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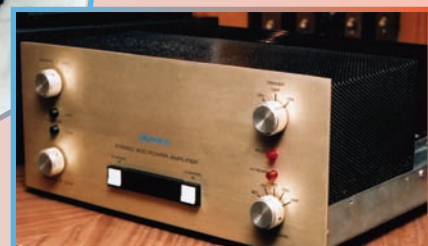
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Tube Amp Design Breakthrough

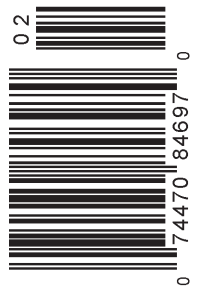


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Clamped Bias, AB2, RC Coupled Amp

By Jim Carlyle

Introducing a world-first performance breakthrough in tube power amp circuit design.

For 45 years I have been lamenting the impossibility of bias-stabilizing an over-driven audio amp. The grid current of attracted electrons always shifts the operating point towards current cutoff until serious cross-over distortion thoroughly spoils the fidelity. Coupling transformers are the obvious solution, but they limit negative feedback, take up space, and cost big money for two wideband low-distortion units, such as Sowters of England.

Well, not anymore. One day—during weight-training—I realized that in nearly all push-pull amplifiers the negative peak voltage swing on the grid is not actually doing anything useful because that tube is already cut off. Bingo. Connect the grid through an appropriate zener to earth, and the built-up electrons on the coupling capacitor will shoot to earth in spurts on the negative voltage peaks.

Yeah, it works, eliminating the high-drive-induced zero-crossing kink and giving more power to the voice coil, because now you can increase the drive beyond AB1 right into AB2! In fact, as my oscilloscope photos show, you can even choose a zener value to make the average grid voltage drift less nega-

tive(!) at high power, rather than more negative! Please yourself.

HOW IT WORKS

Figure 1 shows the basic schematic of the system. The plain diode D prevents

the positive-swing grid drive from shorting to earth through the zener. Separate diodes are advised for parallel tubes. The resistor R reduces the transient load discontinuity seen by the driving valve, which can cause ringing



PHOTO 1: The completed tube amp.

PHOTO 2: An underview of the 550mm × 330mm 37kg behemoth, showing a newly installed preamp filter choke below the top left corner.

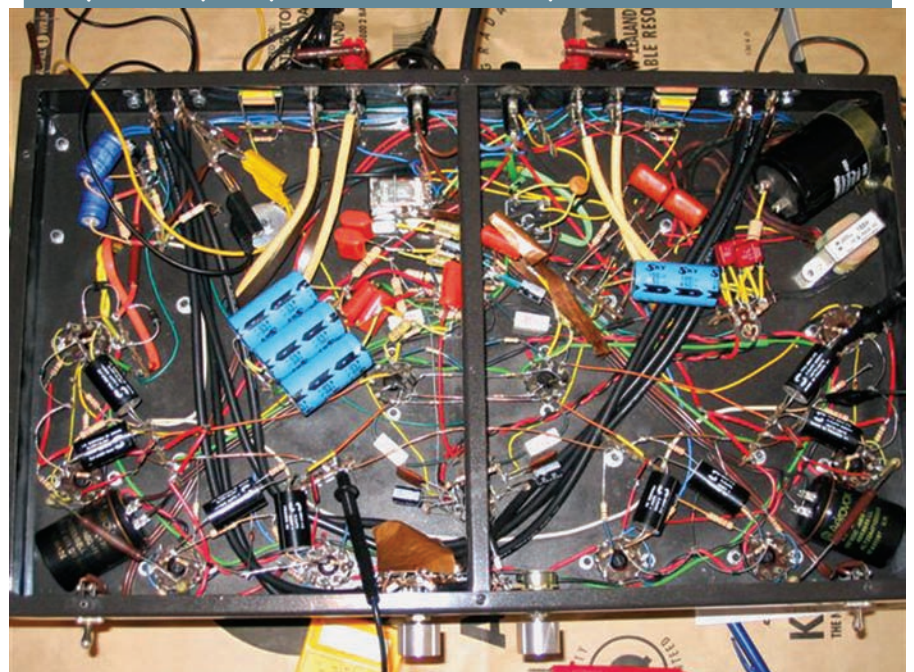
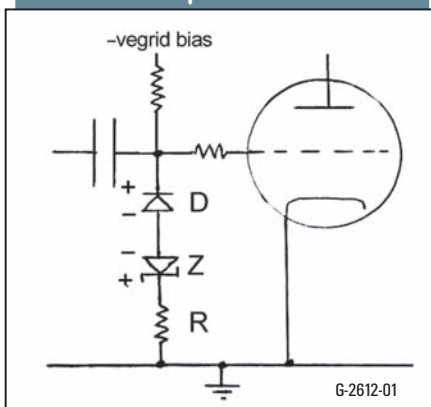


FIGURE 1: Simple circuit mod.



around the feedback loop. Try about 1k Ω per grid, as with my 330 Ω for four grids.

Figure 2 (a,b) is my full amplifier schematic included in the spirit of full disclosure, in the name of investigative research, where scrutineers want the complete picture.

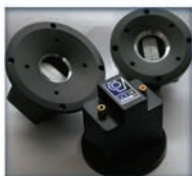
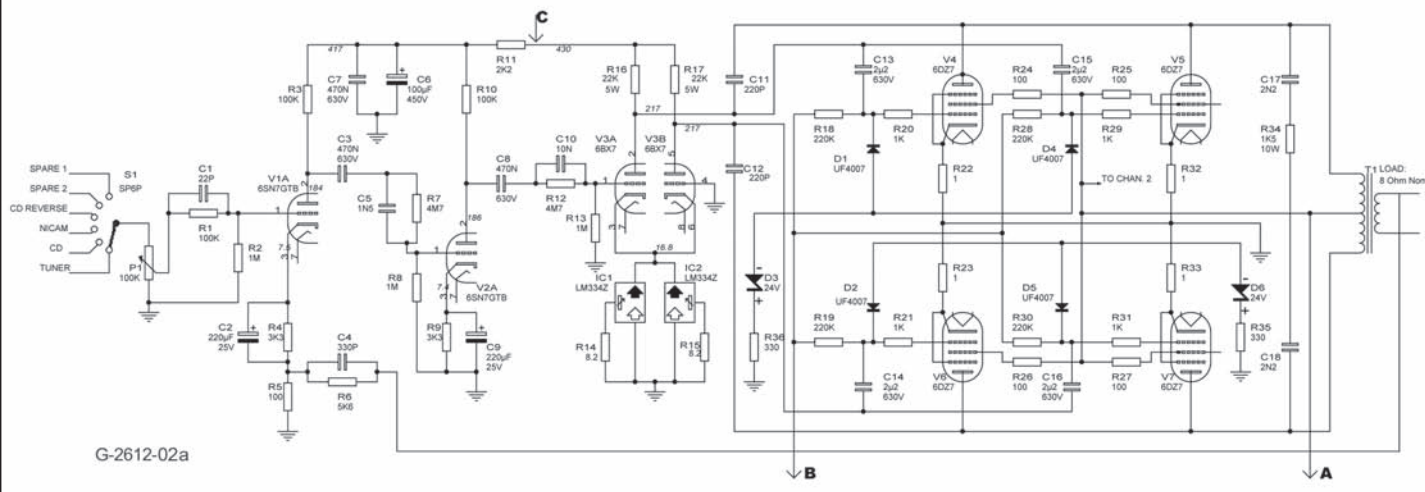
Photo 3 shows clean soft clipping at over 60W RMS into a 5 Ω load on a nom-

inally 8 Ω output. Previously only 40W had been available before developing a nasty zero-crossing kink, which sounds terrible on music because it happens thousands of times per second, in contrast to transient peaks being clipped infrequently. **Photo 4** shows one 6DZ7 cathode current smoothly clipping at over 400mA (!) from a quiescent current of 50mA. **Photo 5** shows the actual grid

signal reaching overdrive to +4V, and clamping gently below -24V, as hoped, with the fixed-bias supply being -12V.

Switching off the 700Hz overdrive revealed that the average grid potential had dynamically stabilized at about -10V during that overdrive, which means nearly pure class A (-9V) during dynamic conditions! What a dream scenario for maximum sound quality. And plate power dissipation should

FIGURE 2a: Complete clamped-bias amp circuit.



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Frequency Response: 7K-100KHz (-18dB/oct crossover inside)
Crossover Fc: 18KHz/96dB/2.83V/1M (300W)
13KHz/92dB/2.83V/1M (300W)
Impedance: 8 ohm / 12 ohm Ribbon Size: 15 x 25 x 0.006mm



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not be a problem, because dissipation equals power input minus audio power output.

With a lesser zener, say -18V, you could actually drift from class AB1 to genuine A2, rather than AB2! I have not tried that, but I have tried upping the fixed bias to -19V, giving quiescent current of only 2mA, implying class B. The high-drive waveforms were still all virtually the same as above, confirming that class B low-drive was drifting to class AB2 in overdrive.

Very interesting, but this, of course, gave a rather thin sound quality at quiet levels. Historically, the problem has been the opposite: quiescent class A drifting to class B due to grid current charging the coupling capacitors, and giving a mushy gradual overload as your amp gets its zero-crossing current strangled off. Actually, cathode bias alleviates this problem by drifting down when the current falls, which is why many experimenters over the years have reported that cathode bias sounds better on music in spite of giving lower sinewave-test power.

WRAPPING UP

Well, there you have it. Anyone is welcome to use my dynamic-bias-clamp, which might soon sweep the tube amp board, pretty much as ultralinear did around 1954. If it's taken half a century for the "next big tube thing," better late than never. And, it is easily retro-fittable to any existing RC-coupled push-pull tube amp in just a few minutes.

PHOTO 3: Clean heavy-overdrive output clipping beyond 60W, without the usual "shoulders" that appear in RC-coupled amps due to grid bias shift, and sounding like severe crossover distortion rather than subtle clipping.

PHOTO 4: The overdrive grid drive voltage, 5V/div, with 0V set at 2cm above center.

PHOTO 5: The quiescent grid voltage sitting at about -12V.

PHOTO 6: One 6DZ7 (twin EL84) cathode current achieving over 400mA peak on overdrive. There are eight 6DZ7s, which is like sixteen EL84s.

PHOTO 7: One 6DZ7 quiescent cathode current at around 40 to 50mA.

For full benefit, a slightly lower load impedance would actually help, because then you could swing more peak current.

To those concerned about acceler-

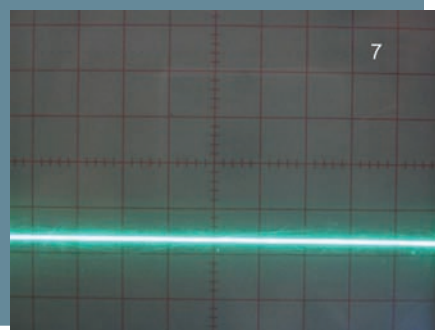
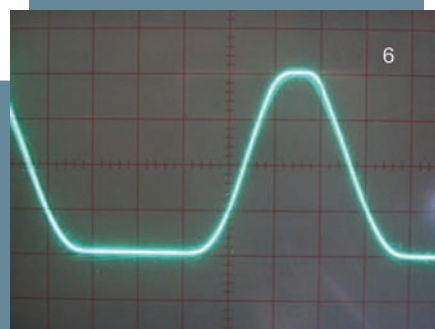
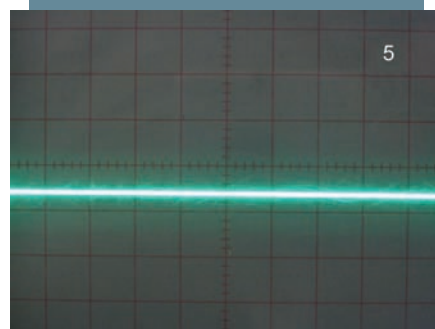
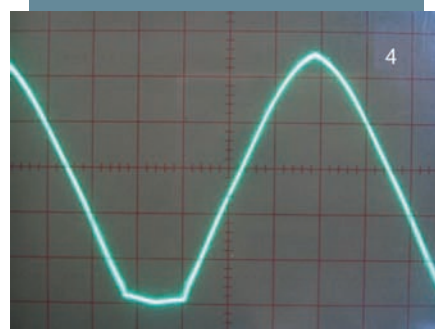
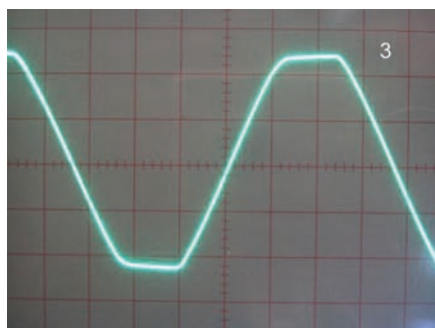
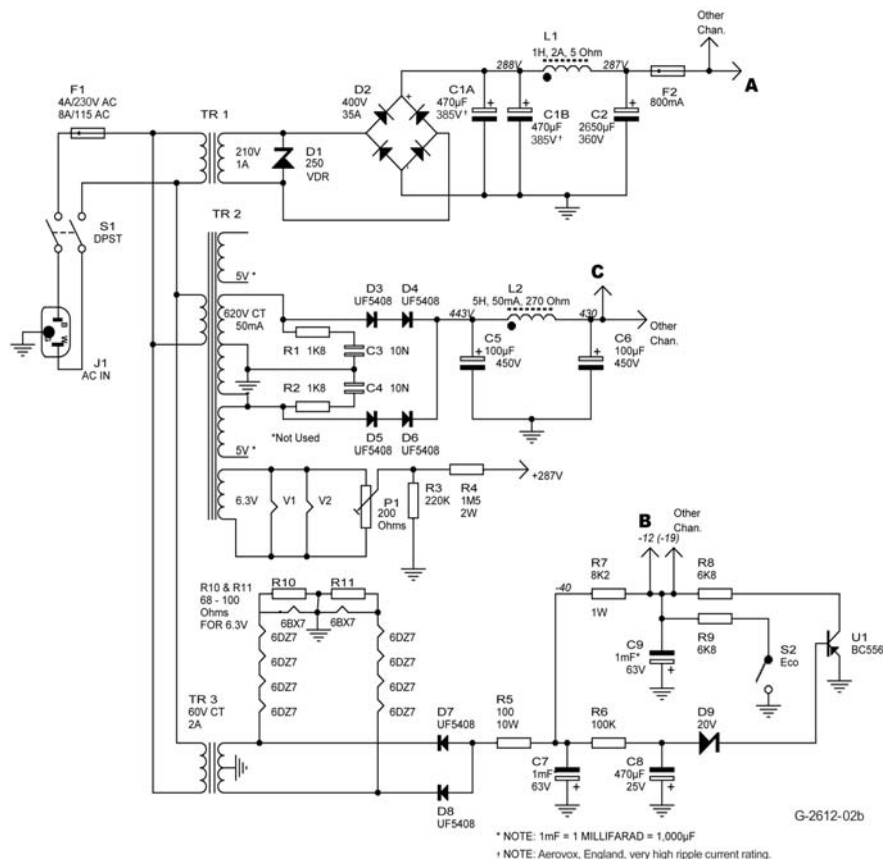


FIGURE 2b: Complete clamped-bias amp circuit.



ated tube-aging, I had, during my feed-back stabilization adjustments (because 26dB is quite a lot of feedback), some high power oscillations at 150kHz, and later at 2Hz, causing overdrive so huge that I could hear one 18 lb output tranny ticking audibly at 2Hz. But everything survived, and the tubes were apparently unaffected, meaning same quiescent current, and still capable of 400mA peak.

The power response is -1dB at 7Hz relative to 60W. The output impedance is 0.2Ω. I designed the 18 lb output transformers myself, 14-section coil, using the *Radio Designers Handbook*, and a retired army-tech wound them for me. I hand-painted the end-covers with Hammerite paint from England to achieve the unique kaleidoscopic color scheme.

Finally, I won't be making the usual constructor's fatuous comment on sound quality, such as "it sounds like Westinghouse 300Bs on steroids, and so on," because we all know that every mother's own baby is the most wonderful baby in the world anyway. *ax*

AMP CIRCUIT PARTS LIST

C1	22P
C2, C9	220μF
C3, C7, C8	470N
C4	330P
C5	1N5
C6	100μF
C10	10N
C11, C12	220P
C13-16	2μ2
C17, C18	2N2
D1, D2, D4, D5	UF4007
D6, D3	2V
IC1, IC2	LM334Z
R1, P1, R3, R10	100k
R2, R8, R13	1M
R4, R9	3k3
R5, R24-7	100
R6	5k6
R7, R12	4M7
R11	2k2
R14, R15	8.2
R16, R17	22k
R18, R19, R28, R30	220k
R20, R21, R29, R31	1k
R22, R23, R32, R33	1Ω
R34	1k5
R35, R36	330
S1	SP6T
T1	Output
(1750Ω2 p-p, 40H, 5mH, 18 lbs)	
V1, V2	6SN7GTB
V3	6BX7
V4-7	6DZ7

POWER SUPPLY PARTS LIST

C1	22P
C2, C9	220μF
C3, C7, C8	470N
C4	330P
C5	1N5
C6	100μF
C10	10N
C11, C12	220P
C13-16	2μ2
C17, C18	2N2
D1, D2, D4, D5	UF4007
D6, D3	24V
IC1, IC2	LM334Z
R1, P1, R3, R10	100k
R2, R8, R13	1M
R4, R9	3k3
R5, R24-7	100
R6	5k6
R7, R12	4M7
R11	2k2
R14, R15	8.2
R16, R17	22k
R18, R19, R28, R30	220k
R20, R21, R29, R31	1k
R22, R23, R32, R33	1Ω
R34	1k5
R35, R36	330
S1	SP6T
T1	Output
V1, V2	6SN7GTB
V3	6BX7
V4-7	6DZ7

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DAVID: The Giant Killer

This author's latest horn box slays the competition with big-time performance in a small package.

By Bill Fitzmaurice

The David loudspeaker (**Photo 1**) is a high-efficiency design that offers a level of performance you once could only obtain with cabinets two times or more its size. The heart of this box is a horn-loaded subwoofer (**Figs. 1 and 2**). Utilizing a long-throw high Bl 8" driver, it achieves low-frequency extension and sensitivity once reserved for large horn-loaded cabinets equipped with fifteens.

As you can see in **Fig. 3**, the response of the subwoofer section almost equals that of the legendary Klipschorn, but from a far smaller and much easier to build package. Note the differences in the corner, wall, and freestanding

placements in the bass; you can adjust the bass response quite a bit just by moving the cabinet.

Trace D shows the response at 30° off-axis. Most experts agree that superior imaging and realism requires no more than a 6dB differential between the on-axis and 30° off-axis response. David's 30° off-axis response averages within 3dB of on-axis. Average sensitivity of 94dB/2.83V/1m allows powering with low wattage tube amplification.

DRIVER SELECTION

You can configure the midbass/tweeter section of the David in a variety of ways. The prototype uses an MTM, but it could also use a single high-efficiency wide range Fostex, Lowther, or Jordan driver, because the woofer horn can keep up with even the most sensitive full-range drivers. You may use high-end audiophile drivers, ribbon tweeters, or even the same inexpensive drivers used in the original—the Pioneer A11EC80-02F midbass and the MCM

PHOTO 1: The completed speaker.



FIGURE 1: 3-D view of cabinet insides from the back.

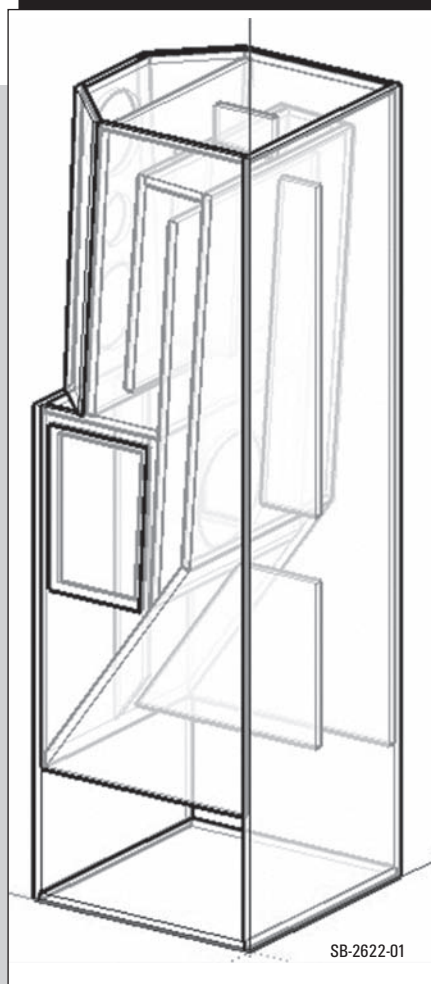
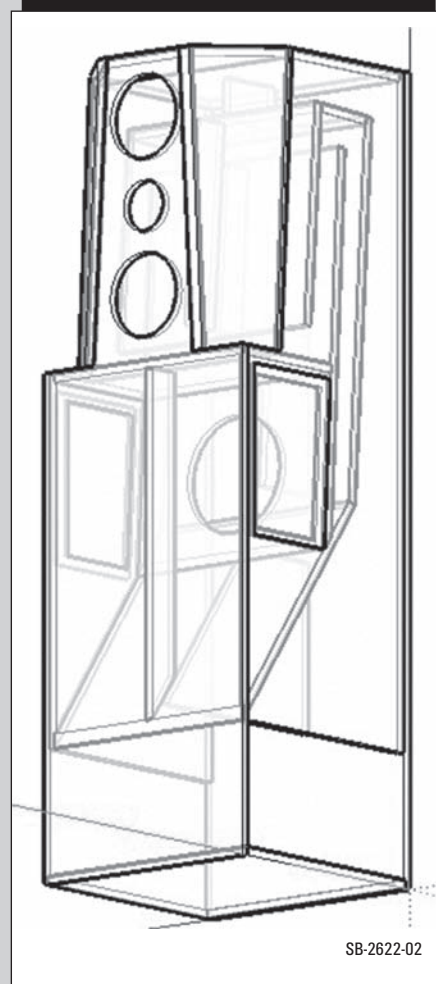


FIGURE 2: Front 3-D view of cabinet.



BK-16 Kit

Madisound is pleased to offer the **BK-16 folded horn kit**.



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We have chosen the **Fostex FF165K 6.5"** full range for use in the BK-16 cabinet. The **FF165K** has a **Kenaf fiber cone**, inverted foam surround and aluminum dust cap.

The **FF165K** is run full range with a frequency response out to 15kHz. The **T90A** super tweeter has been added to cover the upper frequencies. The **T90A** is a top-mount horn tweeter with an **Alnico magnet**. The tweeter is rolled off on the low end with a **Fostex Tin & Copper foil capacitor**. The system frequency response is 55Hz to 35kHz at 95dB.



The **BK-16** cabinet is made from **Baltic Birch** plywood and is sold flat, unassembled, unsanded and unfinished. Cabinet dimensions are 9.75" W x 14.75" D x 29" T.

The kit includes:

- Pair of Cabinets - Flat
- Pair of FF165K - full range
- Pair of T90A - horn tweeter
- Pair of DB-Cup - Input cup
- Pair of Crossovers
- Nordost 2-Flat wire for tweeter
- Instructions

Kit Price: \$635.00 /pair

Parts without cabinets: \$439.00 /pair

Cabinets only: \$98.00 each

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(www.mcmminone.com) 53-325 dome tweeter. The preferred subwoofer drivers are the MCM 55-2421 and the Tang Band W8-740. An alternative is the Dayton Quatro 8 (www.partsexpress.com), which lacks the output capability of the MCM or TB, but is still adequate.

You may run the speaker full range, with a sub amp, or even in tri-amplification. The original's average impedance is 6Ω, with the woofer at 8Ω (the driver is rated at 4Ω, but the high acoustic impedance horn doubles that figure), the mids at 4Ω, and the tweeter at 8Ω (**Fig. 4**). Being as sensitive as it is, this is not a difficult speaker to drive.

CONSTRUCTION

I built the prototype from ½" Baltic birch plywood. You may also use MDF, especially if you plan on a painted or veneered finish. I used all dado joints, cutting grooves into the cabinet sides with a router, but plain butt joints are

OK. With dado construction you can hold the parts together with small nails or trim staples while the adhesive sets, but for butt joints drywall screws—piloted and countersunk—would be better. As for the adhesive, I recommend a caulking gun applied polyurethane base construction adhesive, which expands while curing to fill gaps and ensure airtight joints.

Note that you must determine the final sizes of all parts by dead reckoning, because the actual thickness of the materials used will determine part lengths. Start by rough-cutting your materials. The front and back measure 15" wide x 40" (two pieces). Note that the rest of the parts measure 14" wide, so cut accordingly. With 5 x 5' Baltic birch you'll need two sheets, and you'll have some left over. If you use dado construction, make the interior parts ¼" wider to account for the material that will go into the dados.

FIGURE 3: Subwoofer response.

