But if they do I will be glad to assist them in any way I can.

I really enjoyed your editorial about advertisers and I'm glad you have enlisted SMM as your new representative. I purchase about 95% of my supplies from your advertising clients! I am always building or repairing something for myself or others and I never buy at "surplus" stores. I have in the past and found that a lot of their materials are not very trustworthy, being so outdated, and they never(!) take back any defective parts.

My experience with AES and others that advertise in your magazine is that they are always very professional and willing to correct any problems. As a matter of fact, I just recently acquired the materials from Antique Electronics to start the construction of a 6B4-G push-pull triode amplifier and, again, their service is great, and Steve in the order department is fantastic to deal with! This is the only style of amp that I have never built and I'm excited to see how it will sound. Many hobbyists think it is difficult to find a power transformer to supply the 320V needed, but I find that if you use the Hammond #272JX wired as a choke input load or the Plitron wired as a capacitor input load, the voltages come out just fine. I'm going to be using the Hammond because I like the classic look.

I was just sitting here listening to TBM play Errol Garners "Misty" on a Hammond organ and getting ready to start my project when I decided to write you this letter. I wish you the best in all of your endeavors and I thank you for your expert knowledge concerning the magazine business. Without you we hobbyists would surely find ourselves lacking—in more ways than we can imagine!

Rick Spencer
Clovis, Calif.

BALANCING ACT

I thoroughly enjoy *audioXpress*! As a former *Speaker Builder/Glass Audio* subscriber, I must say I have been very pleased with the bringing together of these formats. All of you continue to do a carefully planned balancing act in terms of including "something for everyone" in each issue.

Based on letters in some issues, it must be impossible to please everyone all of the time, but I always look forward to each and every issue. Even if there are articles that may not be of immediate interest, they are always educational and informative. In building some of the published circuits, for example, I have taken what I thought to be interesting ideas from one schematic and incorporated them into another.

Erik Mandaville
Missouri City, Tex.

MINI-TYPE

My primary reason for writing is to register a complaint about the size of some of the print in your articles; specifically, the one beginning on page 40 of the September issue. The article appears to be interesting, but I can't really tell since I can't read the print in the figures. I would guess that I'm not alone; I'm getting up in years and need reading glasses. Occasionally, when working on a circuit board or a model, I must resort to additional magnification and added light, but with this article, even that didn't help. For instance, I can't find node 188.

It seems to me that it would make sense to expand figures and schematics to the point that more of us older readers could see and read values and component designations without having to find a magnifier. I get a couple of other technical magazines and rarely have trouble seeing what they present, and they are free!

Bill Macy
Billikin3@cs.com

HELP WANTED

I want to replace and upgrade the 12″ woofers in my Transaudio 1012B speakers. To do this I need to know the frequency response of the original woofers.

Each speaker has a 12″ woofer, 4″ midrange, and 3″ tweeter. Upon checking, the only thing I can tell you about the woofer is that the number "GW1204" appears on the woofer's frame. Other than the part number on each speaker, the only numbers inside, or outside of the cabinet is the name Transaudio, the model number, and the serial number, but no other numbers.

The Transaudio 1012-B was produced in 1978. The 1012-B's "baby brother" was the 1010-B, which featured a 10″ woofer.

I didn't see anything else from Transaudio after 1980, I think. Being that Transaudio has been gone for over two decades, I doubt any information is available for my 1012-Bs, but I thought I would write you and find out—just in case you could help me. If you can't then perhaps you know of a source which I can contact.

Any information or advice you can give me will be greatly appreciated. ❖

Neal A. Haight
4516 Hillsborough Dr.
Castro Valley, CA 94546

Readers with information on this topic are encouraged to respond directly to the letter writer at the address provided.

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AD INDEX

Yard Sale

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Glass Audio back issues: Complete sets—1993: 4 issues, $12 shipped; 1995: 6 issues, $15 shipped. Also, 1989 #2, $6 shipped, and various issues (call/write for list) from '92, '94, '97, and '98: each issue, $2 plus nominal postage. John Agugliaro, 845-947-2748, JAGUGL4545@aol.com.