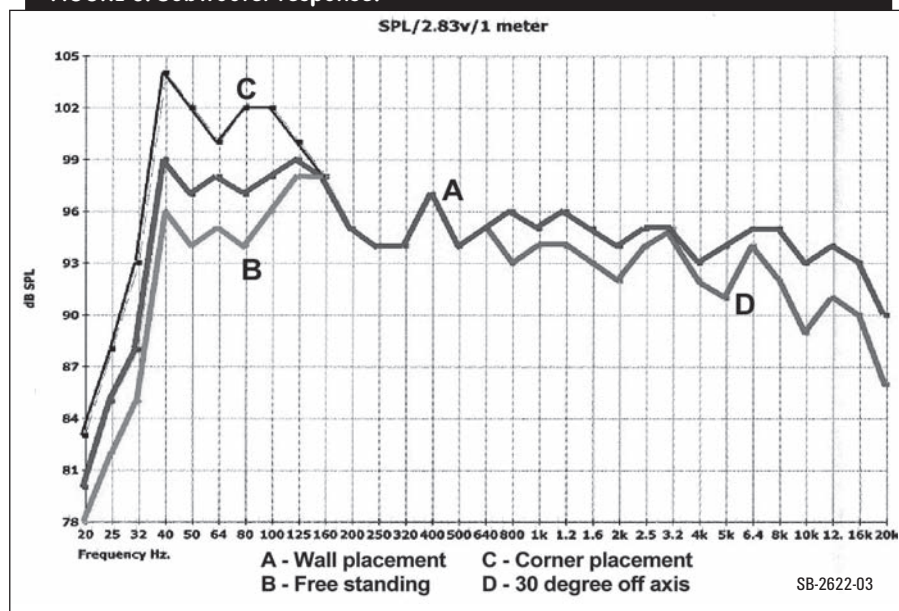
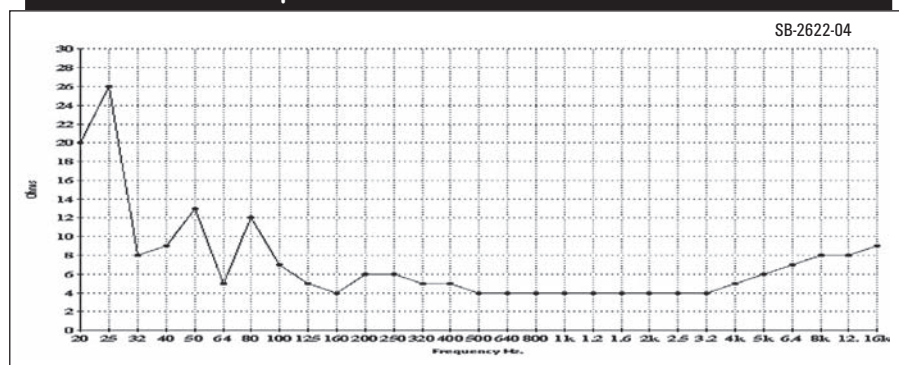


FIGURE 4: Tweeter response.



The next step is to draw the locations of the joints on the cabinet sides per **Fig. 5**. Draw them all on the inside, and also on the outside where plates 1, 2, and 6 form the woofer chamber, keeping in mind that the front and back overlap the sides. The internal clearances are plate-to-plate. Measure them starting at the horn mouth, going backwards through the horn, so that whatever size discrepancy caused by your material thickness will end up altering the volume of the midbass/tweeter chamber, which is not critical.

With dado joints rout a 1/8" deep trough at all the joints with a 1/2" pattern-following router bit. Clamp or screw the sides to the workbench, clamp or screw a straight guideboard on the joint, and rout away. Make sure to fill any holes you make later on with adhesive. When you finish the first side do the other, in mirror image.

FIGURE 5: Internal dimensions.

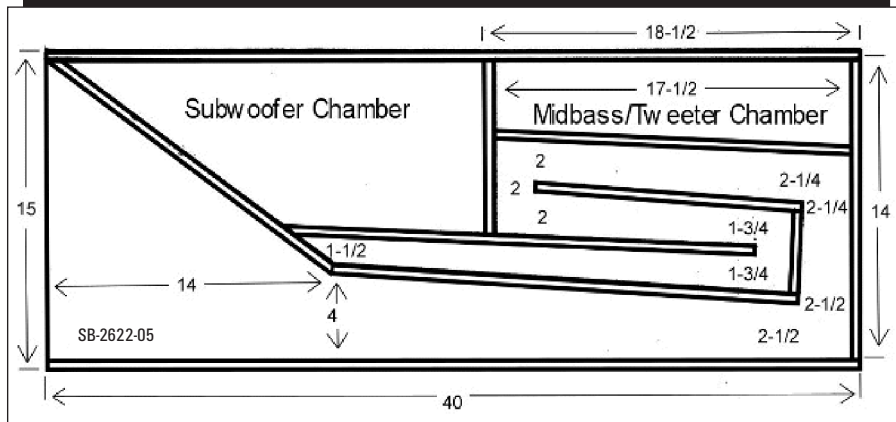


Diagram illustrating the location of the subwoofer chamber and side removal area. The diagram shows the rear of the vehicle with the subwoofer chamber and side removal area labeled. The subwoofer chamber is located in the center, and the side removal area is located on the right side. The diagram is labeled with numbers 1 through 8, indicating the location of the subwoofer chamber and side removal area.

Subwoofer Chamber

Side Removal Area

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FIGURE 7: Brace locations.

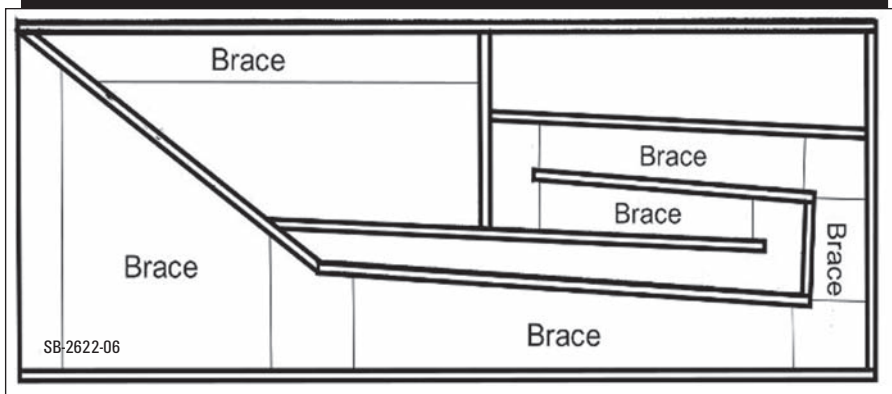


FIGURE 8: Midbass baffle wings.

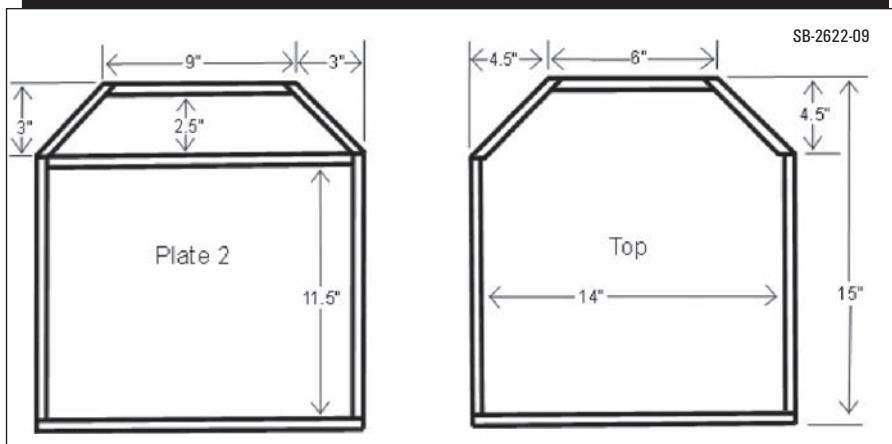


PHOTO 2: Clamping inside panels.



PHOTO 3: Installing the brace for plate 6.



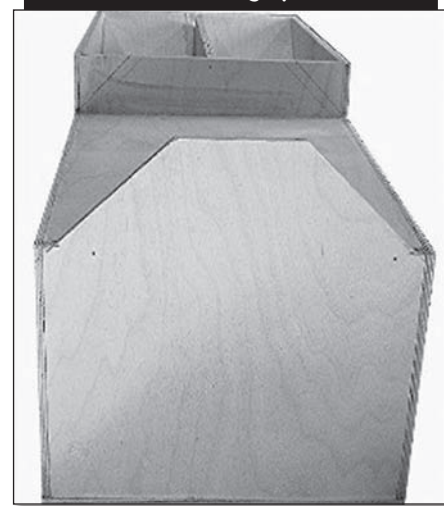
as you install each plate. Either butt- or dado-joint them as you prefer. For dado jointing, again use a $\frac{1}{8}$ " groove $\frac{1}{8}$ " deep; the easiest way to cut these is with the router mounted in a router table with a fence. Determine each plate length, cut it to size, and dado for the brace if you're using dado joints. Note that the angles of the plate ends are small enough that you can cut them square and the expanding urethane adhesive will fill the gaps, except for plate 1 as already noted, and the mating ends of plates 5 and 6, which are cut at 15° .

Install plate 1, then measure, cut, and install plate 2, using a square to align the parts and a clamp to hold them square until fastened (**Photo 2**). Add the other side to the assembly, and attach the brace that runs between plates 1 and 3. Next attach the brace plate 3 that runs between it and plate 7. Then install plates 4 and 5, remembering to first attach their braces, and then plate 6.

Don't cut the end of plate 6 where it meets the cabinet front at an angle, instead cut it long enough to overhang the joint. Attach the brace to plate 6. Apply plenty of adhesive to the plates 5 and 6 joint to fill the gap caused by the angle of the joint. From the inside attach the brace that runs from plate 2 to plate 6. Trim off the extra from plate 6, clamping a board across the cabinet bottom to serve as a platform for a jigsaw.

Use a square to install the brace from plate 2 to plate 6 (**Photo 3**), and use a sander to make it flush. Attach

PHOTO 4: Buttoning up the box.



assembly. Start installing the internal plates per the assembly order in **Fig. 6**. Cut out plate 1, marking on it the joint with plate 2. Cut the end of plate 1 where it meets plate 6 at a 45° angle; it's not perfect, but the adhesive will fill the gap. Cut a 7" hole in plate 1 for the woofer; if you plan on mounting the driver with T-nuts, install them now and trial-fit the driver.

Braces are located midway across the plates (**Fig. 7**); determine their sizes by dead reckoning

plate 7. After the adhesive has set, sand the excess side material for a nice smooth joint. Cut out the top and install

according to **Fig. 7** and **Photo 4**.

Drill a hole through plate 2 for two speaker wires (**Photo 5**), leaving a cou-

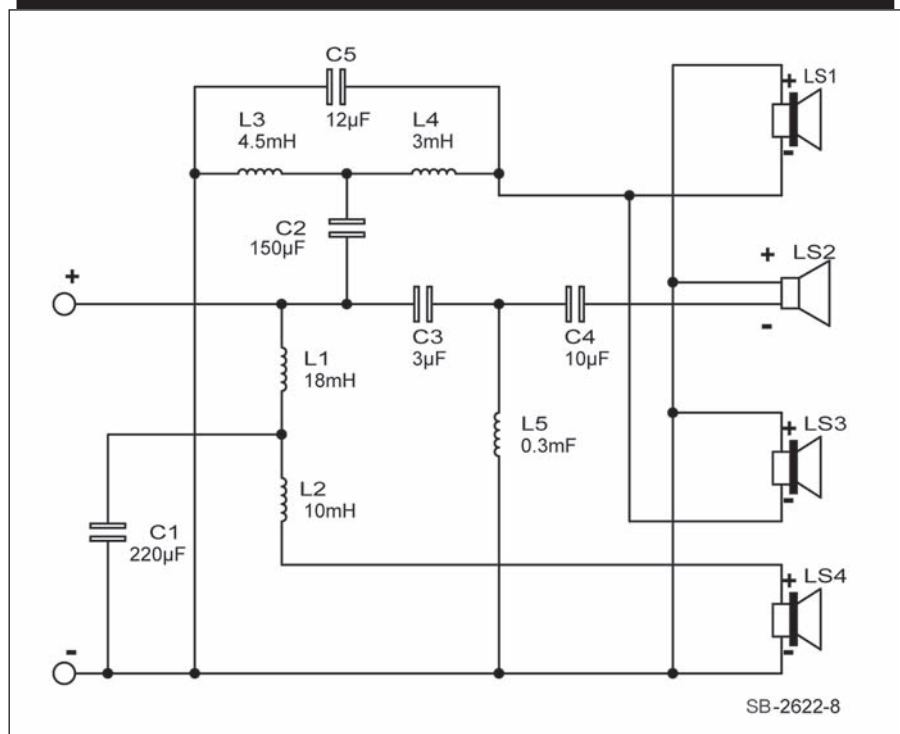
ple of feet hanging on either side, and seal the hole tight with adhesive or hot melt glue. Also note the lines drawn on plate 2. These are where the joints will be with the midbass baffle wings; **Fig. 8** provides more detail. Both here and with the top, don't worry if things aren't quite perfect according to the diagrams, as variations in the thickness of your materials will influence the finished measurements, but not enough to affect performance.

FRONT AND BACK

If you're using dado joints, dado the inside of the front and back panels where they will joint. These panels are 48" long, leaving an 8" overhang at the bottom of the cabinet. Where the front will intersect, the baffle wings cut it at a 45° angle (**Photo 6**).

Rough-cut the baffle wings larger than necessary by about ¼" in each dimension, with the edges that will mate at the cabinet sides cut at a 45° angle. Place these where they will join the cabinet, the 45° edge meeting the side, and trace where the other joints

FIGURE 9: Crossover schematic.



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PHOTO 5: Accommodating speaker wires.

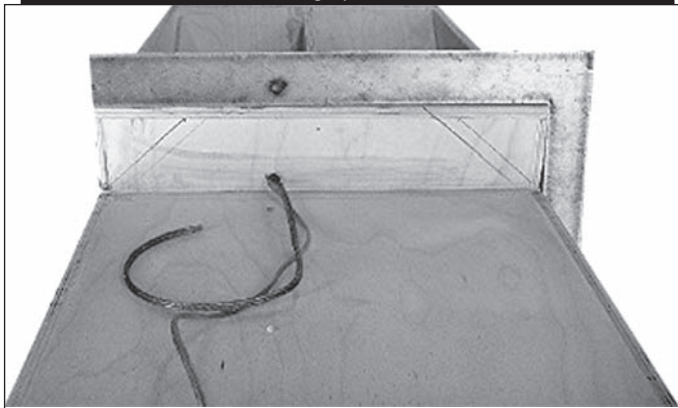


PHOTO 6: Front panel and baffle wings.



will go. Trim it to size but leave 1/8" or so extra; install it and do the same with the other wing (**Photo 7**). When the adhesive has set, sand the joints flush, as well as all the cabinet edges. Depending on how you plan to finish the box, you may

choose to make the edges rounded over or crisp and sharp.

Using the lines previously drawn on the outside of the box, rough-cut a port-hole about 1" from where plates 2, 6, and the front intersect the side, about 10" long (**Photo 8**). Do the same on the other side. With a straight guideboard clamped or screwed to the cabinet, use a router with a trim bit to finish the holes as in **Photo 9**.

Reaching through the opposite port-hole, trace onto plywood the port-hole openings to produce covers for them. Rim the openings with plywood to produce mounting flanges for the covers (**Photo 10**). Be very sure that the sub driver will fit into the cabinet. Then cut and trim the port-hole covers, making sure they fit properly.

Cut the holes for the midbass drivers and the tweeter with a jig-saw or hole-saw. Keep the drivers pretty tight to each other, with the upper midbass driver mounted close to the top of the cabinet. If you prefer to use a router to

cut insets in the baffle for flush mounting the drivers, don't go more than 1/4" deep. Install the bottom. Note how the final mouth of the horn is formed by having the front and back longer than the sides, and the mouth exits to both the left and right.

PHOTO 7: Installing the baffle wings.

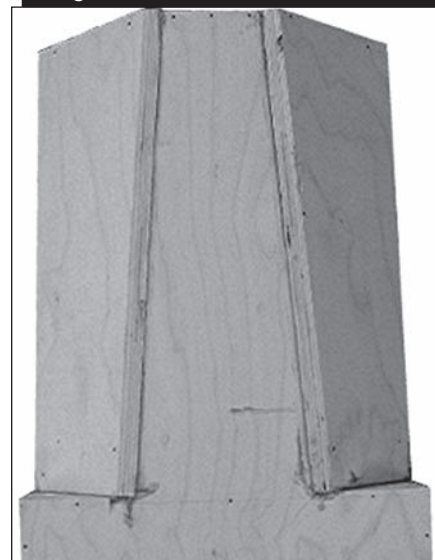


PHOTO 8: Rough cut for the port-hole opening.

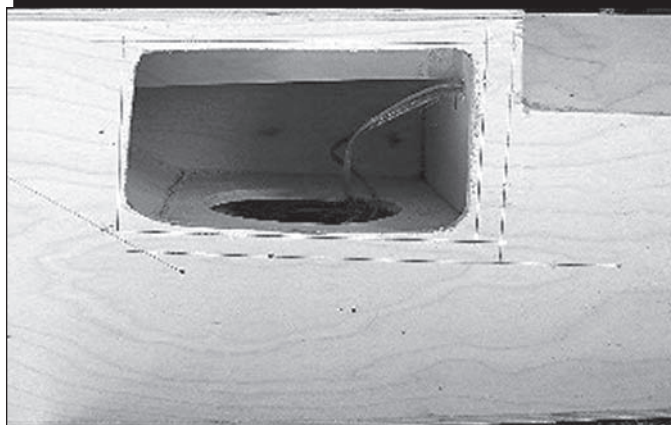


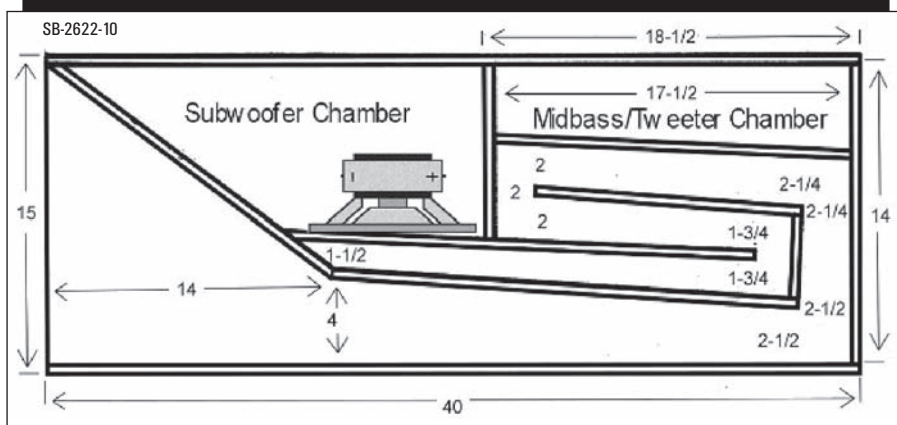
PHOTO 9: Finishing the porthole.



PHOTO 10: Flanges for the cover.



FIGURE 10: Woofer shown attached.



FINISHING TOUCHES

Now is the time to put a finish on the box—whatever you wish, from bare wood to paint to a fine veneer. When that's complete, mount the drivers, connecting the wires to them when you do so, wiring the two midbasses in parallel. Loosely but thoroughly stuff the midbass chamber with polyester pillow stuffing (stuffing is not used in

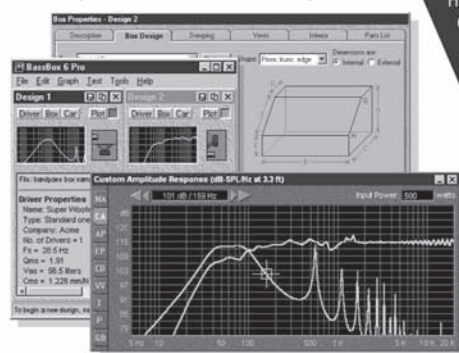
the woofer chamber). Weather-strip one of the porthole flanges and screw the cover in place.

On the cabinet, lower-back mount binding posts—one set if you're going full range, two sets if you're using a separate sub amp, three sets for bi-amping. Drill a hole through plate 6 for the wires to pass through to the woofer chamber. Wire the posts, feed the wire into the woofer chamber, and seal the hole airtight. For a separate sub amp, wire one set of posts directly to the woofer; for tri-amping, wire the three sets of posts to their respective drivers.

The best place to mount the crossover is on the remaining porthole cover. When you wire the binding posts, make sure to have plenty of wire so you can wire to the crossover with the cover off the box. **Figure 9** is a schematic of the crossover for full range use with the suggested drivers, with crossover points at approximately 200Hz and 4kHz. The values of the 3rd-order LP filter on the woofer appear to be those for a far lower crossover point, but that compensates for the rising response of

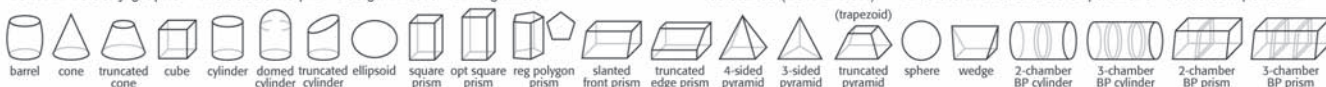
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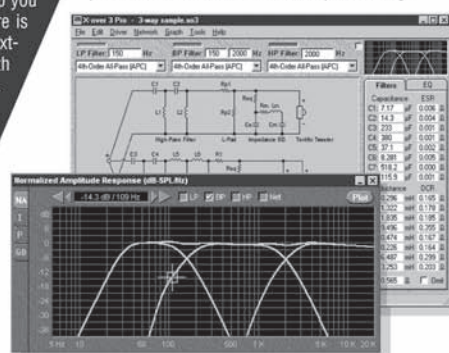
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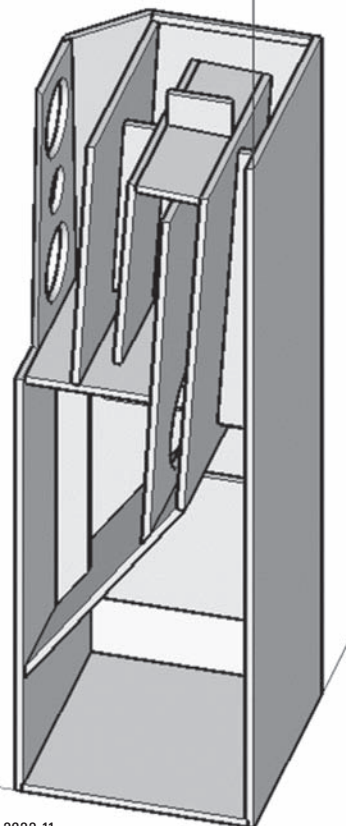
- 1st, 2nd, 3rd and 4th-order "ladder" filter topologies. • Parallel crossover topology. • 2-way crossovers offer Bessel, Butterworth, Chebyshev, Gaussian, Legendre, Linear-Phase and Linkwitz-Riley filters. • 3-way crossovers offer All-Pass Crossover (APC) and Constant-Power Crossover (CPC) filters. X-over Pro's capabilities scale to fit the amount of driver data. • A crossover, filter or L-pad can be designed with just the nominal impedance of each driver. • With full Thiele-Small parameters, impedance equalization can be designed and the performance graphed.
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the woofer horn. The 18mH, 10mH, and 3mH inductors must be iron or ferrite core with low (less than 1Ω) Re values; air core is fine for the 4.5mH and 0.3mH inductors.

The 220μF and 150μF caps are electrolytics, though the 150μF value will work better if you parallel-wire a 140μF electrolytic cap and a 10μF poly cap. All remaining caps must be polys. Securely attach all the crossover components to the porthole cover and wire it up. Note that the midbasses and tweeter are wired with inverse polarity relative to the woofer. With a separate sub amp, use only the components for the midbass/tweeter crossover, also eliminating the 150μF cap and 4.5mH coil midbass high-pass filter.

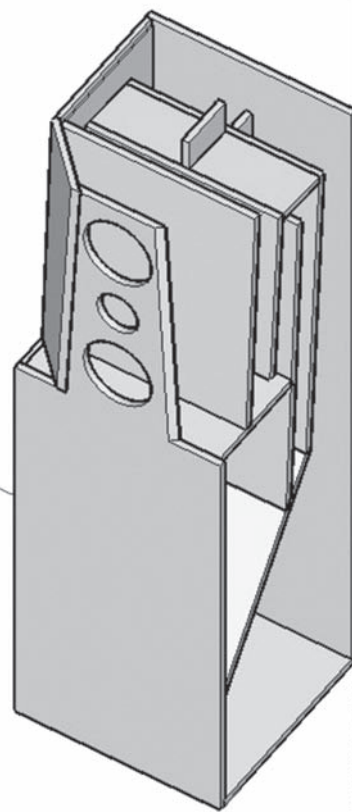
With tri-amping, a crossover is not required, but still use a 10μF DC protection cap in series with the tweeter. When everything is wired, weather-strip the mounting flange and screw the cover in place. Finally, if you wish, add either feet or isolation spikes to the bottom. *aX*

FIGURE 11: Cutaway from side.



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FIGURE 12: Cutaway from top.



SB-2622-12

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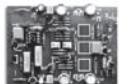
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An IM Signal Generator Upgrade

By Charles Hansen

Is it time to update your signal generator?

Follow these tips from an audio expert.

Use an intermodulation distortion (IMD) signal generator based on the FET-based design by Erno Borbely¹. It produces a CCIF 19 + 20kHz and a multi-tone (MIM) 9 + 10.05 + 20kHz test signal. It has served me well for the last eight years. The only downside is the noise floor around -90dB. Progress in audio equipment design has made this noise floor the limiting factor in some of my measurements. I also have an occasional need for the CCIF 11 + 12kHz test signal, which my old IMD generator does not produce.

The other problem is its size—a 19" × 12" × 4.5" rack unit chassis. This is because it also contains a steep 80dB 1kHz bandpass filter to isolate the 1kHz IMD product and direct it to an external meter. Ever since I purchased a 16-bit audio spectrum analyzer, I no

longer use the 1kHz filter portion of the IMD generator, and with all the other test equipment at my bench, I also wanted a smaller IMD unit.

A lamp-stabilized Wien-bridge sine-wave oscillator design² has a minimum number of components, and can achieve a low distortion level of 0.0025%³. One of the weaknesses of this circuit is the higher distortion at low frequencies due to the response lag in the lamp. This would not be a problem in my design because the lowest frequency I planned

on developing is 9kHz.

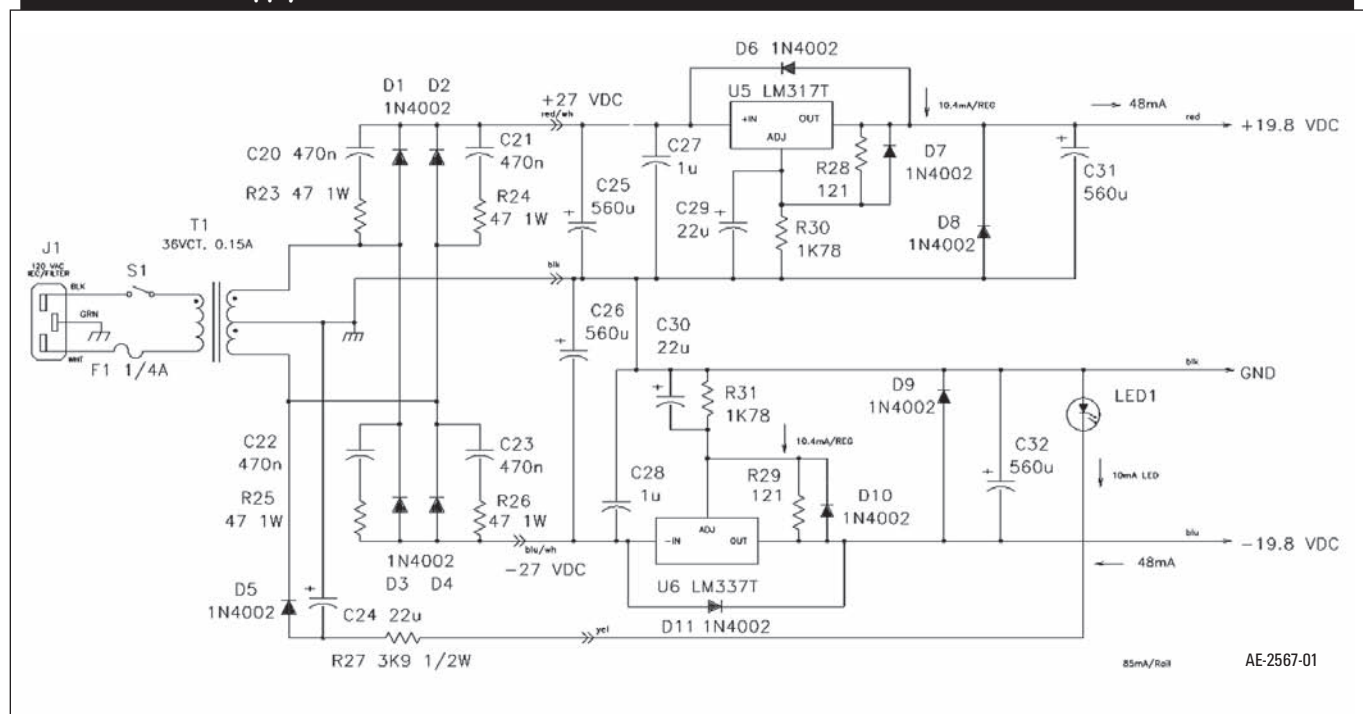
Parts cost, including the chassis, is less than \$90.

POWER SUPPLY

The power supply for the IMD generator circuitry is shown in Fig. 1. The raw supply of ±27V DC is obtained by full-wave rectifying the output of center-



FIGURE 1: Power-supply schematic.

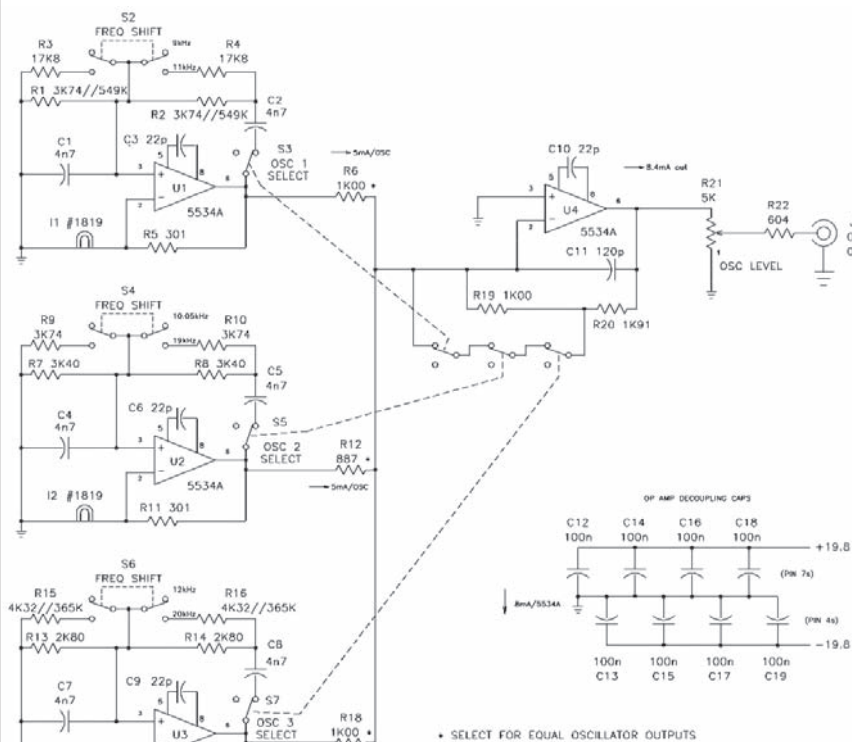


tapped transformer T1 and rectifying it with diodes D1-D4. R-C snubber networks are connected across each diode to attenuate transients due to rectifier switching. LED1 serves as the power-on indicator.

The resulting DC voltages are filtered by C25 and C26, and then applied to linear regulator ICs U5 and U6 to produce power-supply voltages of $\pm 19.8\text{V DC}$. I wanted as high an output voltage as possible to drive low-gain amps and preamps. The regulated supply voltage is 90% of the maximum 22Vcc rating of the oscillator op amps, and is my upper limit to ensure long-term reliability. C29 and C30 reduce the output noise from the regulators. Capacitors C31 and C32 are the regulated DC output filter capacitors. Diodes D6-D11 protect the linear regulators in case of accidental faults.

The schematic diagram of the IMD generator is shown in Fig. 2. The unit is composed of three low-distortion sine-wave oscillator circuits (U1-U3) and summing amplifier U4. Each oscillator provides one of two different output

FIGURE 2: IMD generator schematic.



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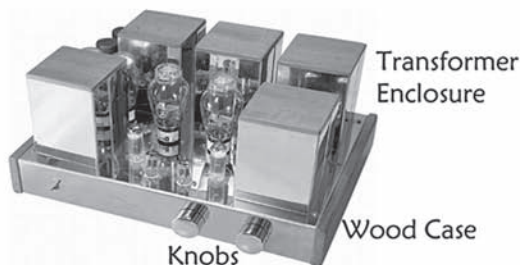
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frequencies, and can be disabled when not in use.

Because all three oscillators are identical except for the resistors that determine the frequency of oscillation, I will explain the oscillator at U1. S2 selects between two pairs of matched 1% metal film resistors (R1-R4) that select the tuned frequency. The parallel RC Wien network is provided by the resistors selected by S2 (R1 or R1 in parallel with R3) and capacitor C1. The series RC Wien network is provided by resistor R2, or R2 in parallel with R4, and capacitor C2. These networks provide the positive feedback across U1 and the phase shift necessary for oscillation.

The gain of U1 is set by R5 and the resistance of lamp I1. The negative feedback control for gain stability is provided by the positive temperature coefficient of lamp I1. When power is first applied, the lamp is at its cold resistance of about 87Ω. The gain is about 4.5, so the oscillator voltage builds up.

As the output voltage increases, the current through I1 increases and so does its resistance. This reduces the gain of U1. With the proper selection of R5, the oscillator establishes a stable operating point, and a low-distortion sine-wave output results. Note that I used a #1819 lamp instead of the more commonly used #327. This is based on my earlier experiments with lamp-

based Wien-bridge oscillators⁴.

Each oscillator drives the output inverting summing amplifier U4 through a series select-at-test (SAT) resistor. The exact SAT resistor value is based on the stable oscillator output voltage, which depends on the individual lamp. It is necessary to adjust the gain of U4 to just below its clipping output voltage whether only two oscillators are in use for the CCIF test tones, or all three are in use for the multi-tone output.

Take the case in which the 19 + 20kHz CCIF signal is used. S4 selects the 19kHz tone from U2 and S6 selects the 20kHz tone from U3. S3 is opened and turns off the third (U1) oscillator by disconnecting C2 from the output of U1, thus opening the positive feedback path. This is necessary to prevent crosstalk that would occur if the unused oscillator were left running. S3 also removes the short across R19, so the overall voltage gain of U4 is set to -2.91. This produces an output voltage at U4 pin 6 of 12V RMS.

When all three oscillators are operating for the multi-tone IMD signal, switches S3, S5, and S7 short R19, decreasing the gain to -1.91 to reflect the additional summing voltage component from U1. This reduced gain maintains the output voltage at U4 pin 6 to 12V RMS.

Operating just one of the oscillators

at a time produces a single sine-wave output, but at a reduced output voltage of just over 6V RMS since the gain of U4 remains -2.91. This is also the mode used to check the THD, amplitude, and frequency of each individual oscillator.

The IMD output signal is applied across pot R21, and R22 sets the output impedance to 600Ω. C11 limits the -3dB bandwidth of U4 to 455kHz. The 22pF caps at each op amp are necessary since the 5534A is not stable for gains less than 5 without these compensation caps. Caps C12 through C19 are local decoupling capacitors located at the IC package supply pins.

The circuitry is quite versatile. You can add additional oscillators for other IM measurements such as SMPTE, or even a square-wave oscillator section for DIM testing. The switching will become more complicated because the gain of U4 will need to be adjusted for each type of IM signal you want to generate, but 3PDT miniature toggle switches are readily available.

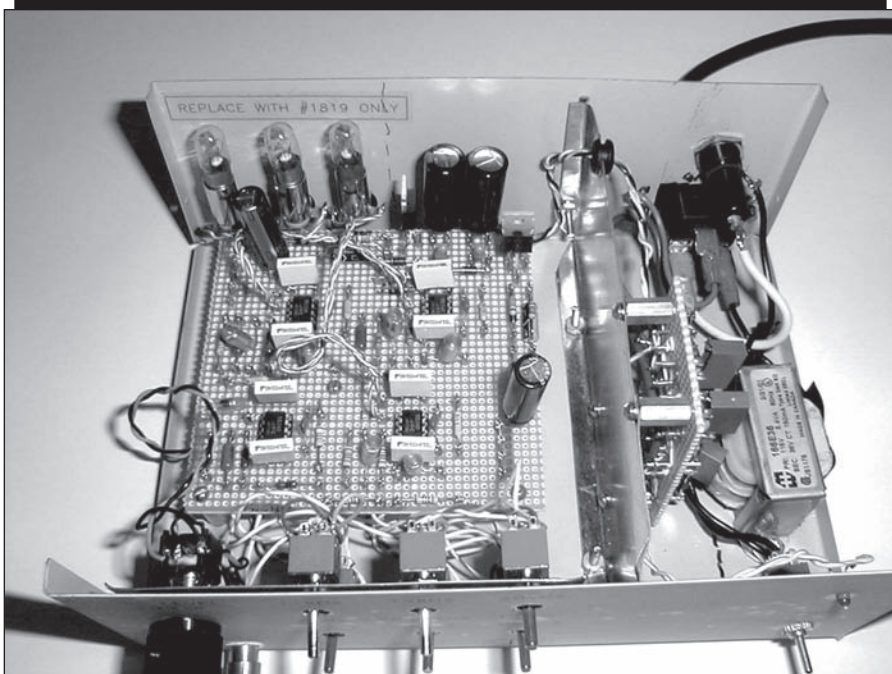
CONSTRUCTION

The circuitry makes use of high-quality components to ensure that THD+N and stability are the best possible. The frequency accuracy depends on how well the oscillator RC network components are matched. Matched pairs of polystyrene caps were selected for each individual oscillator from a large sample of 5% tolerance parts. Similarly, matched pairs of metal film resistors were selected for each individual oscillator from a large sample of 1% parts. Note that R1, R2, R15, and R16 are actually parallel pairs of resistors. These were needed to achieve the exact output frequency to within 0.2%.

Photo 1 shows the front view of the IMD signal generator. The three upper toggle switches select the oscillator frequencies. The lower switches turn the oscillators on or off. The output level control is to the left above the phono output jack.

Photo 2 shows a top view of the perfboard that allows all the components to fit within the 8.25 × 6.125 × 3.06" chassis specified in the parts list. The transformer and rectifier/snubbers are on the right side of the unit with a galvanized steel shield between the power

PHOTO 2: Interior of unit.

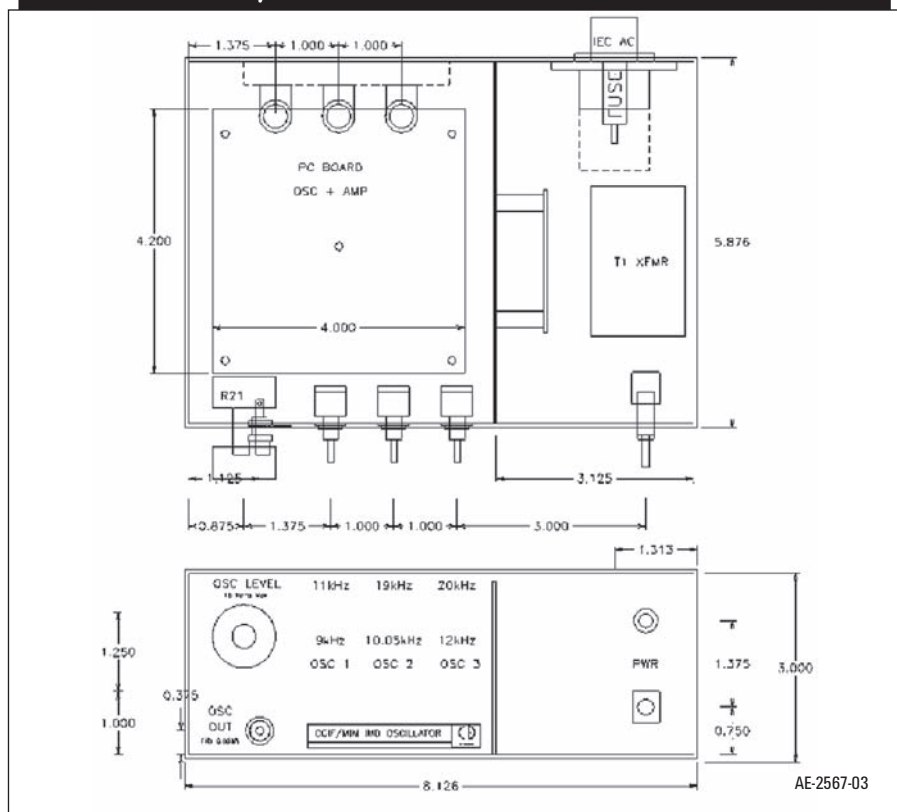


section and the oscillator/amplifier section. I used an IEC line cord receptacle for the AC power. Scrape the paint off the chassis where the green IEC ground wire connection is made. The three lamps are mounted on the rear wall of the chassis with rubber grommets to isolate them from vibration. **Figure 3** shows the mechanical layout of the chassis.

The power-supply regulators are on the perfboard. I used star ground and DC supply connections to minimize oscillator crosstalk. The oscillator and amplifier circuits float with respect to the chassis to avoid any ground loops between the IMD generator and the equipment under test.

Figure 4 shows the front panel designations for the switches and controls. Make an exact size copy for use in locating the holes for the components. You can use transfer lettering, or photocopy the figure to scale on a sheet of matte self-adhesive drafting appliqué film as I did. This film is available at drafting and art-supply houses under the brand name Letraset Letracopy Creative or

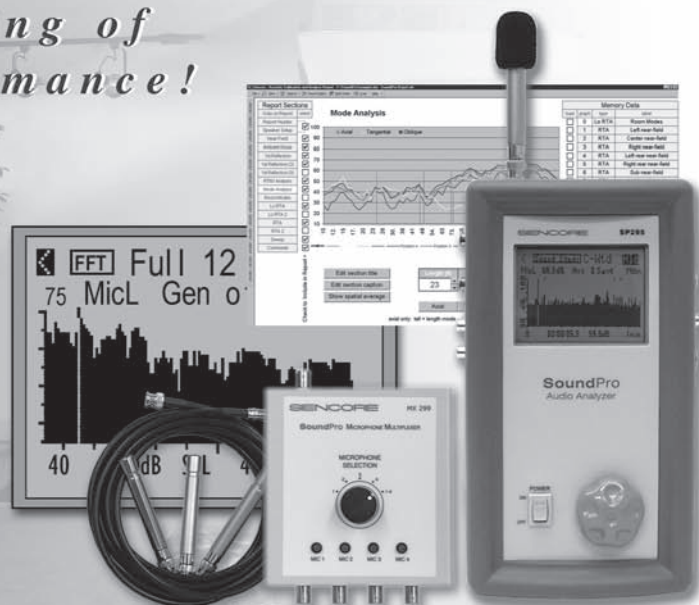
FIGURE 3: Chassis layout.



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Chartpak DAF8 appliqué film (see TAA 2/94, p. 29 for details).

A light-color paint (which the speci-

fied chassis has) is required since the photocopy lettering is black. Apply the lettered film to the front of the unit in

the proper location. Cut through the film at the drilled holes and then spray it with two coats of clear polyurethane for protection.

TABLE 1: PARTS LIST.

SYMBOL	VALUE	DESCRIPTION	VENDOR	PART NUMBER
C1,2,4,5,7,8	4700pF 50V 5%	Polystyrene radial	Mouser	23PW247
C3,6,9,10	22pF 50V 5%	Ceramic NPO	Mouser	140-50N5-220J
C11	120pF 50V 5%	Polystyrene axial	Mouser	23PS112
C12-19	100n 100V 10%	Polyester Stacked Film	Mouser	581-BN104E0104K
C20-23	470n 100V 10%	Polyester Stacked Film	Mouser	581-BN154E0474K
C25,26	560µF 50V	Aluminum	DK	P11261-ND
C31,32	560µF 35V	Aluminum	DK	P11239-ND
C27,28	1µF 50V	Tantalum	Mouser	581-TAP105K050SCS
C24	22µF 50V	Aluminum	Mouser	140-XAL50V22
C29,30	22µF 25V	Aluminum	Mouser	140-XAL25V22
D1-D11	1N4002	1A 200V	Mouser	625-1N4002
F1	1/4A 3AG	Fuse	RS	270-1002
I1-3	24V 0.04A	Lamp, # 1819	Mouser	606-CM1819
LED1	LED	Red T-1 ¾	Mouser	351-5110
J1		IEC Receptacle, faston	Mouser	161-0707-1-250
J2		Phono Jack, Gold	Mouser	161-4221
R1A,2A,9,10*	3k74 1%	Metal Film	Mouser	271-3.74K
R1B,2B*	549k 1%	Metal Film	Mouser	271-549K
R3,4*	17k8 1%	Metal Film	Mouser	271-17.8K
R5,11,17	301 1%	Metal Film	Mouser	271-301
R6*,18*,19	1k00 1%	Metal Film	Mouser	271-1.00K
R7,8*	3k40 1%	Metal Film	Mouser	271-3.40K
R12 *	887 1%	Metal Film	Mouser	271-887
R13,14*	2k80 1%	Metal Film	Mouser	271-2.80K
R15A,16A*	4k32 1%	Metal Film	Mouser	271-4.32K
R15B,16B*	365k 1%	Metal Film	Mouser	271-365K
R20	1k91 1%	Metal Film	Mouser	271-1.91K
R21	5K	Pot, Conductive Plastic	Mouser	594-148-7502
R22	604 1%	Metal Film	Mouser	271-604
R23-26	47Ω 1W 5%	Metal Oxide	Mouser	281-47
R27	3k9 ½W 5%	Carbon Film	Mouser	293-3.9K
R28,29	121 1%	Metal Film	Mouser	271-121
R30,31	1k78 1%	Metal Film	Mouser	271-1.78K
S1	SPST	Mini Toggle	Mouser	108-MS550K
S2,3	DPDT 3A	Mini Toggle, red tip	Mouser	10TF115
S4,5	DPDT 3A	Mini Toggle, yellow tip	Mouser	10TF215
S6,7	DPDT 3A	Mini Toggle, blue tip	Mouser	10TF315
T1	36 VCT 0.15A	Transformer	Mouser	546-166E36
U14	NE5534AP	Op Amp, Single Audio	DK	296-7042-5-ND
U5	LM317T	Reg, +Adj, TO220	Mouser	512-LM317T
U6	LM337T	Reg, -Adj, TO220	Mouser	512-LM337T
Z		Enclosure, 8 × 3 × 6	R-S	270-274B
Z		Fuse Holder, Panel Mount, Rt Angle	Mouser	441-R328
Z		heatsinks, (2) TO220	Mouser	532-530614B00
Z		Lamp Socket,(3) T3-¼ Mini Insul	Mouser	35LH003
Z		Socket, (4) 8-pin gold WW DIP	Mouser	575-213308
Z		Socket, (4) 8-pin wire wrap DIP	Mouser	575-293308
Z		DIP 8-pin wrap Ids (4)	DK	K470-ND
Z		Phone jack FW 0.26 ID	Mouser	561-D2562
Z		Phone jack insulator 0.26 ID	Mouser	561-SW25
Z		Standoff Nylon (7) 4-40 × ¾	Mouser	561-TSP4
Z		faston terminals (3) for IEC recept		

(* may require select at test parts - see text)

ADDITIONAL ITEMS:

Perfboard, control knob, nylon spacers, LED mounting ring, hookup wire, power cord, solder, hardware, and so on.

Drafting applique' film (Letraset Letracopy Creative or Chartpak DAF8) is available at art or drafting supply houses. Other options are Clear Laser Labels by Avery (various sizes, four-digit part numbers start with #566_), BEL Inc. ink-jet decals (beldecals@bellsouth.net), or Worth Poly polyester laser labels from Worth Data (barcodehq.com).

INITIAL TESTING

Before installing the four op amps, plug in the unit and check for the proper power-supply voltages at the corresponding IC socket pins (+19.8V at pin 7 and -19.8V at pin 4). If all is well, power down the unit (waiting for the power-supply caps to drain) and install the ICs. Apply power again and check for oscillation at U1, U2, and U3 pin 6. You are looking for approximately 2.1V RMS from each oscillator. If any of the oscillators do not operate, you may need to adjust the value of R5, R11, or R17. If you have a THD meter, you can also adjust R5, R11, and R17 for minimum distortion.

Next, check all six operating frequencies. You may need to adjust the values of the series and parallel Wien network resistors to achieve the exact frequencies. The three MIM frequencies are more critical because the two lowest even-order and odd-order IM products (950Hz and 1050Hz) are very close together.

Finally, adjust the values of R6, R12, and R18 for any differences in the oscillator voltage amplitude (you could use a cermet pot here, but lamp life should be unlimited because the voltage across the lamp is but a small fraction of its rated voltage). Each individual oscillator should produce about 6.2V RMS at the output jack with the level control at maximum.

SUPPLIERS

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Local electronics store

FIGURE 4: Front panel designations.



The response flatness is very good for the Wien-bridge oscillator, so there should not be any amplitude difference between the two frequencies from any given oscillator.

MEASUREMENTS

Figure 5 shows the 19 + 20kHz CCIF spectrum of my original IMD tester, with its noise floor rising as high as -90dB. **Figure 6** shows the same CCIF signal from the lamp-based IMD generator that is the subject of this article. **Figure 7** shows the 11 + 12kHz IMD signal, and **Fig. 8** shows the 9 + 10.05 + 20kHz MIM signal. The 0dBfs output in

all cases is 12V p-p. *ax*

REFERENCES

1. "Multi-Tone Intermodulation Meter," TAA 2/89, p. 7, 3/89, p. 13, 3/90, p. 18; Borbely, Erno.
2. "Build your own Audio Oscillator," *Popular Electronics* 7/97, 9/97, 10/97; Miller, Franklin J.
3. "Wien Bridge Based Oscillators," Linear

FIGURE 7: 19 + 20kHz CCIF signal.

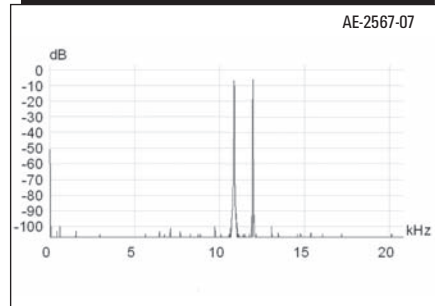
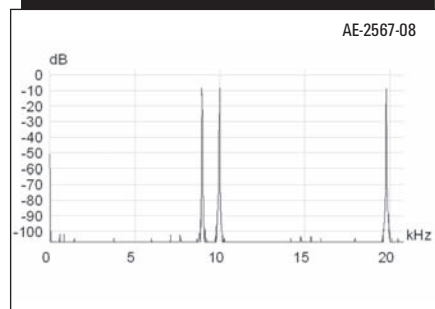


FIGURE 8: 9 + 10.05 + 20kHz MIM signal.