12 Radio Shack 4″ woof/mid #40-1022, $6 each, brand new—never used; 2 Madisound 6.5″ woofers 6204R, $15 each, w/metal grilles, brand new—never used; 2 JBL 10″ woofer model 127A, $20 each, used, but in very good shape; 2 home brewed subwoofers, both self powered; 1 pair of Allisons vintage speakers, triangular shaped w/2 sets of three way each. jmateus@tellurian.net, or call 973-579-7849.

TTs: Thorens TD150 with Empire 888; Technics SLD22 with Audiotechnica Series #3. Receivers: Pioneer SX434; Lafayette LR 100. Int. Amp.: Marantz 1122dc. Speakers: BIC Ventura Formula 2; Fisher Heritage series XP-10, 3-way, with 15″ woofer—all magnet structures 5# or more. Offers invited. dmeraner2@att.net.

WANTED

Audio Engineering magazines (old name for *Audio* magazine) prior to Feb., 1948. E-mail: EsotericTT@aol.com, or call 708-431-4560. Mike Stosich.

Radio Shack/Lineaum tweeters or parts. Charles King 203-499-2461 or stellavox@excite.com.

Carver M400 magnetic field power amplifier. Audio transformers and bias supply transformers for ESLs (ElectroStatic Loudspeakers). Contact Angel Rivera 941-798-9227 (Florida), e-mail: alrivera@tampabay.rr.com.

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Tools, Tips, & Techniques

Quick Desoldering

I needed to repair a graphic equalizer that had a shorted op amp. Because the unit had 20-odd ICs all soldered directly into the PC board, this was not a simple job. I pulled out my trusty desoldering iron, an MCM 21-945, and set to work. After removing two ICs—and 30 minutes later—I thought "there has got to be a better way!" This is what I came up with (*Photo 1*).

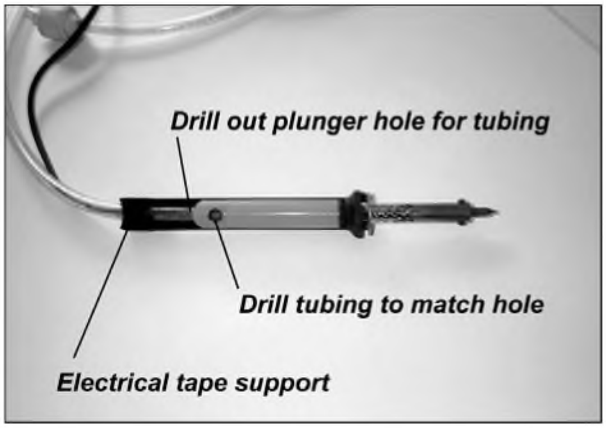

PHOTO 1: Modifications to desoldering iron.

PHOTO 2: "Hi-tech" vacuum interface.

I had on hand some flexible, plastic, 5/16″ ID tubing. Stripping down the desoldering iron, I discarded the plunger assembly and the release button. I drilled out the hole where the plunger shaft had exited the barrel—to make a very tight fit around the outside of the tubing—and reassembled the iron. I wrapped masking tape around the other end of the tubing until it fit tightly into the hose of my workshop vacuum (*Photo 2*). I added a length of electrical tape to support the tubing as it left the iron. I was done—or so I thought!

Trying it out, I realized the iron wouldn't melt solder. All the air passing through the tip was cooling it too much. I needed a way to control the airflow, allowing it only when I actually wanted to remove solder. I drilled a hole into the tubing the same size as the hole where the release button had resided. This did the trick.

I melt the soldered joint then block the hole with my finger and away the solder goes. It took only 20 minutes to remove all the other ICs, with some of them just falling out of the board. This unit has saved me countless hours since I did the modification. ❖

Paul Whiteman
Toronto, Canada