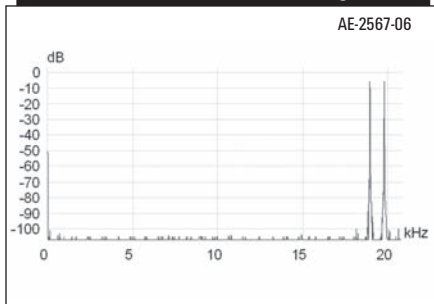
Technology Application Note 43, 1993 *Linear Applications Handbook Volume II*, Figure 39, pp. AN43-29, 30.

4. "Precision Sine Wave Oscillator," Hansen, C., *Audio Electronics* 5/98, pp. 8-23.

FIGURE 5: Original IMD tester 19 + 20kHz CCIF signal.



FIGURE 6: 19 + 20kHz CCIF signal.



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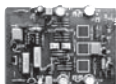
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Dynaco 400 Assimilation

This classic amp benefits from a facelift using today's components.

By Julian Evetts

The Dynaco 400 was the Tyrannosaurus Rex of power amps in 1973. I wanted one badly, but I was bringing up children and settled for the more affordable Dynaco Stereo 80, which pumped out music for the next 20-odd years. It also survived three teen-age boys.

In 1998, I spotted a Dynaco 400 in a used equipment shop, paid way too much, and dragged it home—all 60 pounds! Unfortunately, my “Dynasaur 400” was seriously flawed. It had a nasty habit of going DC and destroying speakers.

PHOTO 1: The rebuilt Dynaco 400.



I spent more money unsuccessfully attempting to repair a unit with too many previous ham-handed repairs and modifications. Eventually I decided it was too old and complicated to fix and too big to use as a doorstop. Resistance was futile; it needed to be assimilated with newer technology (Photo 1).

I decided on a rebuild that would retain the outward appearance of this classic amplifier, while replacing the innards with new components. After looking around, I chose to upgrade using a couple of compact 200W all-MOSFET amplifier modules from Marchand Electronics in Rochester, NY (Photo 2). You can buy the PM224 modules as fully assembled and tested

units, as kits with the PC boards and all parts, or as bare PC boards. See the Marchand site for details, specifications, and schematics.

The rebuild is straightforward. Strip everything but the massive Hammond power transformer from the old chassis (Photo 3). Remove the faceplate and hinge the front part of the chassis forward. Clean the wiring from all front panel con-

PHOTO 2: PM224 Module: 220W RMS into 4 or 8Ω. Better than 0.01% distortion at 1kHz. Differential (balanced) or single-ended inputs. AC or DC coupling (jumper selectable).

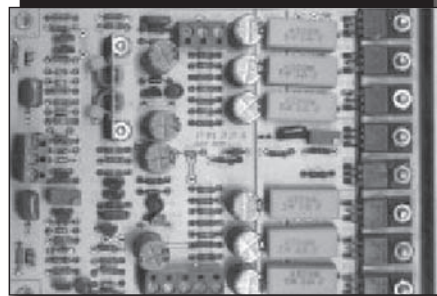


PHOTO 3: The bare chassis with Hammond transformer.

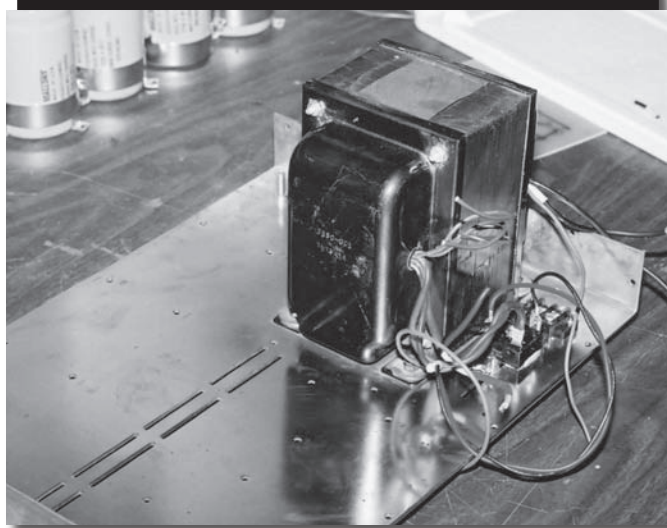
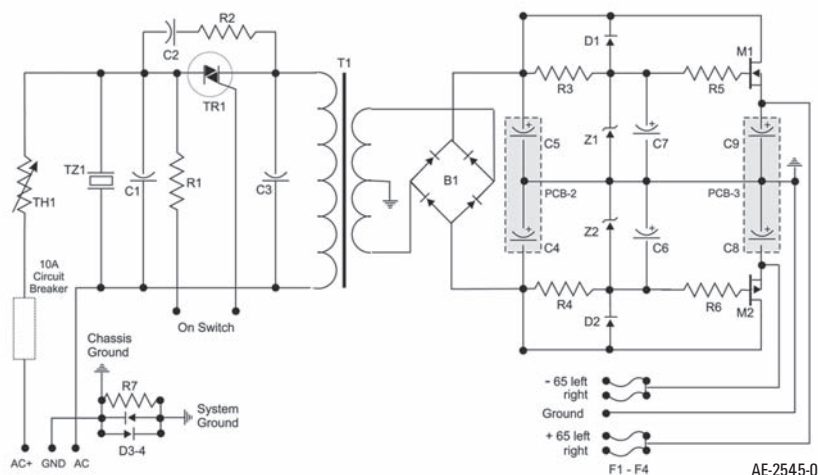


FIGURE 1: Regulated 65V power supply for Marchand PM224 power amplifier modules.



AE-2545-01

trols. Only the power switch and pilot light will be functional in the new amplifier.

The old Dynaco yields about \$200 in parts—a 1,000 VA transformer, heat-sink, rectifier bridge, circuit breaker, switch, speaker fuses, connectors, and a sturdy chassis with lovely big rubber feet. On a good day, you can pick up an old, one-channel-good Dyna 400 for \$50 on eBay; you'll need to talk to the Teamsters about getting it to your place.

Despite the utility of the Dynaco's original 15A power switch, I chose to use a triac to turn the amp on, believing that MOSFETs are more susceptible to switching spikes and noise than the transistors in the old amp. Audiophiles are, as Nelson Pass notes, a superstitious group. The triac power-on circuit is the same as that used in the Pass/Thagard A75. The voltage regulator circuit is taken

PHOTO 4 (top): PCB 1 mounted and voltage regulator MOSFETs bolted to the transformer shield.

PHOTO 5 (center): Power supply elements mounted and wiring ready for modules and heatsink to be attached.

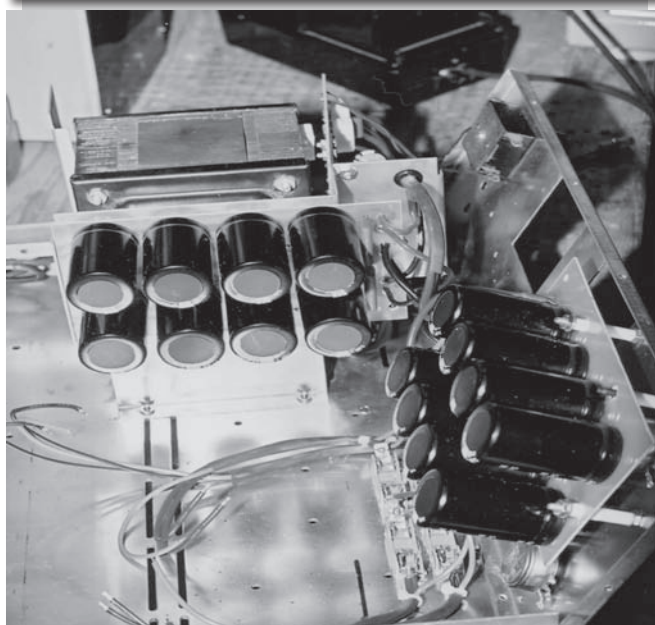
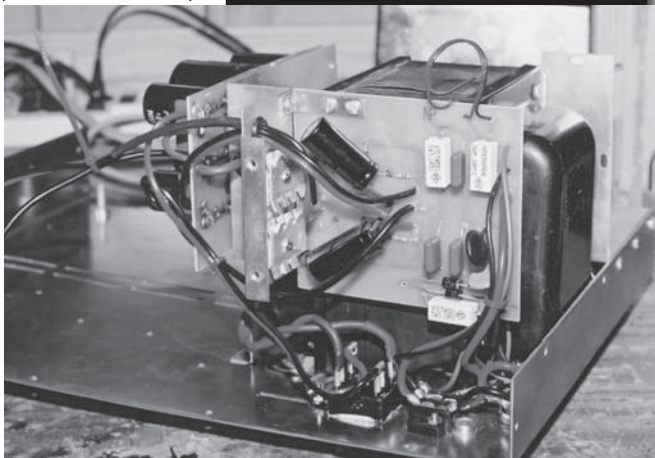
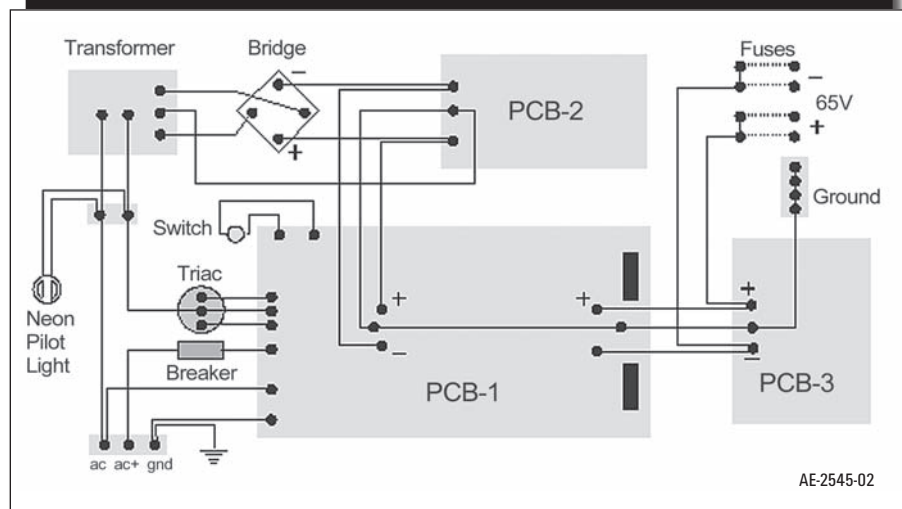


FIGURE 2: 65V power supply—block diagram.



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PHOTO 6: Power amp modules bolted to the heatsink.

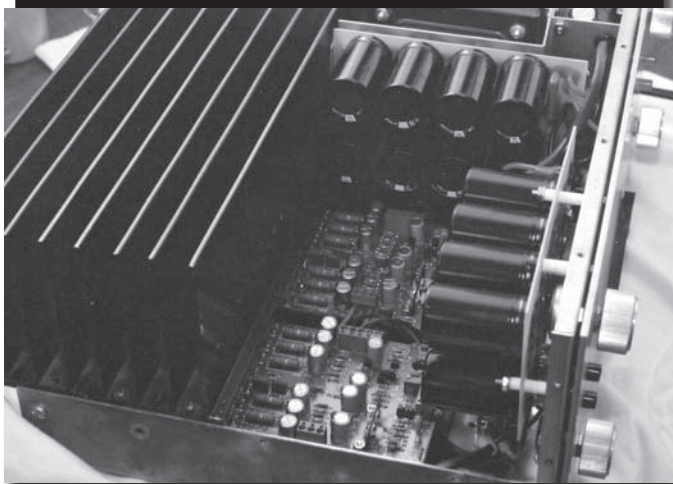
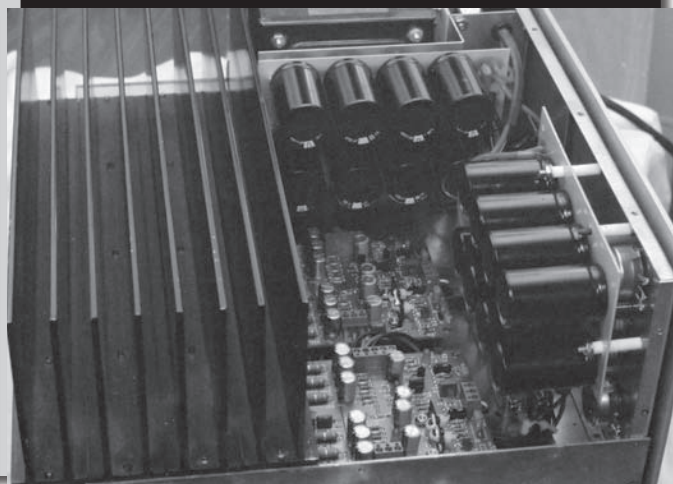


PHOTO 7: Assembled unit. PCB 3 on the right; PCB 2 at the rear; modules below.



from Pass's "Complementary Zen" design. Thanks for both.

Replace the old Dynaco PC30 power distribution board in front of the transformer with the new combined power-on and voltage regulator board, PCB-1. Drill a couple of holes in the transformer shield to bolt down the large voltage regulator MOSFETs (**Photo 4**). I used two old heatsinks as washers

on the top of these devices, but they really don't generate much heat. Place insulating pads or mica washers underneath them.

The large capacitors, C6 and C7, need to be bent over parallel to the board so that they clear the front panel—leave their legs long and cover with heat shrink-wrap. Connect the power switch and pilot light. The original Dynaco pilot is a neon light with a dropping resistor. Connect it across the transformer primary.

The unregulated side of the supply in my amp sits at about $\pm 72V$ (**Photo 5**). The voltage regulator section drops this to a very stable and noise-free $\pm 62V$. This is not the 65V specified in the design, but it is close enough.

Both sets of filter capacitors are mounted on their own PC boards; 40,000 μF before the regulator and 60,000 μF after. PCB 2 fits on the far side of the transformer shield and PCB 3 on the inside of the front panel. You'll need some

spacers to get the front-panel board to sit above the old amp's controls. Use the fuse block from the old amplifier for positive and negative supplies to the left and right amplifier modules.

The two power amplifier modules from Marchand (**Photo 6**) bolt directly onto the base of the oversized Dynaco heatsink. They sit an inch above the floor of the chassis when the heatsink is replaced. They are

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PARTS LIST - Dynaco 400 Assimilation.

- 1 TH1 Thermistor 2.5 Ω - 8A Inrush limiter - (Thermometrics CL 30)
- 1 TZ1 MOV - 250V - 2,500A Transient voltage suppressor (Siemens 510K250)
- 1 T1 Transformer Hammond 1,000 VA, 140V, split secondary
- 1 B1 30A - 250V Rectifier bridge
- 3 C1, 2, 3 .047 μF - 250V Interference suppression capacitor - metalized film
- 8 C4,5 4,700 μF - 100V Electrolytic (Panasonic TSUP - ECO-S2AP472DA)
Capacitors (A-D) and (F-I) on PCB 2
- 2 C6, 7 1,000 μF - 100V Electrolytic - Axial - long leads
- 8 C8, 9 6,800 μF - 80V Electrolytic (Panasonic TSUP - ECO-S1KP682DA)
Capacitors (A-D) and (F-I) on PCB 3
- 4 C4,5,8,9 0.1 μF - 250V Interference suppression capacitor - metalized film
Capacitors (E) and (J) on both PCB 2 and 3
- 3 R1, 2, 7 5 Ω - 2 or 5W Small power resistors
- 2 R3, 4 2.2k Ω - 1/4W Carbon
- 2 R5, 6 221 Ω - 1/4W Carbon
- 1 M1 IRFP 140 IR - MOSFET - N Channel- 23A - 100V
- 1 M2 IRFP 9140 IR - MOSFET - P Channel - 23A - 100V
- 2-Mounting kits or mica washers for M1,2 - T0-247 case
- 2 Z1,2 Zener diode 68V - 1W
- 2 D1, 2 1N4004 - Diode - 1A
- 2 D3, 4 1N5401 - Diode - 3A
- 4 F1, 2, 3, 4 Fuses 3A, fast blow
- Miscellaneous: terminal blocks, heatsink compound, wire, ties, hardware.

very compact and fit well. You can wire the inputs and outputs of the amp modules before sliding the heat-sink back into the chassis.

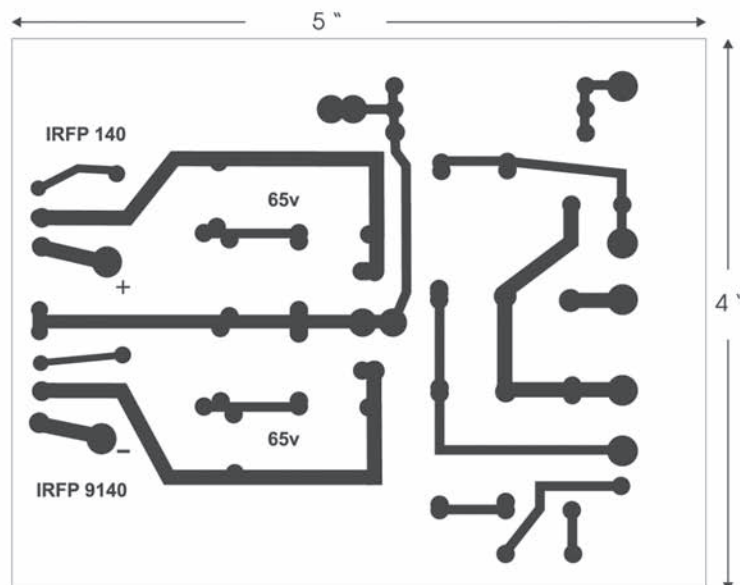
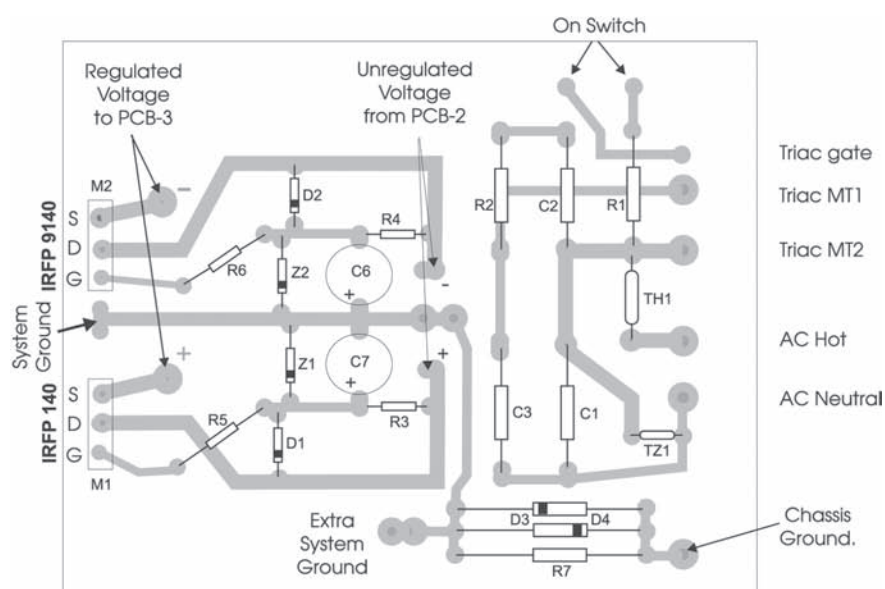
Thanks to quick-connect strips on the boards, hooking up power and ground wires is a two-minute

PHOTO 8: Assimilated "Dynasaur 400" unit returned to service. Shown here slowly burning in on a mix of Asterix, Orbital, Delerium, Enigma, and Infected Mushroom.



FIGURE 3 (top): PCB 1 voltage regulator, top side.

FIGURE 4 (bottom): PCB 1 voltage regulator, copper side.



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job. As with the original amplifier, this one opens up nicely for service. The front panel tilts forward for access to the power supply. The heatsink and amplifier modules drop out as a single part once the power and ground wires are disconnected (**Photo 7**).

For now, I am very happy with the rebuilding effort (**Photo 8**). In many respects, the rebuild performs better than the original amplifier and runs much cooler. For the budget conscious, this is a 400W high fidelity amplifier that costs less than a dollar a watt. For people who like to

party, this amp really makes a couple of 15" drivers jump. I'm sure glad the kids are not around to mess with my sound system any more. *aX*

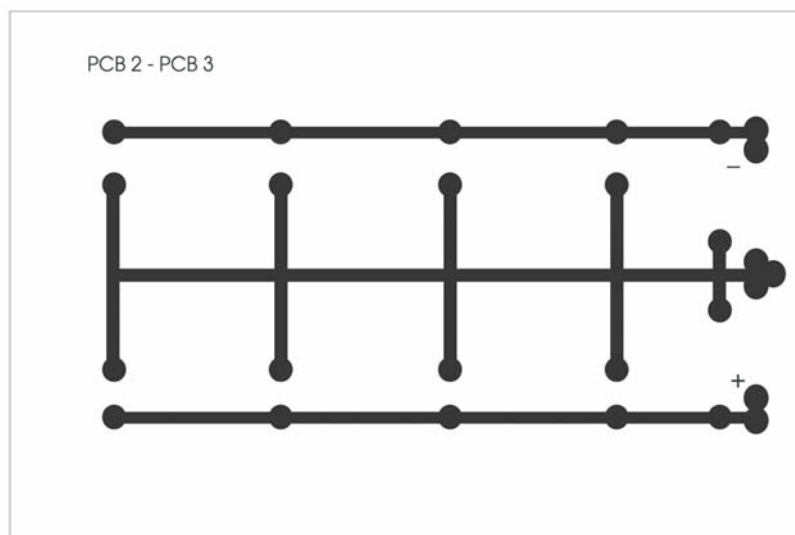
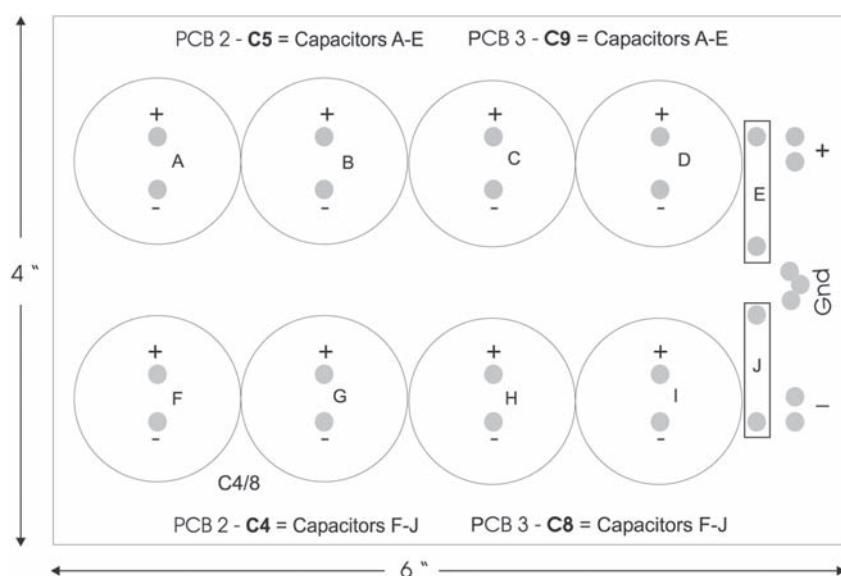
Sources:

Pass, Nelson, "Zen Variations, Part 5: The Complementary Zen," Oct. '03 *audioXpress*.

Thagard, Norman, and Pass, Nelson, "Build the A75 Amplifier," *Audio Amateur* 4/92 and 1/93.

The Unofficial Dynaco Home Page—<http://home.indy.net/~gregdunn/dynaco/>.

FIGURE 5 (top): PCBs 2 and 3 filter capacitors, top side.
FIGURE 6 (bottom): PCBs 2 and 3 filter capacitors, copper side.



AE-2545-05,06

MAKING PC BOARDS

IT'S EASY to make good-looking printed circuit boards using the photofabrication technique.

Print the Artwork – You will need a few sheets of ordinary transparency film—the kind they use to make slides for overhead projectors. Buy it at Staples or Office Depot. You can use a laser or an ink-jet, but make sure that



the type of transparency film matches the printer. I find that I get the best results from ink-jet film in an ink-jet printer set to enhanced mode. The PCB artwork will print full size from this Acrobat file.

The drawings are dimensioned, so check them with a ruler to make sure your printer isn't scaling. Cut out the 4 × 6 design for the copper side of the filter capacitor boards and the 4 × 5 design for the copper side of PCB-1. The photosensitive boards are an inch larger on one side than the designs. Cut them to size after they have been etched.

Expose and develop the boards – Follow the instructions that come with the photosensitive boards. Expose and develop one board at a time. Remove the protective film from the copper side of

Marchand Electronics, Rochester, New York—<http://www.marchandelec.com/>.

Manual—PM224 Power Amplifier Module—<http://www.marchandelec.com/ftp/pm224man.pdf>

Julian Evetts lives in Calgary, Alberta, and works as a consultant and job skills analyst. He grew up on a prairie farm with no electricity, which perhaps deepened his appreciation when he finally did get it. He was bitten early by the electronics bug, built a first crystal set at ten, and due to circumstances, patched together many battery-powered radios in the years that followed. Thus deprived of high-powered amplifiers in childhood, he has perhaps overcompensated in adulthood. He likes to listen to electronica, techno, and trance.

the first presensitized board. Cover it with the artwork and lay a sheet of glass on top to keep the artwork snug to the board. Expose under a fluorescent light for about ten minutes. I always place text in the artwork for PC boards to make it less likely that I will reverse the artwork and produce a mirror-image board.

At the end of the exposure period, remove the glass and the artwork from the board. Dip the board in the developer solution and gently wash away the photoresist to reveal the copper underneath. Rinse immediately and completely under cold running water to stop the process. Expose and develop the other two boards.

Etch the boards using the ferric chloride solution following the instructions on the label. This is the only chemical I have seen that will destroy a stainless steel sink, so be really careful.

Before starting, warm the etching solution by letting the bottle stand in hot water. Put the PC board or boards in the etching tray and cover with warm ferric chloride. Agitate to speed the etching process.

When the copper around the design has been etched away, rinse the board in cold water and scrub lightly with steel wool. Mark the boards and cut to final size—that's an inch off one edge of each board. Drill holes from the copper side of the boards and you're finished.

Photofabrication supplies – Buy a photofabrication kit, such as MG-416K (Photo A). It has everything you will need:

boards, developer, ferric chloride, gloves, and a nifty developing-etching tray.

Otherwise: 1 - Developer (MG - 418)

1 - Ferric Chloride (MG-435)

1 - 4 × 6 presensitized single-sided PCB (MG - 606)

2 - 6 × 6 presensitized single-sided PCB (MG - 609)

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www.mgchemicals.com/techsupport/index_proto.html ■

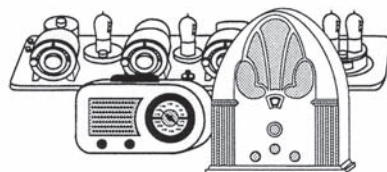
Photo A: Photofabrication Kit from MG Chemicals.

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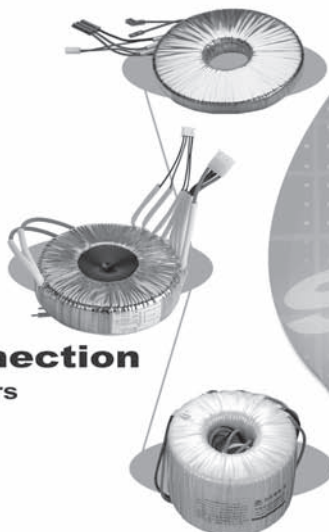


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Lowering A Loudspeaker's Mechanical Noise Floor, Part 2

Discover how reducing panel vibrations will improve the sound of your system.

By Mark Sanfilipo

In Part I I looked at the maths and theory underpinning the mechanics of panel vibration. Here in Part II, I'll take an empirical look at panel vibration, investigating with both accelerometer and microphone. The mechanical and acoustical performance of a loudspeaker cabinet's panel will be assessed at various stages of construction, with an eye to noting any improvements along the way. This report will conclude with a simple before/after comparison of the acoustical output of one of the test cabinet's panels, followed by a subjective assessment of the system's performance.

DAMPING

There are two modes of mechanical damping commonly employed in loudspeaker construction: *extensional* and *shearing*.

In *extensional* mode (free substrate layer) a second, viscoelastic material absorbs kinetic (vibrational) energy from, in this case, a vibrating plywood panel substrate, converting it into heat

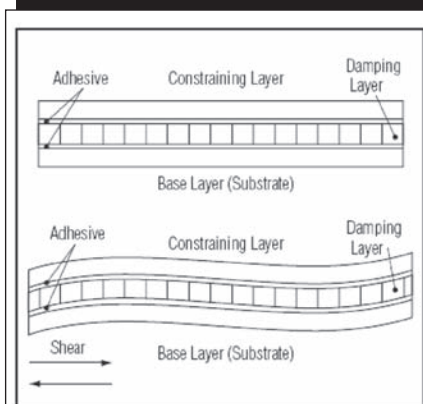
(Fig. 1). The effectiveness of this approach depends upon the degree of panel flexure as well as the mechanical and physical properties of both the damping and substrate layers. Additionally, it's useful to know the mechanical properties of the adhesive used: if the mechanical impedance of the adhesive is low enough, the damping pad will function as a constraining layer. Whether this is actually desirable or not can only be determined by further measurement.

Shearing mode (constrained layer damping), as seen in Fig. 2, is an alternate approach in which a vibrating panel is constrained by a second panel of sufficiently high mechanical impedance. In this mode, the two panels—one vibrating, the other constraining—are joined by an adhesive viscoelastic damping layer. Energy is dissipated through tensional forces of shear deformation.

Intuitively, it might appear the thicker the adhesive damping layer, the more damping. In practice, however, the opposite holds true: the thinner the layer, the more effective constrained layer damping (CLD).

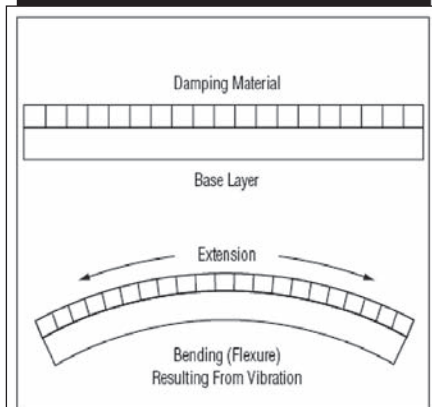
So which method is best? In my ex-

FIGURE 2: Shearing mode.



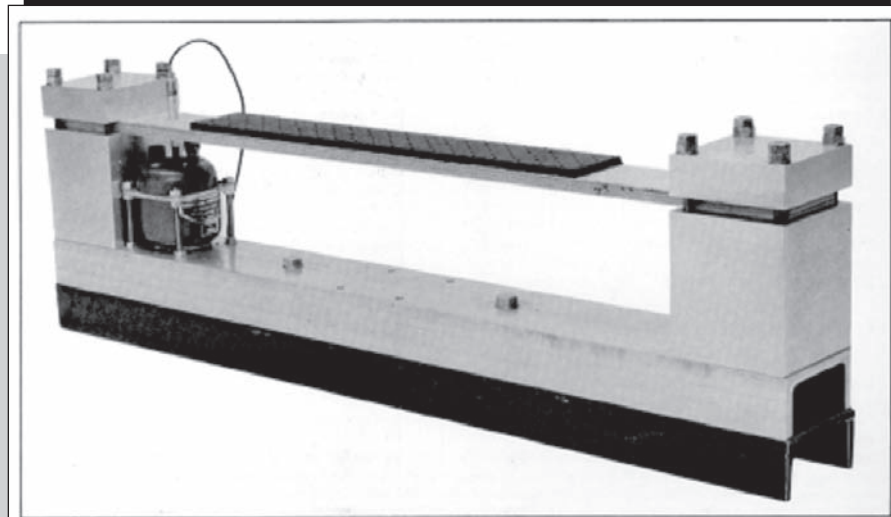
SB-2592-02

FIGURE 1: Extensional mode.



SB-2592-01

PHOTO 1: Sample test rig (courtesy BBC).



perience (and all else being equal) CLD has generally provided for a higher degree of panel damping when compared to extensional mode damping, as applied to plywood cabinets constructed for mid-bass or even LF purposes. However, it's more costly and less forgiving regarding the selection of the appropriate materials. If your budget allows for it, I recommend you experiment with both approaches. If your budget is tight, stick with the less expensive, non-CLD approach, which works well, as you'll see.

TEST RIG

In order to investigate the vibration-damping effectiveness of various materials, I built a test rig that allowed for quick, repeatable measurements of various, commonly available North American plywood panel samples along with various common, commercial damping materials. Similar in appearance to that shown in **Photo 1**, the rig included two solid, heavy mount points to which the rectangular-shaped panel samples were clamped at each end.

The panel samples were set in motion by the impact of a 100 gram disk-shaped weight, dropped from a height of 30cm. The test panels were angled 45° along their long axis. In doing so, the disk, when dropped on its edge, would impact, then fly off the panel, coming to rest off the panel, thus not interfering with the post-impact vibrational behavior of the individual wood samples.

Working up a series of before/after comparisons, as seen in **Figs. 3 and 4**, didn't require absolute measurement values; therefore, using an uncalibrated accelerometer is a valid approach. I could have used a calibrated accelerometer but chose instead to experiment with an inexpensive accelerometer I'd seen mentioned in both online forums and print media.

The Measurement Specialties ACH-01 worked quite well, and the $\pm 3\text{dB}$ 2Hz-20kHz typical amplitude response, along with a $\pm 150\text{g}$ dynamic range, made it a very cost-effective, suitable choice. It's available through retailers such as Digi-Key and makes for a worthwhile addition to any DIY audiophile's toolkit.

FIGURE 3: Plain board vs. damping material, lightly placed.

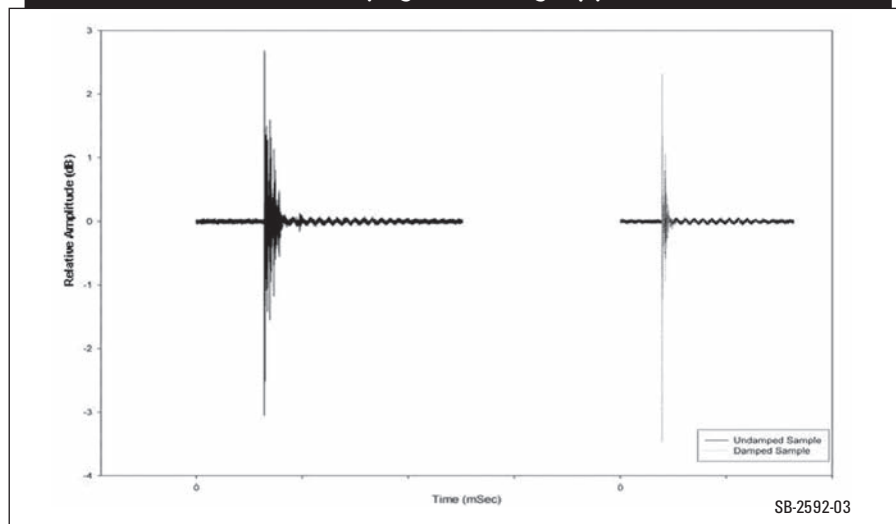
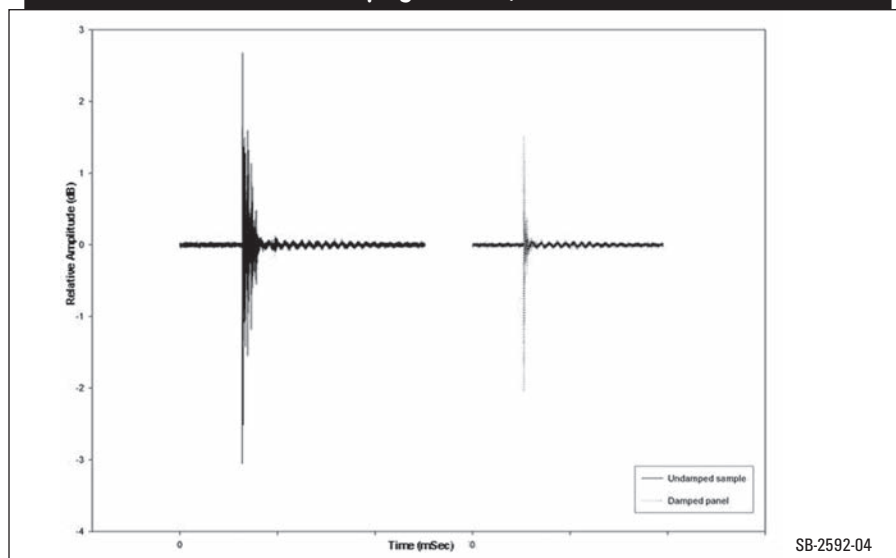


FIGURE 4: Plain board vs. damping material, self-adhered.



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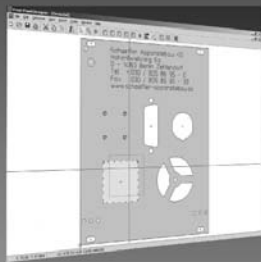


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The wood I chose for this project from the half-dozen test sample candidates was 19mm (.75") thick, void-free Baltic birch. I frequently use this (or its marine-grade derivative) and have found it to be a consistently top-quality engineered wood product (**Photo 2**).

The damping material used in this case was a multi-ply "panel" built up from heavy, self-adhering roofing material, which, sold in long rolls, you can easily cut to fit as needed (**Photo 3**). Multiply the amount shown by 5.5 to get an idea of the total amount used in the construction of both cabinets.

The half-dozen plywood test samples used were (free length) approximately 70cm (27.6") long and just over 12cm (4.7") wide. I chose the free length as being a reasonable approximation of a typical loudspeaker cabinet's panel height. I chose the width to keep mechanical impedance low and to minimize transverse resonance modes. The results of the test series shown in **Figs. 3 and 4** are taken from the Baltic birch test sample.

ACCELEROMETER RESULTS

Whether you use an inexpensive accelerometer, such as the ACH-01, or a more expensive calibrated accelerometer, a few caveats are in order. First, due to the sensitivity of accelerometers suitable for measuring loudspeaker panel vibration, it's easy for extraneous vibrational noise from nearby fans,

motors, traffic, and so on, to creep into your measurements. Second, be consistent in your choice of base or support for your cabinet under test. Measure your cabinet while it rests on, for example, the concrete floor of a garage. Then re-measure it with the cabinet perched on top of a wooden workbench. This will produce different results, invalidating even comparisons based on relative amplitude measurements.

PHOTO 2: Sample Baltic birch panels.

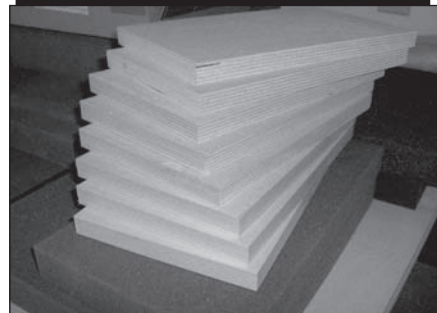


PHOTO 3: Sample damping material.

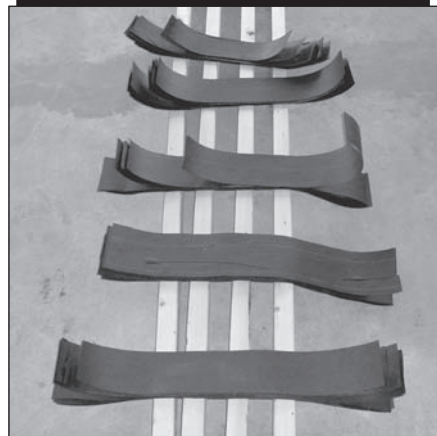
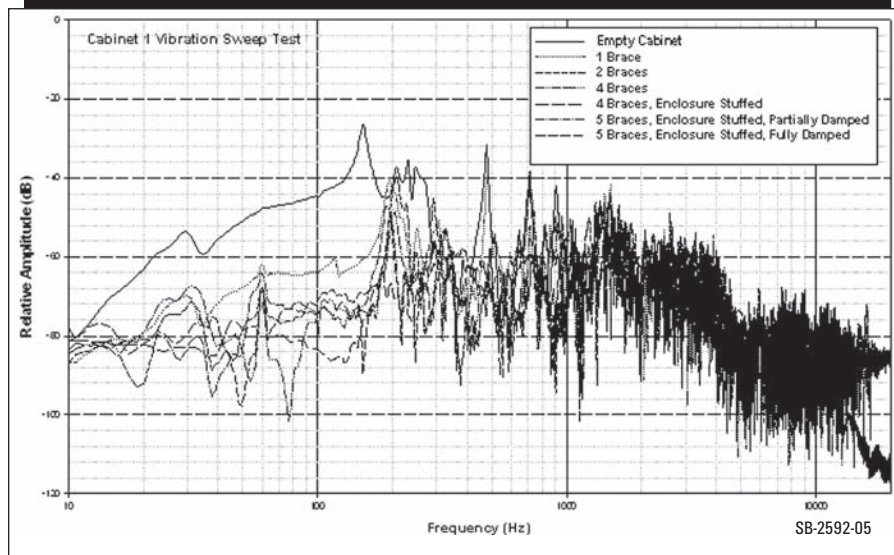


FIGURE 5: Vibrational panel responses.



On the left-hand side of **Fig. 3**, you see the amplitude response generated by the accelerometer fixed to the undamped panel sample. On the right-hand side is the response of the same sample but with the damping material lightly affixed to the panel. (Both impulse responses are on the same amplitude/time scale.) Already a significant difference is evident. Clearly, the plywood panel, when damped, settles much more quickly than when undamped.

The damped response in **Fig. 7** resulted when the damping material was directly adhered to the plywood test sample. In **Fig. 4**, you can see an improved settling time when comparing the damped amplitude response with that in **Fig. 3**. Clearly, adhering each layer to its neighbor, then adhering the multi-ply damping panel directly to the plywood test sample is preferable.

It's tempting to think that when it comes to the application of a damping material, the more the better. However, there exists a point of diminishing returns, where more is not better. From an empirical standpoint, building a damping pad of the particular material used in this project one-half to three-fourths the thickness of the plywood panel substrate is a good starting point when setting out to determine optimal thickness.

BRACING

After I assessed the effectiveness of the free-layer damping approach, it was time to build the test enclosures.

I designed the enclosure to have four solid cross braces, oriented horizontally, fixed to the interior side of the cabinet's side and back panels, with the front edge close to, but not touching, the inside of the faceplate. An FEA simulation done prior to construction indicated four cross braces would produce acceptable results.

To gauge the effectiveness of the bracing and damping approaches, I made a series of measurements at different points in the construction process using the ACH-01 accelerometer, along with a calibrated microphone for capturing acoustic measurements, useful in subsequent analysis. The enclosure sported double-thickness (38mm)

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FIGURE 6: Before/after sweep results.

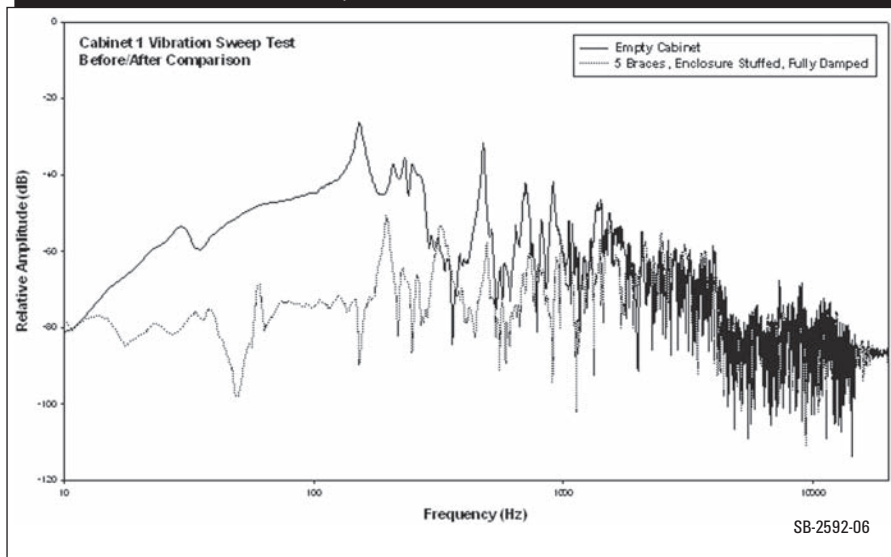


FIGURE 7: Before/after impulse comparison.

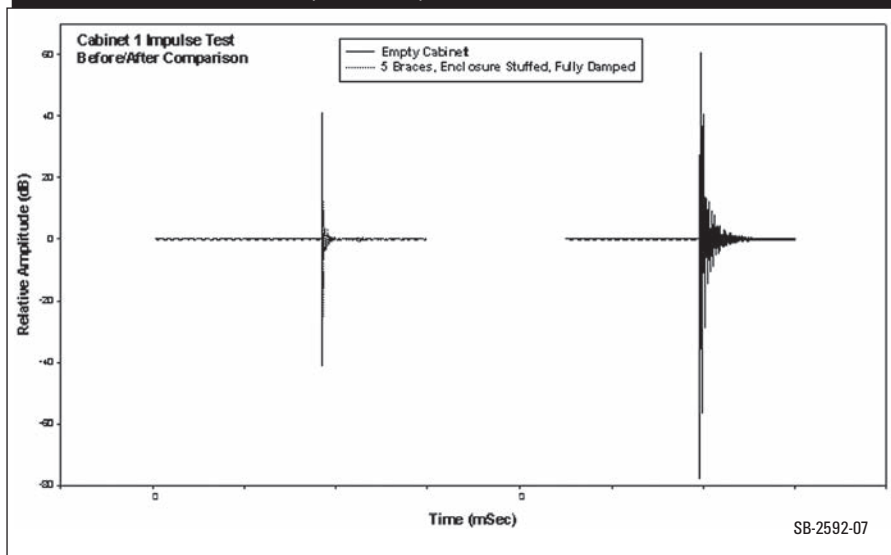


FIGURE 8: Spectral analysis, before/after knuckle rap test.

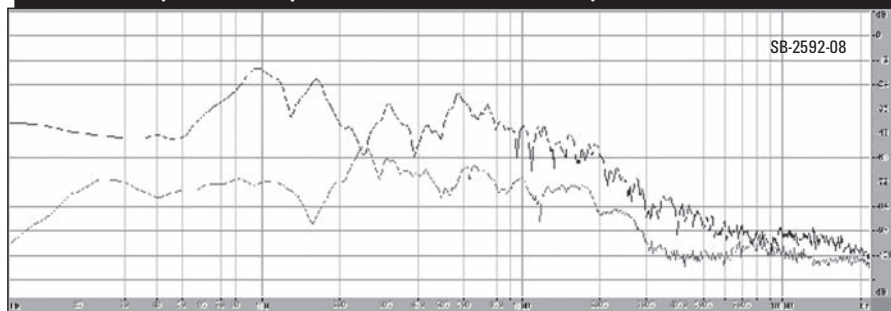
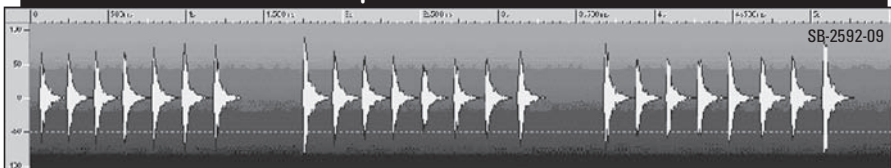


FIGURE 9: "Before" knuckle rap test.



front, top, and bottom panels, with the sides and back being single thickness (19mm) panels. **Photo 4** shows one of the enclosures midway through construction.

In addition to the four horizontal, shelf-style braces, the front panel was cross-braced as well, with a thin shelf style 5th brace running from the inside center point of the faceplate to the back of the enclosure. The use of extensive bracing and damping added to the already substantial weight of the cabinets. The completed enclosures weighed over 100kg (220 lbs) each!

In **Fig. 5** you can see an heroically busy collection of plots, each sweep indicating the vibrational behavior of the panel to which the accelerometer had been affixed (the center of the left or top panel in **Photo 4**) at various stages of construction. You can see the net before/after result of all the bracing, damping, and stuffing in **Fig. 6**.

In **Fig. 6**, note a general downward shift in amplitude, accompanied by an upward shift in frequency when comparing the "after" curve with the "before" curve. Keep in mind these are not dB SPL curves, but the relative response of an accelerometer responding to the cabinet panel's acceleration at each frequency along the sweep.

Figure 7 is the "before" and "after" vibrational response to impulse stimulation. In this particular graph, "before" is the response plotted on the right, the "after" plotted on the left. The improvement in settling is striking.

Thus far, you've seen what the accelerometer has had to say. To see what the microphone has had to say, I employed the noble knuckle-rap test. I removed the accelerometer, rapped away at the spot formerly occupied by it, with microphone positioned nearby. In **Fig. 8**, the "before" curve (top) presents a generally higher relative amplitude response than the "after" curve (bottom). The difference, however, sounded much greater than it appears here.

Figures 9 and 10 are the time-domain versions of the data presented in **Fig. 8**.

CLOSING COMMENTS

Measurement indicated that the various readings taken to reduce panel vibration were effective in lowering the

mechanical noise floor of the system. However, objective assessment tells only part of the story, and subjective assessment is required to complete a picture of any system's performance. **Photo 5** shows the test system.

Though the cabinets are attractive in their unfinished state, with an appearance reminiscent of, for example, some of Axiom's current line of products, they will eventually have a finish

ABOUT THE KEF B139 DRIVERS

I first became acquainted with KEF B139 drivers back in the late 1970s. In those days I was working in a recording studio "programming" a Moog synthesizer. ("Programming" back then meant plugging the Moog's patch panels full of patch cords.) I remember carrying around a three-ring binder full of preferred patches. Ancient history!

The synthesizer fed its output, via a Dynaco power amp, into a pair of IMF transmission-line loudspeakers that used the B-139 as the woofer. The bass those things could crank! My teenage ears were impressed.

Outside of live music situations, I had never heard such low bass at such volumes. (Remember, we're talking late '70s here; monster subs were not as commonly available as they are today.) Years later I would see B139s used in Wilson Audio's \$100k + US WAMM loudspeaker system.

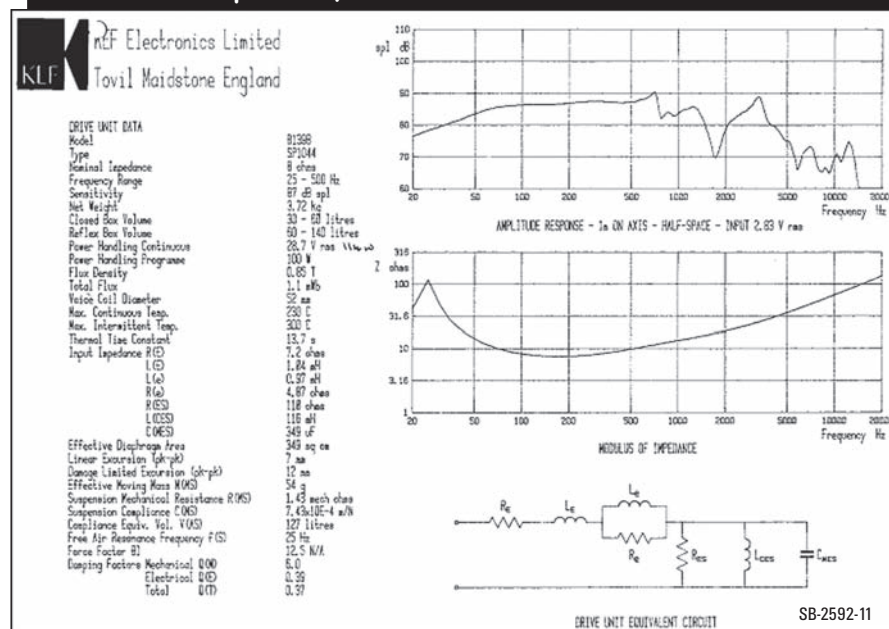
Over the years I acquired half a dozen of the units for my driver collection. I used them mainly for research purposes. Because they had just been lying around gathering dust, when it came time to design a general-use system for the house, it was a no-brainer to design the mid-bass modules for the system around them.

Though technology has certainly provided for many substantial advances in driver quality since the B139s were first introduced over 30 years ago, there are some things they do well to this day.

FIGURE 10: "After" knuckle rap test.



FIGURE 11: B139 spec sheet, ca 1980s.



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applied. The choice of finish, though, is left to my wife—an effective approach for increasing WAF or SAF.

I conducted two listening tests, each lasting about an hour, using a variety of material with which I and four other listeners were very familiar. The upstream test gear used in both instances was the same (Hafler DH-500 power amp, and so on).

At the conclusion of both tests, results were tabulated. From the raw data collected, I threw out everything people *thought* they heard. Due to the

very informal, unscientific nature of the test trials, I was interested only in certainties. I also discarded data for which no consensus could be reached. In the end a few things did emerge.

First, resolution of low-level detail was dramatically improved. This was *not* some at-the-fringe, barely discernible effect; it was a substantial improvement. On this point, all listeners were unanimous. I heard things in recordings I have listened to for *decades* that I had not heard before.

Second, overall mid-bass clarity

improved, resulting in, among other things, improved sound staging. Individual instruments, such as cellos in classical recordings were now discernible as individual instruments, where before they tended to sound more like massed string choruses, made up of not clearly identifiable individuals. The tonal differences between different pianos, such as Steinway, Yamaha, Blüthner, and so on, was now clearly discernible.

It appeared that the “before” mechanical noise floor had imparted a euphonic sameness to everything played through the system. Now that “sameness” had been largely, though not entirely, eliminated. Overall, the system’s presentation had become more realistic and musical, resulting in a more satisfying listening experience. Not bad when you consider how old a technology the KEF B139 drivers represent. *ax*

PHOTO 4: Sample enclosure being constructed.

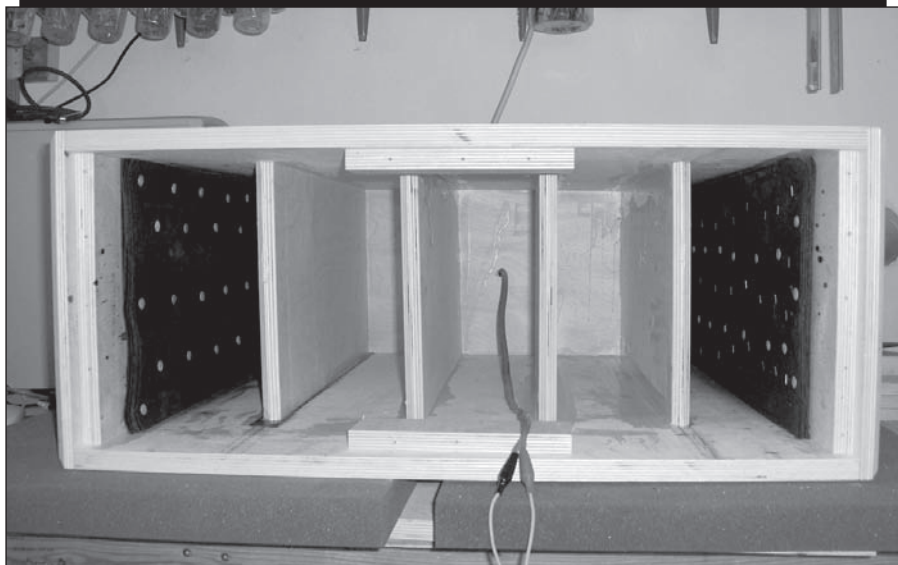


PHOTO 5: Nearly completed cabinets, with the author’s 7-year-old daughter, JB, standing beside cabinets for a sense of scale.



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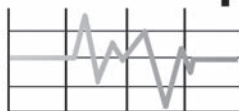
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Following the Selectronic Triphon, Quad Amp, and GrandMOS product reviews in *audioXpress*^{1,2}, Selectronic decided to extrapolate the Quad Amp into a full-range power amplifier with the same amplification module but utilizing a larger power supply. The result is the pure Class-A ProFet amp, available either in a stereo 2 x 15W version or mono 50W (bridged) version. This review features the mono 50W amplifier (the front nameplate with "60W RMS" is an early version of the re-rated amplifier).

The ProFet has no capacitors in the signal path, so any DC voltage offset in your preamplifier will be amplified by the ProFet. If this is a problem, a link capacitor will be necessary (Selectronic advises you to consult your dealer).

Selectronic also advises that the ProFet is designed to give its best performance at a stabilized temperature. This will be attained after a warmup

of 10 to 15 minutes. They suggest that you keep the power permanently on if you are listening to music frequently. They also recommended not to connect or disconnect anything while the ProFet is operating.

I don't know whether kit versions of the ProFet amplifier are available,

as they were with the earlier components. The review samples of the ProFet units were pre-assembled.

INSIDE THE AMPLIFIER

Photo 1 shows the front panel, which has only a blue LED indicator to the right of the ProFet name. The front



PHOTO 1: Front view of unit.



PHOTO 2: Rear view of unit.

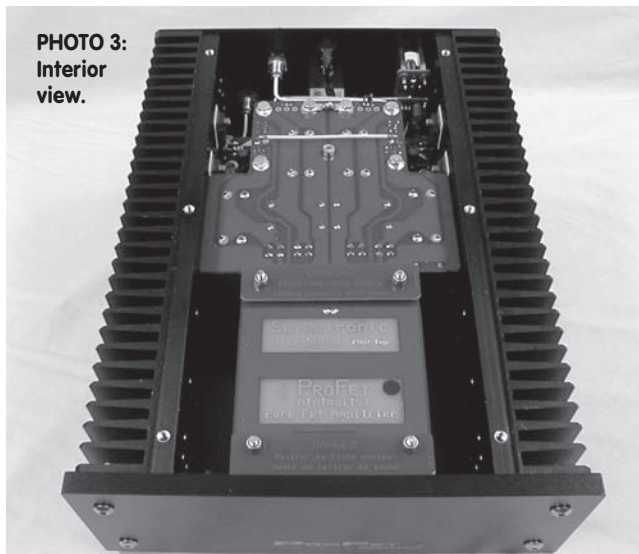


PHOTO 3: Interior view.

panel is 9mm thick black anodized aluminum, and the unit is similar in construction to the GRAND MOS, Triphon, and Quad.

The back and bottom of the crossover are constructed of 2mm black-painted steel. Heavy finned aluminum heatsinks are used along each side, and the 1.5mm steel top is perforated to enhance cooling. The unit is very stiff and rugged, even with the top removed. The bottom rails of the heatsink serve as the "feet," and there is adequate finger space under the unit to easily lift it. The heatsink fins have no uncomfortable sharp edges when you move the amplifier about.

The rear panel (**Photo 2**) has the IEC power receptacle, a recessed high-quality gold-plated Teflon-insulated RCA unbalanced input jack, an XLR balanced input jack, and a pair of high-quality gold-plated speaker binding posts. The speaker binding posts, in accordance with EU requirements, are not on US 0.75" spacings, so you cannot use dual banana plugs. The third pin of the AC receptacle is connected to the chassis. The power transformer primary was factory-wired for 120V mains.

Photo 3 shows the crossover with the cover removed. The ProFet is a bridge-tied-load (BTL) design. The entire audio circuitry, including the inputs and speaker jacks, float with respect to the chassis. The large Huiran R-type power transformer occupies the center front of the chassis.

The T-shaped power supply PC board (Manufacturer's note: The T-shaped pcbs are double sided FR4 epoxy with 70µm copper) uses eight 10,000µF reservoir capacitors. The power transformer secondary windings and the DC power to the amplifier boards are hard-wired. The power supply transformer uses a Faraday shield between the primary and secondary windings to reduce the interwinding coupling capacitance.

The two double-sided amplifier PC boards are located along the sides of the unit. Each amplifier board has two JFET drivers with bias adjust pots and two MOSFET output transistors. The capacitors are Wima polystyrene and 1% silver mica types. All

resistors are 1% or better metal film.

Very little interconnecting wiring is used. Connections between the rear panel speaker binding posts and circuit boards are made with Teflon-covered solid wire.

TOPOLOGY

A schematic was not furnished with the unit, but the topology is described as being similar to the Triphon Quad amp. Indeed, there are only four active parts on each board: two matched TO-92 JFET drivers and one each J162 and K1058 power MOSFETs mounted on each heatsink.

A small PC board mounted behind the RCA and XLR input connector board has a Burr-Brown DRV134 balanced line driver IC and two non-polar aluminum supply filter capacitors. The DRV134 is used to convert the unbalanced RCA input to the balanced signal needed to drive the differential BTL power amplifier.

The selection of the desired input is made by means of the miniature toggle switch placed between the two

input connectors. The toggle switch selects either the XLR (directly) or the balanced output signal of the DRV134.

The main power supply board contains a total of 80,000µF distributed over eight reservoir capacitors. The ±15V linear regulator ICs for the DRV134 are located on the right rear of the main power supply board.

MEASUREMENTS

Making measurements on the ProFet presented a number of challenges. Neither the input jacks nor the BTL output stage has a common connection to ground or to the chassis, so I needed to work out a ground isolation scheme that allowed me to make measurements on the audio output without shorting one of the speaker connections to ground. Along with that was the concomitant requirement not to introduce excessive hum or noise. My distortion test set has an input ground lift switch, and I ended up using my laptop for the computer-based DSO because the desktop com-



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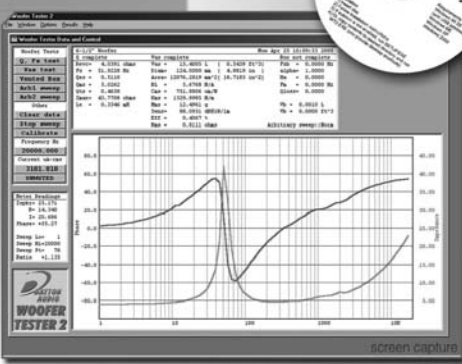
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FIGURE 1: Frequency response.

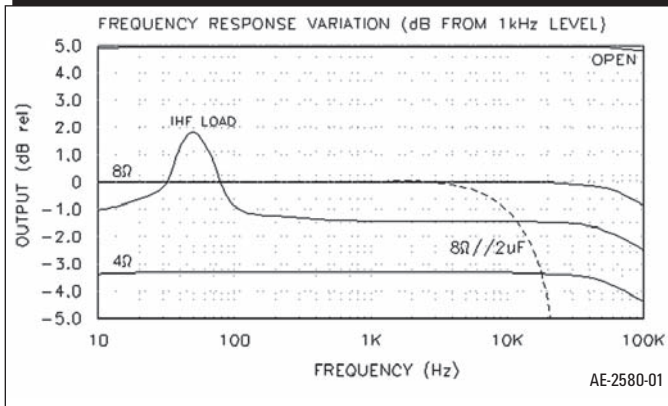
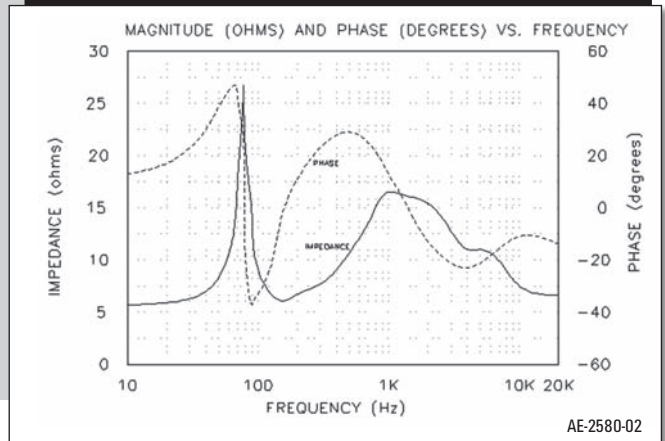


FIGURE 2: NHT SuperOne impedance and phase.



puter I usually use eventually makes its way to power line ground through the data cable. In order to monitor things with my analog scope and for some measurements, I needed to use a differential plug-in.

Selectronic recommends that you power the ProFet at least 15 minutes before listening. I operated the ProFet amplifier at 10W into 8Ω for one hour. While the ProFet is pure Class-A, the heatsink temperatures increased to only 40° C (104° F). The THD+N reading at the end of this run-in period was a bit lower than when I began: 0.52% versus 0.61%. There is only a low-level pop when starting up or shutting down the amplifier.

Output hum and noise measured a low 0.38mV (-84dB, unbalanced input terminated with 600Ω) and was inaudible with my ear against the speaker. I also measured -12mV of DC offset, which is quite respectable for a direct-coupled amplifier without a servo circuit.

The low unbalanced input impedance of 10kΩ may reduce the LF response of capacitor-coupled output sources or preamps. The balanced XLR input impedance of 1kΩ is a bit higher than the usual 600Ω.

The unbalanced input gain at 2.83V RMS output into 4Ω and 8Ω loads was 23.1dB and 26.6dB, respectively. The overall voltage gain of the ProFet

from its balanced XLR input consistently measured 0.5dB higher than that of the unbalanced input. Because the DRV134 chip has 6dB of gain, this small difference may be due to the 1k resistors at the common arms of the input select toggle switch that loads the DRV134. The ProFet does not invert polarity from either input. The XLR input uses the AES standard pin 2 hot. The output impedance was a high 6R8 at both 1kHz and 20kHz.

The frequency response (Fig. 1) was within -1dB from DC to 110kHz, at an output of 2.83V RMS at 1kHz into 8Ω, and wasn't down -3dB until 317kHz. The response into a 4Ω load was -3.3dB lower. The response with a complex load of 8Ω paralleled with a 2μF cap (a test of compatibility with electrostatic speakers) was essentially the same as the 8Ω load alone, until 3kHz, where it began a steep rolloff. There was no noticeable problem driving this load.

The IHF speaker load, which has an impedance peak at 50Hz, produced more than 3dB total change in the in-band frequency response. The ProFet amplifier will be quite sensitive to variations in speaker impedance with frequency. Damping factor is just over 1 across the entire audio band.

Dennis Colin made a good suggestion to perform THD testing with a loudspeaker³. As part of my measurements, I connected one of my NHT SuperOne satellites through 6' of 12-gauge Monster cable. The impedance and phase of the SuperOne versus frequency is shown in Fig. 2.

THD+N vs. frequency is shown in Fig. 3 for the loads indicated on the

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graph. During distortion testing, I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. I measured all data with the output voltage held constant. With the resistive loads, the THD is commendably flat over the measured frequency range. The high output impedance of the Pro-Fet produces the roller-coaster THD curves caused by the interaction between the amplifier and the varying impedance of the IHF and NHT speaker loads with frequency.

Figure 4 shows THD+N versus output power for the loads and frequencies shown in the graph. There was absolutely no strain right up to the point of maximum power. The amplifier reached its 1% clipping point at 53.6W with the 8Ω load for +0.3 dBV

FIGURE 3: THD+N vs. frequency.

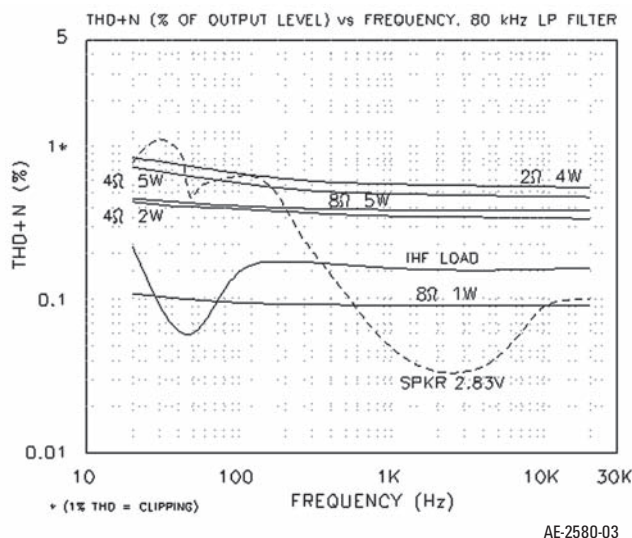
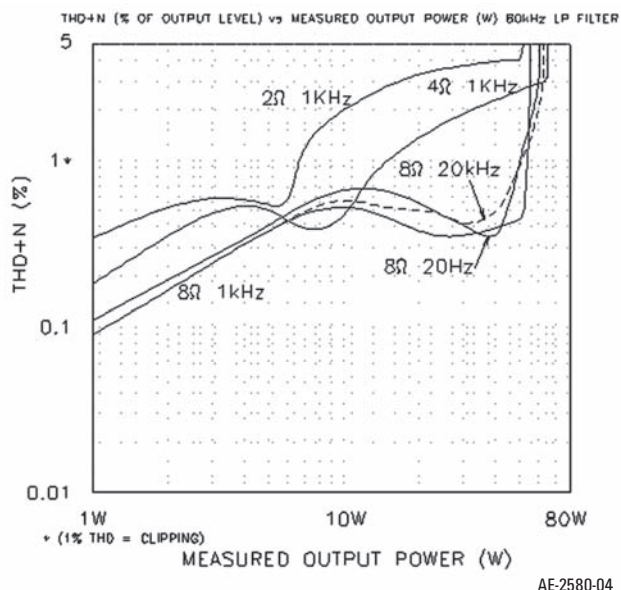


FIGURE 4: THD+N vs. output power.



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TABLE 1: MEASURED PERFORMANCE.

Parameter	Manufacturer's Rating	Measured Results
Power Output	50W RMS (sic) - 8Ω	53.6W 8Ω 1% THD+N
Power bandwidth, -3dB	DC to 300kHz	DC to 317kHz
Input Sensitivity for rated output	1V RMS, 15kΩ	936mV
Total Harmonic Distortion:	<0.05% at 5W into 8Ω at 1kHz	0.32% - 5W, 8Ω, 1kHz
IMD - CCIF (19 + 20kHz)		
MIM (9 + 10.05 + 20kHz)	N/S	0.045% CCIF
0.024% MIM		
Input Impedance	15kΩ	10k1 RCA, 1k0 XLR
Noise	N/S	0.38mV RMS
Signal to Noise Ratio	N/S	-84dB, ref 5W 8Ω
Gain	N/S	23.1dB 4Ω, 26.6dB 8Ω
Output Impedance	N/S	6.8Ω 10Hz - 20kHz

of headroom. It produced only 17.5W with a 4Ω load and 6.2W with a 2Ω load at 1% THD, but the THD curves increased gradually until the output reaches the power supply rail limits. This is more characteristic of a tube amplifier than solid-state.

There is also a slight dip part way up each THD curve as the power

increases. The negative half-cycles clipped just slightly before the positive half-cycles. The heatsink temperatures remained at 40° C and the AC line current was essentially constant, regardless of the audio power delivered, as you would expect

of a Class-A amplifier.

The distortion residual waveform for 5W into 8Ω at 1kHz is shown in Fig. 5. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after

FIGURE 5: Residual distortion.

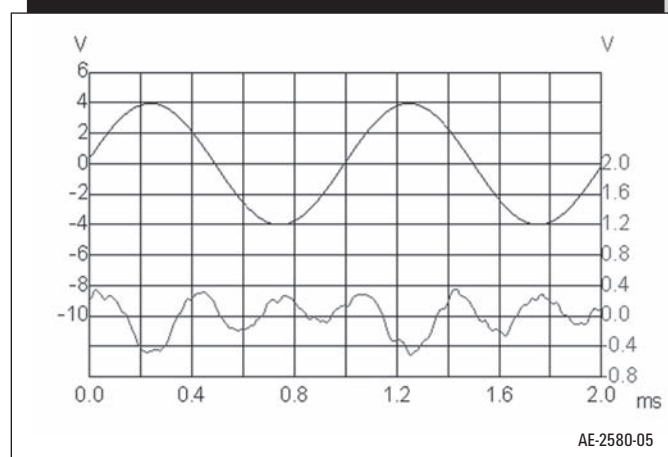


FIGURE 6: Spectrum of 50Hz sine wave.

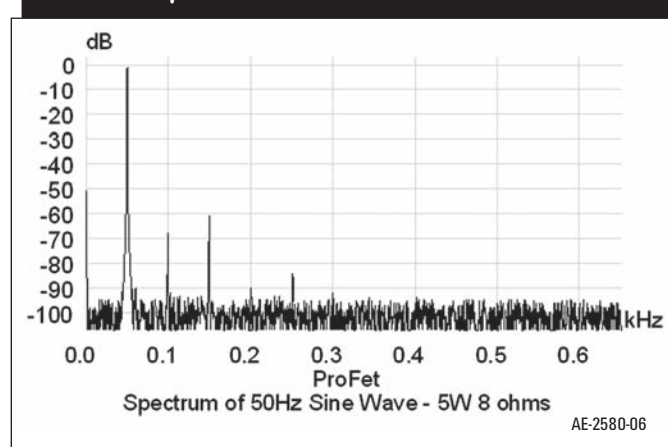


FIGURE 7: Spectrum of 1kHz sine wave.

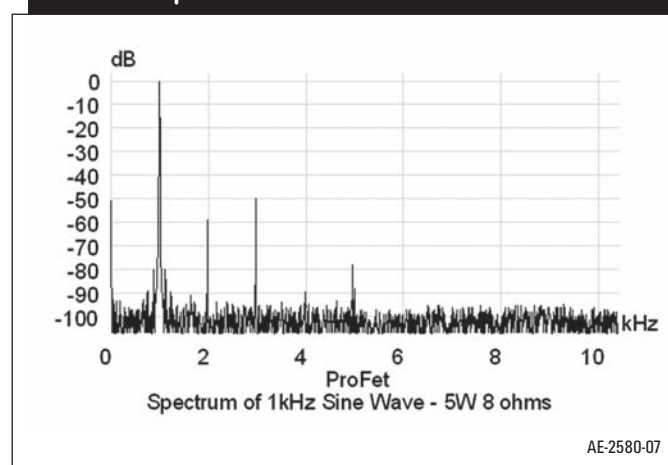


FIGURE 8: Spectrum of 19kHz + 20kHz intermodulation signal.

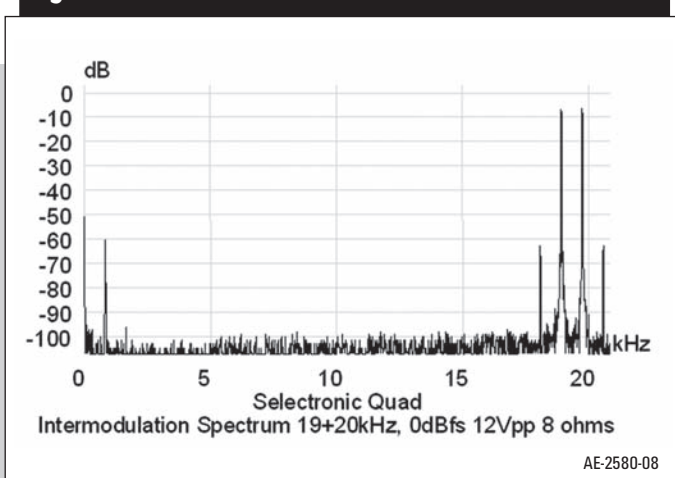


FIGURE 9: Spectrum of 9kHz + 10.05kHz + 20kHz inter-modulation signal.

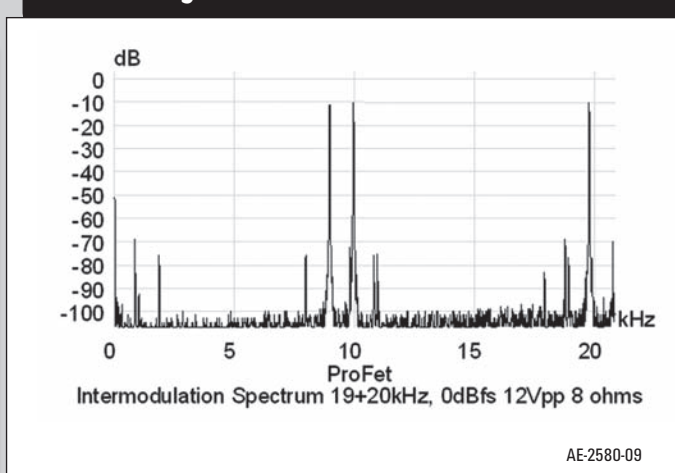
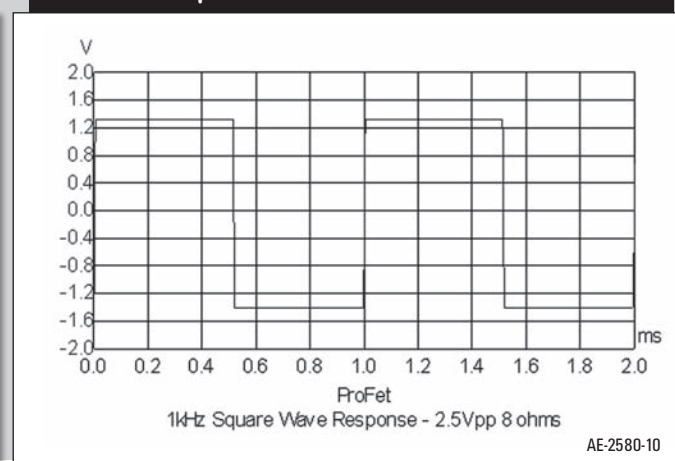


FIGURE 10: Square wave at 1kHz.



the THD test set notch filter), not to scale. This distortion residual signal shows mainly the 2nd and 3rd harmonics overlaid with very low levels of higher harmonics. THD+N at this test point is 0.38%.

The spectrum of a 50Hz sine wave at 5W into 8Ω is shown in **Fig. 6**, from zero to 650Hz. The THD+N here measures 0.41%. The 2nd, 3rd, 4th, and 5th harmonics measure -61dB, -50dB, -89dB, and -80dB, respectively. The calculated THD based on just the first five harmonics is 0.23%. A low-level (-90dB) 60Hz power supply artifact is just present above the noise floor. There is also a 70Hz spike, indicating some intermodulation of the 50Hz test signal with the 60Hz artifact.

The spectrum of a 1kHz sine wave (**Fig. 7**) had a nearly identical distribution of harmonics, with a THD+N of 0.38% and a calculated THD (based on harmonics) of 0.32%.

Figure 8 shows the amplifier output spectrum reproducing a combined 19kHz + 20kHz CCIF intermodulation distortion (IMD) signal at 12Vpp into

8Ω. The 1kHz IMD product is -62dB (0.079%), and the 18kHz and 22kHz products measure -64dB. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, shown in **Fig. 9**) resulted in a rich distribution of mostly odd-order intermodulation products between -69dB and -77dB, and calculated to be 0.024%. This MIM test gives a better indication of the amplifier's nonlinear response, because it is a closer approximation to music than a sine wave.

The output response for the 2.5Vp-p square wave at 1kHz into 8Ω was one of the best I've ever seen (**Fig. 10**). The 40Hz square wave (not shown) showed very little tilt, as you would expect from a fully DC-coupled amplifier. The leading edge of the 10kHz square wave (also not shown) was barely rounded, with no evidence of peaking or ringing. When I connected 2μF in parallel with the 8Ω load, the leading edge of the 10kHz square wave was quite rounded, reflecting the sensitivity of the ProFet to load impedance (see **Fig. 1**).

Test Note: Despite the factory recommendation not to connect or disconnect anything while the ProFet is operating, it was necessary for me to do so in order to change signal sources and vary the speaker load during my tests. The ProFet showed no ill effects from any of this switching.

A comparison of the measured results and the manufacturer's ratings is shown in **Table 1**. *aX*

REFERENCES

1. "Selectronic Triphon II Cross-over and Quad Amp," Hansen, C., *audioXpress* pp. 54-62, 64, Aug. '04.
2. "Product Review, Selectronic GRAND MOS Silver Design Amp," Hansen, C., *audioXpress* pp. 42-48, Sep. '04.
3. "The case for more testing," Dennis Colin letter, *Stereophile*, June 2004, p. 11.

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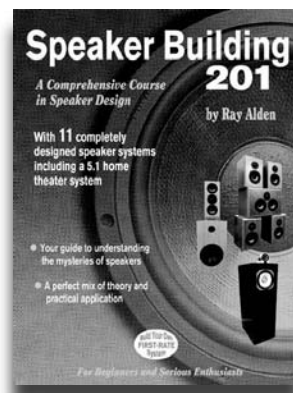
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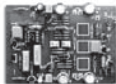


PHOTO 1: PM-2 preamp mixer/monitor.



A Purist Recording Mixer

This box is a triple threat—mike preamp/mixer/monitor—and brings you a giant step closer to experiencing the joys of recording live music.

By Dennis Colin

Named the PM-2 (for preamp/mixer/monitor), this component is designed for two recording applications:

1. Live music, with a “purist” or “minimalist” single stereo pair of mi-

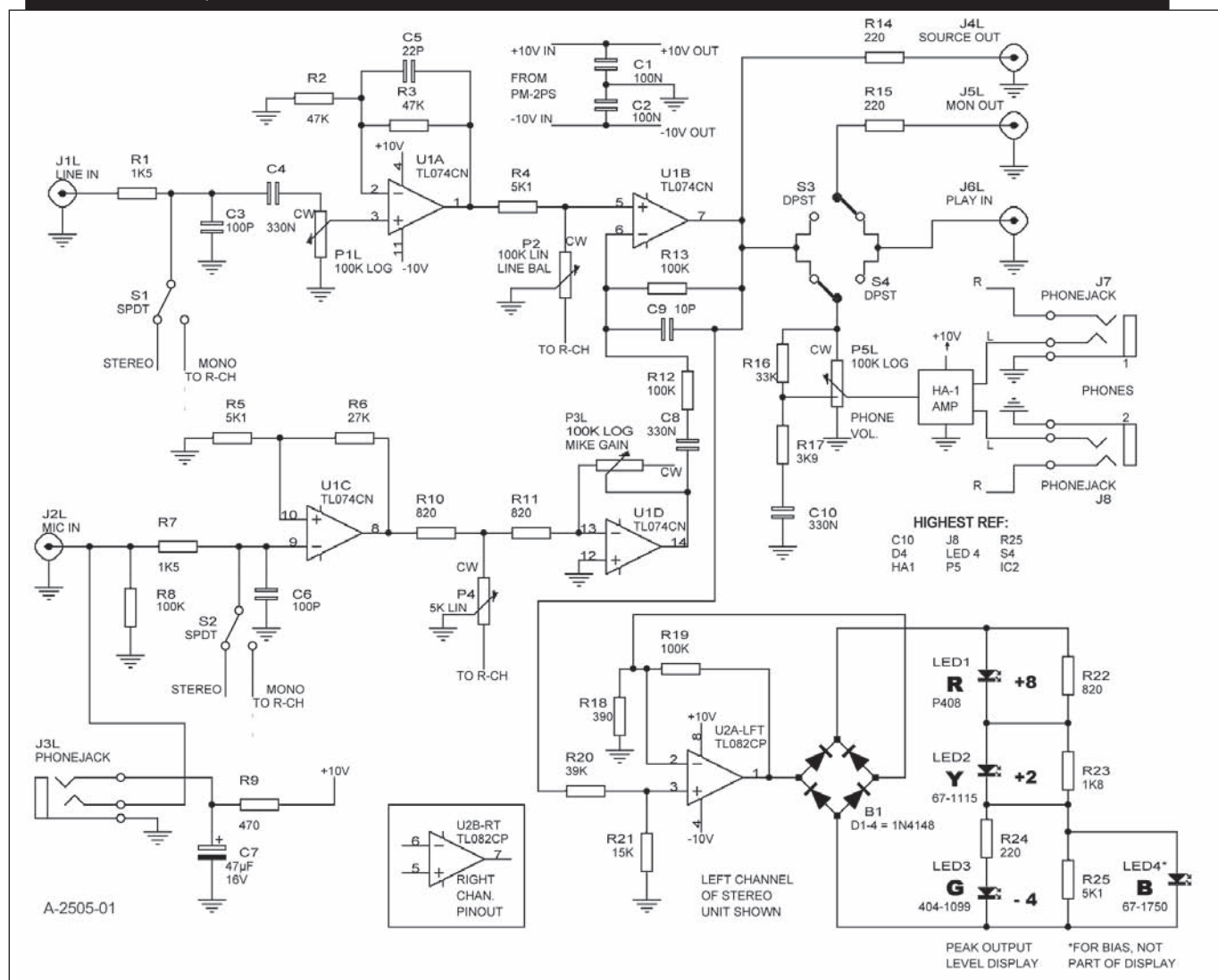
crophones.

2. Transfer of pre-recorded music, e.g., on vinyl or tape, to a CD (or other medium), with the option of real-time voice dubbing.

OVERVIEW

For live music recording, the simplest audio path (“minimalist”) includes only a stereo pair of mikes, with their preamps directly feeding the recorder. The PM-2 audio path (Photos 1 and 2) com-

FIGURE 1: Preamp/mixer/monitor. Left channel of stereo unit shown.



prises a mike pre-amp with line-level inputs summed into the output, if desired. (The line inputs could accept additional mike/mixer outputs, a band's PA feed, a direct electric bass signal, and so on.) With a purely acoustic ensemble in a good-sounding venue, an optimally-

placed stereo mike can often provide astonishing clarity, detail, stage coherence, and spaciousness.

For transferring vinyl, tape, and so on to a CD, use the line inputs, but you can mix in an adjustable amount of mike signals, e.g., for voice announcements between songs, narration, or (for those so inclined) karaoke sing-along.

FEATURES

1. Level and stereo balance controls, and stereo/mono switch, for mike and line channels.

2. Inputs for hi-z dynamic mikes, or electret mikes such as the (extremely high-quality) Mitey Mike II (*SB 4/97*, Dr. Joseph d'Appolito and Richard Campbell). For such electrets, pure DC on the "rings" of stereo jacks allows remote ("phantom") powering.

3. High-quality headphone amp (outputs for two pairs), switchable between the mixer output ("source") or an external signal (usually the recorder's playback) connected to the PM-2's jacks labeled "Play In." The phone amp can drive the high-sensitivity 300 Ω Sennheiser HD650 phones cleanly to 117dB SPL.

4. Monitor outputs for driving a stereo amp and speakers, independently switchable as above ("source" or "play").

5. Peak output level-indicating lights for -4, +2, +8dBu (0.7, 1.4, 2.8V peak).

Note: +8.24dBu = 2.00V RMS = 0dBFS = RedBook CD full scale.

If desired, you could label the level lights -12, -6, and 0dBFS. In any case, the red lights indicate instantaneous



PHOTO 2: Rear panel.

peaks of $\pm 2.83V$, which is "brick wall" digital clipping on a CD, and should never be exceeded.

PREAMP/MIXER SCHEMATIC

I used a TL074 quad op amp (U1 A, B, C, D) for the audio (**Fig. 1**); you can substitute other op amps. But note that while the TL074 output stage is usually considered to be Class AB, the maximum signal current drawn here is less than that stage's bias, so it's operating in pure Class A.

The line input is RFI filtered, DC blocked, and attenuated by P1L, R (for left, right channel), then amplified $\times 2$ by U1A. With P2 (balance) set midway, the signal is reduced by half. This is fed to the non-inverting input of U1B, where the signal "sees" a gain of two because of R12, R13. Thus, the maximum gain (P1 up full) from line in to source out is 2 (+6dB).

MIKE CIRCUIT

The preamp (U1C, D) is admittedly not state-of-the-art in noise level with a 10k Ω source (typical hi-z dynamic mike). With 24nV/ $\sqrt{\text{Hz}}$ equivalent input noise density, root-sum-squared with the 12.7nV/ $\sqrt{\text{Hz}}$ noise of a 10k Ω source resistance, this source noise is increased by 6.5dB (the noise figure of the preamp with 10k Ω source).

However, dynamic mikes are not the best in fidelity for purist recording! I designed the PM-2 primarily for use with the Panasonic WM-60AY electret mike capsule (used in Mitey Mike II). I modified the capsule (more on this later) and incorporated a remote ("phantom") powered preamp, result-

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ing in a sensitivity of 10mV/Pa and 53nV/ $\sqrt{\text{Hz}}$ noise density. The signal (and mike capsule noise) are thus amplified to a point where the PM-2 mike preamp's noise produces only a 0.8dB signal-to-noise ratio reduction (0.14dB with the Mitey Mike II because of its higher 25mV/Pa sensitivity).

Using the WM-60AY with its incorporated preamp (it should be called a "pre-preamp" to distinguish it from the PM-2 preamp), and plugged into the PM-2, the "A" weighted noise floor is 32dB SPL, and the 2% THD point is 133dB SPL (*much* too loud for safe hearing!), resulting in a dynamic range of 101dB, 5dB more than the (sonically questionable) 96dB dynamic range of a CD.

The output of U1C is fed, through the balance control, into the *inverting* input of U1D, then summed into (and polarity restored to non-inverting by) the output stage U1B. With P3L, R at maximum, and P4 (mike balance) mid-way, the maximum 2nd stage gain is 47.3. With the input stage (U1C) fixed gain at 6.3, the overall maximum mike channel gain is (measured at) 298 = 49.5dB.

This is quite a bit less gain than that of mike preamps on tape recorders, or preamps designed to capture the quietest sounds (bird songs, surveillance, and so on). But the PM-2 is optimized for music recording, where the average SPL may reach or exceed 110dB, with waveform peaks of 125dB. With maximum mike gain, the PM-2 with a Mitey Mike II will output 2V RMS (0dBFS on a CD) at 82dB SPL (Table 1). This is sufficient mike gain for any kind of live music; 82dB peak SPL corresponds to about 67-72dB average SPL, about that of quiet conversation.

INDICATOR LIGHTS

U2 (half of the dual op amp per channel) feeds an AC current, nearly proportional to the mixer output voltage, to diode bridge D1-4 (Fig. 1). Rectified, this current is fed to the series string of green, yellow, and red LEDs, each with threshold-setting resistors across them. The blue LED (4) acts as a voltage clamp for the green LED (3), helping maintain brightness balance of the three displayed LEDs (1, 2, 3). You

must use the specified LEDs (available from Digi-Key) for correct thresholds and visibility balance.

HEADPHONE AMP

Figure 1 shows the phone amp (HA-1) fed from a stereo 100k Ω pot (P5L, R) with loudness compensation (10% tap, R16, R17, C10). The latter is effective but not "boomy," compensating down to about 70dB headphone SPL for low-volume reduction in LF hearing sensitivity.

Figure 2 shows the headphone amp schematic. It's the same amplifier described in my "A High Quality Solid-State Headphone Amp" (aX Nov. '05), except biased for $\pm 10\text{V}$ supplies instead of $\pm 13\text{V}$. Q3-4 are a unity voltage-gain power stage, operating in Class A with one pair of 300 Ω phones; with two phone pairs or 32 Ω phones, operation is Class AB at high power, but is still sonically distortion-free. With con-

stant-current biasing (Q1, 2), the output stage drive current is less than the op amp's output stage bias, so U1 operates in Class A.

Frequency response of the overall amp is DC -230kHz (-3dB). The 51 Ω output resistance (from R12, 13) serves three functions:

1. Output is unconditionally stable.
2. Output is short-circuit proof.
3. Similar power is delivered to 300 Ω and 32 Ω phones (about 3dB more to 32 Ω phones, but they tend to be less power-efficient than 300 Ω phones).

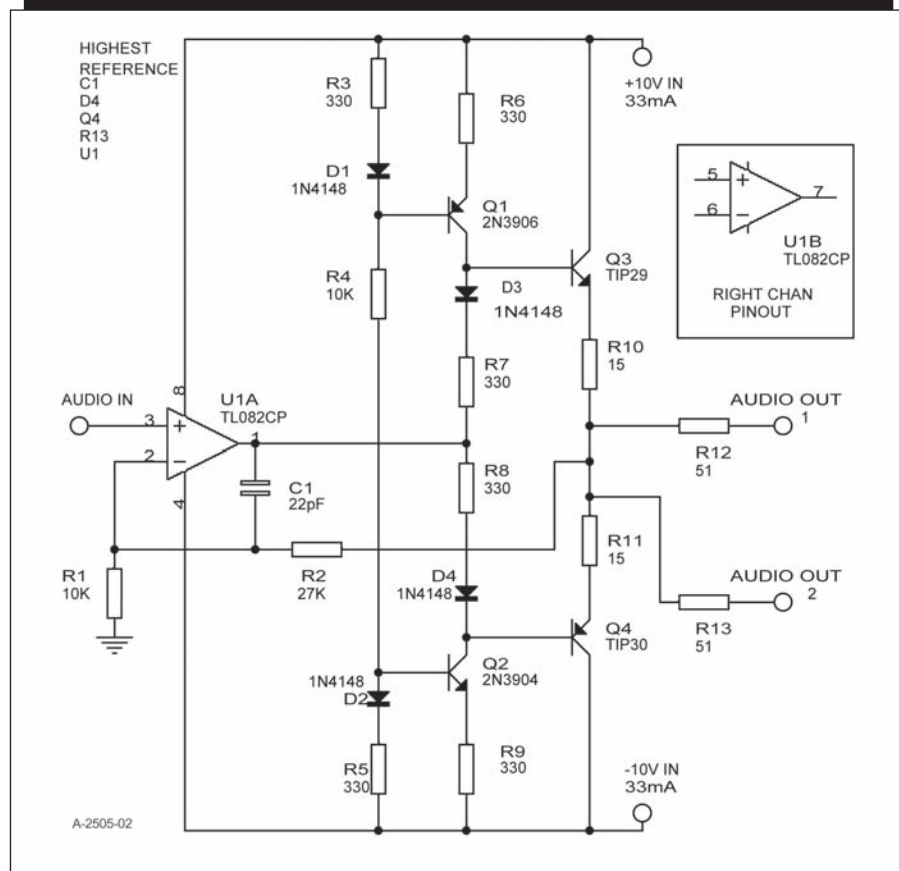
Of course, you can use the PM-2 as just a headphone amplifier (feed your

TABLE 1: Microphone Performance with PM-2.

		Mitey Mike II	Mike of Fig. 6
Maximum SPL	Mike Alone	130dB (3% THD)*	133dB (2% THD)
	With PM-2	125dB	133dB
Sensitivity		25mV/Pa*	10mV/Pa
Noise floor, "A" weighted		$\leq 36\text{dB SPL}^*$	32dB SPL
SPL for 2V RMS (CD 0dBFS)		82dB SPL	90dB SPL
PM-2 out (maximum mike gain) Same as above for 20mV		42dB SPL	50dB SPL
(-40dBFS) out Dynamic range with PM-2		$\geq 89\text{dB}$	101dB

* SB 4/97, by Dr. Joseph d'Appolito and Richard Campbell.

FIGURE 2: Headphone amp.



source player into the PM-2 line inputs) when not making recordings. And if you want to hear outside sounds (wife calling, phone ringing, elephant trying to get into swimming pool) while listening to loud headphone music, you can plug a microphone in and mix it into the music (and hear yourself singing along in private).

POWER SUPPLY

To render AC line hum inaudible, use a "wall wart" external 24V DC supply (Photo 3). But the audio circuitry (Figs. 1, 2) needs a split bipolar supply ($\pm 10V$ regulated used here). So the incoming single-ended 26V (at the 100-150mA

used) needs a "DC center tap" to serve as circuit ground. You could use a series pair of resistors, but unless a small

value (wasting a lot of current), the derived center voltage would be sensitive to positive versus negative supply load

FIGURE 3: $\pm 10V$ power supply for audio preamp/mixer/monitor PM-2. Maximum output current: 250mA if balanced; 60mA imbalance (limited by Q1, Q2 circuit). FWHK FE4823240DO30. *25.9V at 100mA, 120V RMS in Marlin P. Jones # 15371PD Jack CTR. = +. Highest Ref. C4 D5 F1 LED1 P2 Q2 R11 T1 VR2.

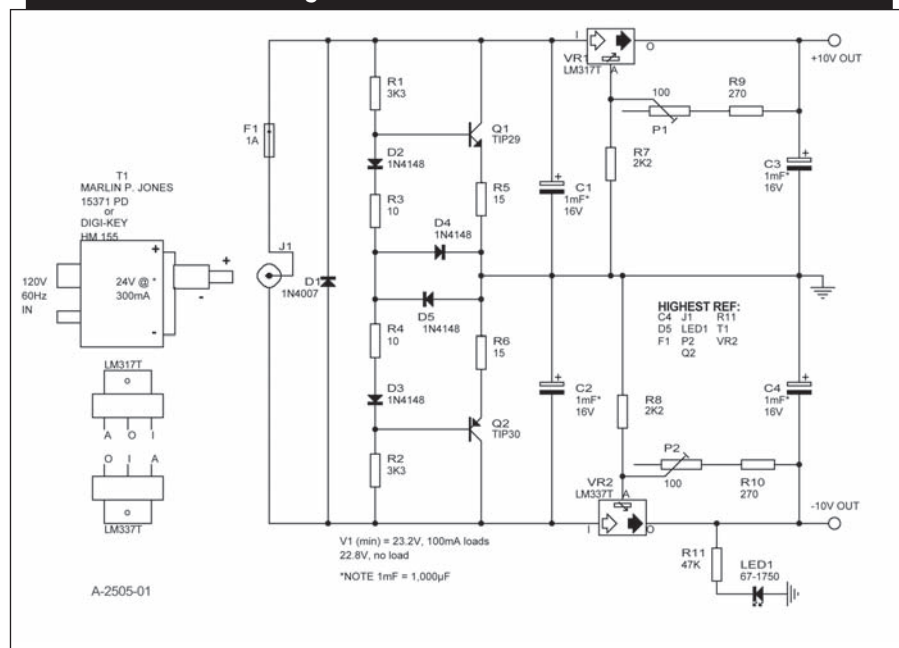
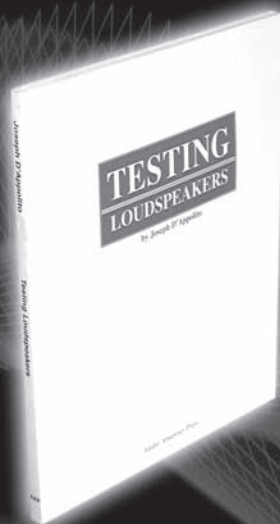


PHOTO 3: 24V DC wall supply.



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imbalance.

Q1, 2 provide a “stiff” (low impedance) center tap. Basically a Class AB amplifier output stage, the bases are biased equally above and below the single-ended input voltage’s midpoint (offset by D2, 3 and R3, 4 to compensate the base-emitter drops of Q1, 2, plus a little more to establish the Class AB bias current, stabilized by R5, 6). D4, 5 limit the voltage drive across the output transistors’ base-to-tap output points, in the event of a supply output-to-ground short, thereby limiting excessive ground current to protect Q1, 2.

A power op amp such as the LM675T can do this supply-splitting function (see National Semi’s app notes), but it’s not unity-gain stable, more components may be needed, and more bias current (than the circuit used here) may be drawn. D1 protects against reverse-polarity input DC (or an AC wall wart mistakenly used), and fuse F1 blows if

all else fails. Following the supply splitter, conventional regulators and 4000 μ F of filter caps provide very clean ± 10 V supplies to the audio circuitry.

Important note: The DC input jack (not shown on schematic; power supply jack J1 on parts list) *must be insulated from the enclosure chassis*. To do this, I mounted the jack centered on a piece of $\frac{1}{16}$ ” thick plastic sheet, about $1\frac{1}{2}$ ” square, then attached this over the inside chassis surface, centered on a $\frac{3}{4}$ ” chassis hole (**Photo 2**). I fastened the plastic with CA glue, but this broke loose after a while, so I re-glued it, then beefed it up with epoxy. I recommend mounting screws, even though I was too impatient/lazy to use them.

CONSTRUCTION

The enclosure I used is by LMB, part number UNC 5-6-6 (Digi-Key L159). It’s attractive, painted textured blue and white, but possibly not large

enough to accommodate a PC board holding all the parts. I needed to use two boards: the audio board (**Photo 4**) being solid copper with breadboard construction, mounted on the inside top of the cabinet, and a supply board (**Photo 5**)—a piece of balsa with parts glued on—mounted on the right side of the other cabinet half (the base). I used two- and three-pin connectors to connect the supply board to the DC input and the audio board, respectively.

Depending on the size required for a *real* PC board, you might need a larger cabinet. **Photos 6, 7, and 8** show my congested, impatient, Mickey Mouse wiring. Note that I didn’t use the power transistors specified—I used OHS (On Hand Stock) units, but the phone amp and supply-splitter are very insensitive to transistor parameters. The TIP29 and TIP30 parts specified are well over-rated for their jobs (**Fig. 4**).

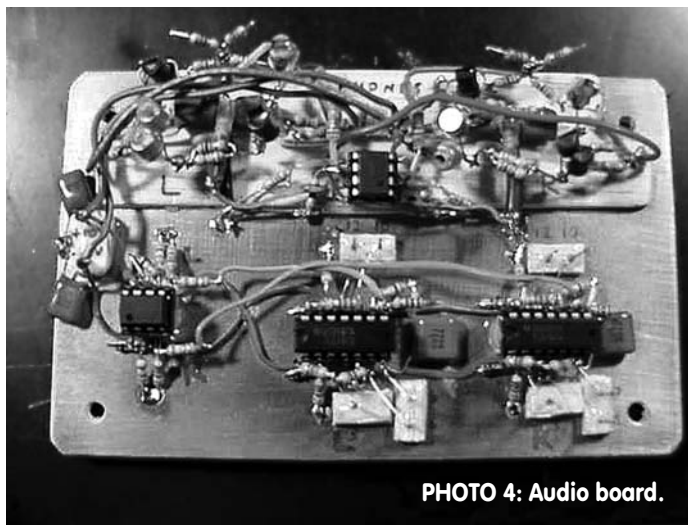


PHOTO 4: Audio board.

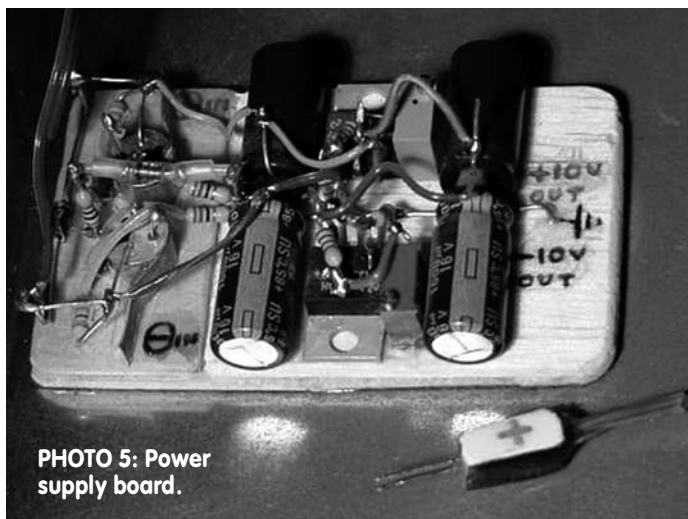


PHOTO 5: Power supply board.



PHOTO 6: Inside view.

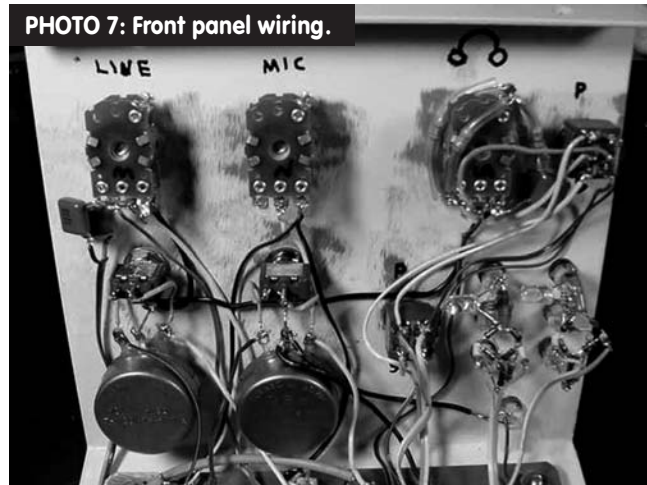
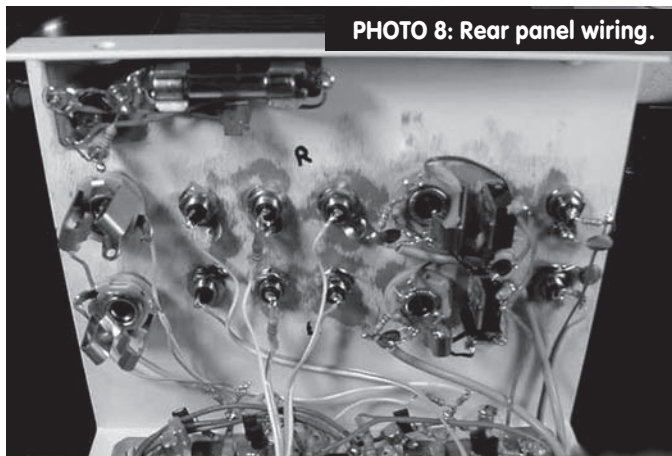


PHOTO 7: Front panel wiring.

PHOTO 8: Rear panel wiring.



SCRAPE THAT PAINT

You must grind off the thick paint (on the cabinet type suggested) to bare metal surrounding all controls and switches, and audio jack mounting holes (you must ground the pot and switch shafts to avoid hum pickup, and so on). You could mount the audio board with screws, but instead, to avoid protruding screw heads aesthetically upsetting the nice smooth cabinet top, I:

1. Located the four board corner mounting hole locations on the inside cabinet top surface.
2. Ground off the paint on a $\frac{1}{2}$ " circle around the marked locations.
3. Silver-epoxied four $\frac{1}{2}$ " #6 machine screws' flat heads to the chassis. "Circuit Works CW2400" is a room-temperature curing conductive epoxy made by Chemtronics (Kennesaw, GA 770-424-4888).

the audio board's solid copper ground plane, provide a very solid audio ground. I used, and recommend using, a layer of $\frac{1}{8}$ " thick balsa wood (or rubber sheet) between the board and chassis to prevent board flexure or possible rattling.

MICROPHONE INPUT JACKS

I used (Photo 2) $\frac{1}{4}$ " phone jacks for hi-z dynamic mikes, and $\frac{1}{8}$ " stereo jacks for electrets (+10V DC on the "rings"). I used $\frac{1}{8}$ " jacks (and plugs on my mikes) so neither mike type could be plugged into the wrong jack. But the $\frac{1}{8}$ " plugs and jacks I used are sometimes intermittent (117dB SPL transient blasts in the phones!), so I recommend $\frac{1}{4}$ " stereo phone jacks for the electret mike inputs. If you plug a dynamic mike into the electret jack, the mono plug sleeve will short the 10V DC. But no harm is done; the 470 Ω resistor (Fig. 1, R₉) will

4. After that cured, I beefed up the joints by applying two-part epoxy around them.

This worked well—the joints have held up, and the screws are well-grounded, which with nuts and lockwashers clamping

safely limit the short current to 21mA.

A HIGH DYNAMIC RANGE RECORDING MICROPHONE

The Panasonic WM-60AY (Photo 9 \$2.64 from Digi-Key, P/N P9959) is used in Mitey Mike II (SB 4/97). Its response is so flat (typical ± 1 dB, 20Hz–20kHz) that *Stereophile* uses a MMII for speaker measurements.

Figure 5 shows what's in the electret capsule, measured data, and a circuit model. The internal FET source, after a 75 Ω resistor, is originally grounded. The MMII article describes how to break this connection, so that you can add an external source-to-ground resistor. This greatly increases the maximum SPL capability.

I modified this and mounted the capsules with heat-shrink tubing (careful—

PHOTO 9: Modified WM-60AY mike capsule with connector.

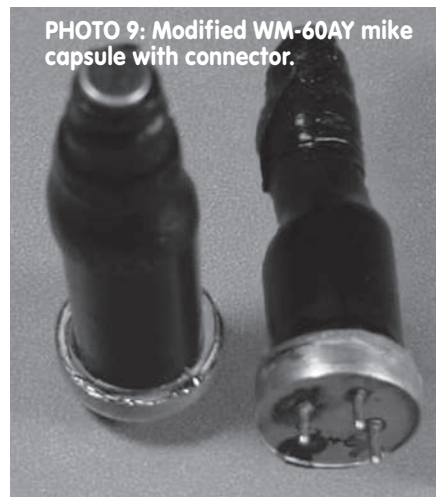


FIGURE 4: Transistor pinouts.

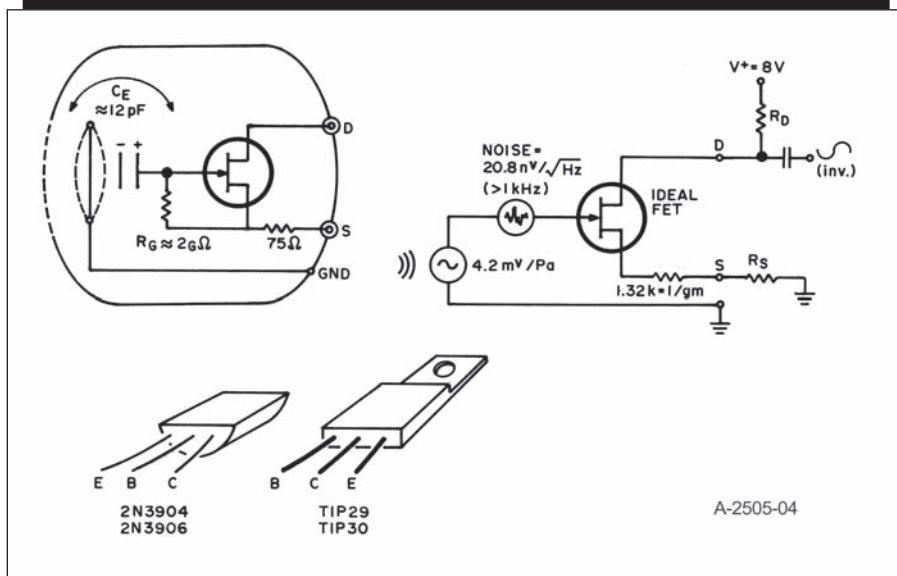
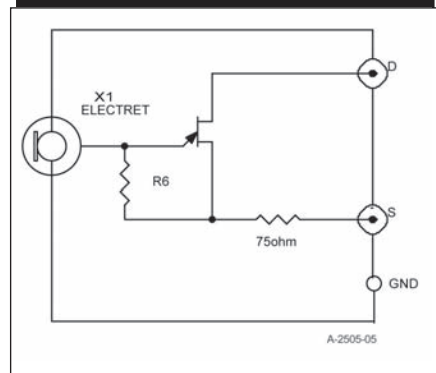


FIGURE 5: Panasonic WM-60AY electret mike. When the original connection between S and gnd. is broken, nearly constant-current ($300 \pm 50 \mu\text{A}$ over ext. R_s range of 0 – 20k Ω) establishes that R_g is returned to FET S, not gnd. Circuit Model Accuracy $\approx \pm 1$ dB for sensitivity and noise (>1kHz) calcs. ($4\text{V} \leq V_{DS} \leq 10\text{V}$)



low heat!) to a 3-pin connector, which plugs into a receptacle at the end of a copper tube attached to a $3\frac{1}{4}'' \times 2\frac{1}{8}'' \times 1\frac{5}{8}''$ Bud box (Photo 10). The "mike wand" end of the box is covered with $\frac{1}{2}''$ thick foam rubber to prevent HF re-

flections back to the mike. The bottom of the box has a standard receptacle for mike stands.

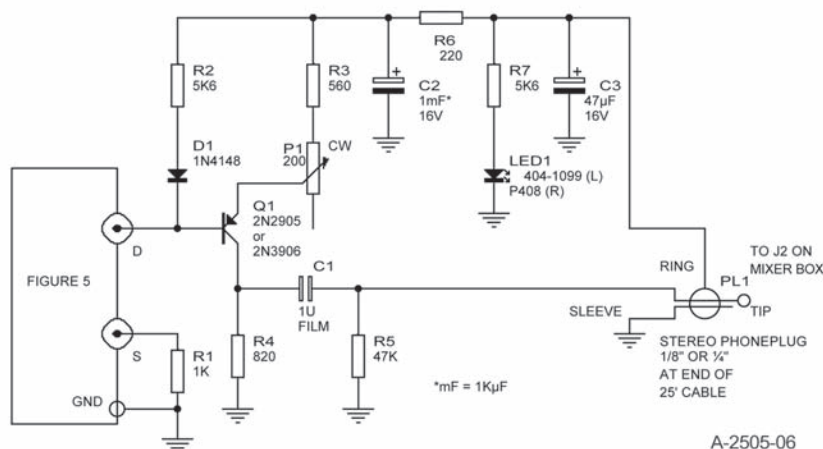
PHANTOM POWERED PREAMP

Figure 6 shows the preamp I built into the mike box (Photo 11). The circuit functions as a very linear current mirror, amplifying the mike sensitivity to 10mV/Pa. Maximum SPL (2% THD point) is 133dB. The preamp output



PHOTO 10: Mike assembly.

FIGURE 6: High-performance microphone. Low noise current-mirror preamp in $3\frac{1}{4}'' \times 2\frac{1}{8}'' \times 1\frac{5}{8}''$ Bud Box, attached to $13'' \times \frac{1}{4}''$ O.D. copper tube, "mike wand."



A-2505-06

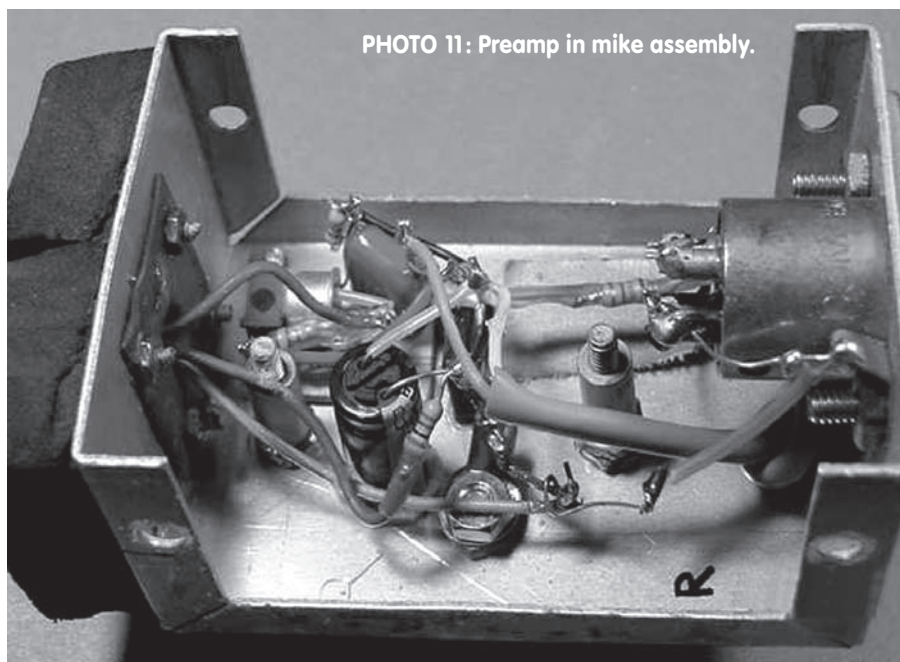


PHOTO 11: Preamp in mike assembly.



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L_e	0.39	S_d	57	SPL	85.9



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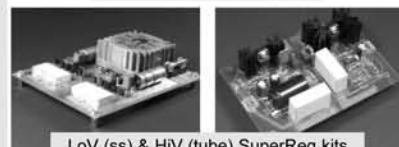
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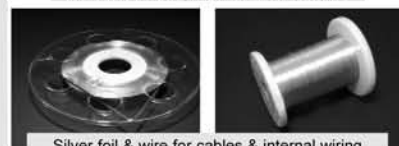
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impedance is 787 Ω , seemingly high. But it takes 2460pF loading (82' of 30pF/ft. cable) to cause 0.25dB loss at 20kHz. This output topology, though, has advantages over a low-z emitter follower or push-pull op-amp output: no amount of capacitive or other loading will produce slew-rate, crossover, or other distortion. The output stage is basically a "solid-state single-ended triode."

Unloaded, the preamp BW (-3dB) is 5Hz-1.7MHz. Driving 50' of 30pF/ft. cable, 20kHz loss is only 0.1dB. Both the preamp and (as configured) mike capsule have very high power-supply rejection ratio (PSRR), so with the regulated and R-C filtered DC fed into one conductor of the two-wire + shield cable, you hear absolutely no AC line hum or crosstalk from the other stereo channel of the PM-2 mixer.

Figure 7 shows the noise spectrum of the mike and preamp. Table 1 shows the performance of this mike, and MMII, feeding the PM-2 mixer. With the mike of Fig. 6, maximum SPL (2% THD) is 133dB, corresponding to an average music waveform SPL of about 118-123dB, louder than a large rock band up close (and *much* too loud for safe hearing!). "A" weighted dynamic range of this mike/mixer combo is 101dB.

THE INPUT SIDE OF AX

audioXpress deals almost entirely with music playback technology. And as every reader knows, even with the best playback system in existence, the sound is at the mercy of the recording, so many of which use too many (and too close) mikes, not to mention the incessant addiction to processors (EQ, compression, digital "enhancements," and so on) that engineers believe they need.

But when you capture (as opposed to "produce") your own recording of live music with a simple (but well-optimized through much experimentation) minimalist mike pair placement, and a direct path to the recorder, then *what you hear* (on neutral playback equipment) is *what was there*.

TABLE 2: PM-2 PERFORMANCE DATA.

(Note: Red Book CD Full Scale = 0dBFS = 2.0V RMS = $\pm 2.83V$ peak)
Maximum output: 5.6V RMS, $\pm 8.0V$ peak
Output impedance: 220 Ω

Line Input
Gain: 0 to 1.94 (+5.8dB)
Input z: 100k Ω
Frequency response: +0, -3dB, 4.8Hz - 123kHz; -0.14dB at 20kHz
Equivalent input noise: Density = 62nV/ $\sqrt{Hz}$, 8.8 μV RMS 20kHz BW
Noise floor = -107dBFS wideband, -109dB "A" weighted

Mike Input
Gain: 0 to 298 (+49.5dB)
Input z: 100k Ω
Frequency response: LF -3dB at 6Hz
HF: (maximum gain) -3dB at 42kHz, -0.9dB at 20kHz
(maximum gain -6dB) -3dB at 68kHz, -0.3dB at 20kHz
(maximum gain -10dB) -3dB at 88kHz, -0.18dB at 20kHz
Equivalent input noise: density = 23.6nV/ $\sqrt{Hz}$, 3.3 μV RMS
20kHz BW
Noise floor: "A" weighted, re 10mV/Pa mike sensitivity = 22dB SPL; re 25mV/Pa = 14dB SPL
Noise figure (NF) = reduction of signal/noise ratio from mike:
mike/preamp of Fig. 6 (10mV/Pa, 53nV/ $\sqrt{Hz}$): NF = 0.79dB
Mitey Mike II (25mV/Pa, $\approx 132nV/\sqrt{Hz}$): NF = 0.14dB
Maximum input to avoid clipping 1st amp stage (before attenuation) = 0.90V RMS
= 133dB SPL w/10mV/Pa, 125dB SPL w/25mV/Pa
Mike input dynamic range (with "A" weighted noise) = 111dB

Miscellaneous
L/R balance variations: $\pm 0.4dB$, attenuation range maximum to -40dB

Headphone amp
Maximum output: 4.93V RMS = 81mW, 300 Ω phones
2.03V RMS = 129mW, 32 Ω phones
117dB SPL with Sennheiser HD650 phones (300 Ω)

Output z = 50 Ω
Signal level (from PM-3 mix output) for maximum phone output = 1.56V RMS = -2.2BFS
Frequency response: +0, -3dB, DC -230kHz
Loudness compensation: maximum at phone volume attenuator = -20dB, constant below -20dB; +3dB at 100Hz, +7dB at 36Hz. Within $\pm 2dB$ of LF hearing reduction from $\geq 100dB$ SPL to 70dB SPL, down to 20Hz.

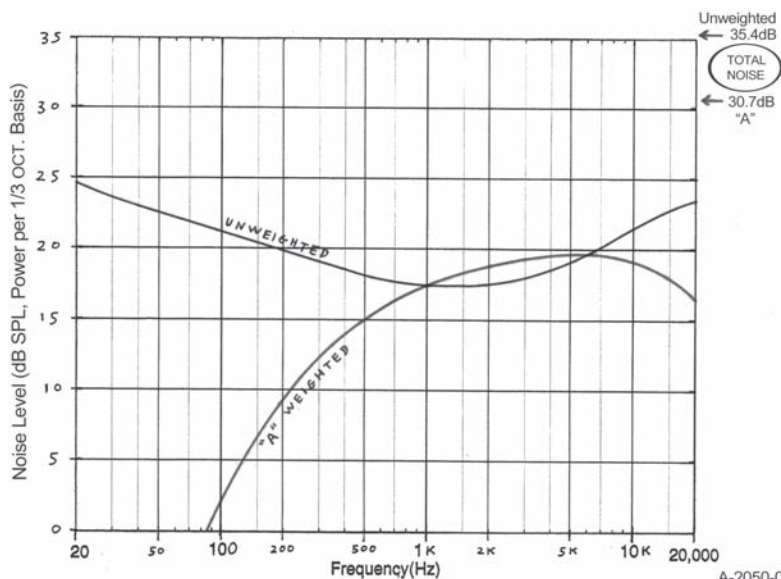
HIGH-PERFORMANCE MICROPHONE

Sensitivity = 10mV/Pa
Frequency Response = $\pm 1dB$, 10Hz-20kHz
Maximum SPL = 133dB at $\leq 2\%$ THD
Pattern = OMNI up to 20kHz
Noise: 52.9nV/ $\sqrt{Hz}$, 1-20kHz
7.29 μV , 1-20kHz
11.7 μV , 20Hz-20kHz
Noise Floor: 35.4dB SPL, Flat
30.7dB SPL, "A" weighted
Dynamic Range, "A" noise to 2% THD = 102dB

With RD = 5k Ω , V⁺ = 8V

R5 Ω	Sens mV/Pa	SPL, dB 2% THD	Noise SPL, dB "A"
0	12.7	125	30.4
300	10.4	127	
500	9.6	130	
700	8.75	131.5	
1k	7.76	133	30.6
1.5k	6.46	135	
2k	5.51	137	
3k	4.30	140	
5k	2.94	144	32.0

FIGURE 7: Noise level, WM-60AY mike and PM-2 MP preamp, 1/3 octave analysis.



Then, if the music and its venue acoustics were very good, you'll have a recording that provides:

1. An accurate, undoctored reference to best evaluate your playback system's fidelity (you might learn more than you wanted to know, or you might be pleasantly surprised).

2. The supreme enjoyment of extremely natural, clear, and three-dimensionally spacious music reproduction. Once you start recording live music, you'll never want to stop. And, hanging around musicians could make you become one—but don't quit your "day job" just yet! *aX*

PM-2 Parts List

Total Qty.	Schematic Ref.			Description	Source
	Mixer	Phone Amp	Supply		
2	C9			Cap, 10pF	
4	C5	C1		Cap, 22pF	
4	C3,6			Cap, 100pF	
2	C1,2			Cap, 0.1µF	
6	C4,8,0			Cap, 0.33µF, 100V poly	Digi-Key EF 1334
2	C7			Cap, 47µF 16V	
2			C1,2	Cap, 1mF, 25V	
2			C3,4	Cap, 1mF, 16V	
20	D1,2,3,4	D1,2,3,4	D2,3,4,5	Diode, 1N4148	
1			D1	Diode, 1N4007	
1			F1	Fuse, 1A, FB	
8	J1,4,5,6			Jack, RCA	
2	J2			Jack, 1/4" phone, mono or stereo	
6	J3,7,8			Jack, 1/4" phone, stereo	Digi-Key SC1094
1			J1	Jack for T1 (DC in)	Digi-Key SC1048
2	LED1			LED, red	Digi-Key P408
2	LED2			LED, yellow	Digi-Key 67-1115
2	LED3			LED, green	Digi-Key 404-1099
3	LED4		LED1	LED, blue	Digi-Key 67-1750
1	P4			Pot, 5klin	Digi-Key RV4N502C
1	P2			Pot, 10klin	Digi-Key RV4N103C
3	P1,3,5			Pot, 100k	Alps 100k Ax2 aud, stereo Radio Shack 271-1732
2			P1,2	Trimpot, 100Ω	
2		Q2		NPN, 2N3904	Digi-Key 2N3904RLRAOSCT
2		Q1		PNP, 2N3906	Digi-Key 2N3906RLRAOSCT
3		Q3	Q1	NPN, TIP29	Digi-Key TIP29BOS
3		Q4	Q2	PNP, TIP30	Digi-Key TIP30COS
2			R3,4	RES, 5% 1/4W 10Ω	
6		R10,11	R5, 6	RES, 5% 1/4W 15Ω	
4		R12,13		RES, 5% 1/4W 51Ω	
6	R14,15,24			RES, 5% 1/4W 220Ω	
2			R9,10	RES, 5% 1/4W 270Ω	
12			R3,5,6,7,8,9	RES, 5% 1/4W 330Ω	
2	R18			RES, 5% 1/4W 390Ω	
2	R9			RES, 5% 1/4W 470Ω	
6	R10,11,22			RES, 5% 1/4W 820Ω	
4	R1,7			RES, 5% 1/4W 1.5k	
2	R23			RES, 5% 1/4W 1.8k	
2			R7,8	RES, 5% 1/4W 2.2k	
2			R1,2	RES, 5% 1/4W 3.3k	
2	R17			RES, 5% 1/4W 3.9k	
6	R4,5,25			RES, 5% 1/4W 5.1k	
4			R1,R4	RES, 5% 1/4W 10k	
2	R21			RES, 5% 1/4W 15k	
4	R6		R2	RES, 5% 1/4W 27k	
2	R16			RES, 5% 1/4W 33k	
2	R20			RES, 5% 1/4W 39k	
5	R2,3		R11	RES, 5% 1/4W 47k	
8	R8,12,13,19			RES, 5% 1/4W 100k	
2	S1,2			Switch, SPDT	
2	S3,4			Switch, DPDT	Digi-Key 360-1156
1			T1	"Wall Wart" DC supply	Marlin P. Jones 15371 PD or Digi-Key HM155
2				24V 300mA	
2	U1			TL074CN	
3	U2		U1	TL082CP	
1			VR1	Pos. Reg, LM317T	
1			VR2	Neg. Reg, LM337T	
1				Enclosure	See text



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AMP UPDATE

With reference to John Stewart's article "Crowhurst's Twin-Coupled Amp" published in the Sept. '05 issue, congratulations for the well-written topic full of details and tables. However, unless I'm missing something, it lacks data on the bandwidth figures. In addition, I would like to know whether, by using higher quality output transformers, the performance improves—and to what extent.

*Atto E. Rinaldo
Italy*

John Stewart responds:

I should have included some information on the bandwidth of the amplifier, so here it is.

Frequency response when using internal balanced negative feedback, as per Crowhurst:

At 10W	1dB down	35Hz and 33kHz
	3dB down	27Hz and 47kHz
At 1W	1dB down	9.6Hz and 36kHz
	3dB down	7.4Hz and 48kHz

But there is breakup in the signal at the low-frequency end. At 1W that happens at 15Hz, while at 10W breakup is at 40Hz. More iron would help a lot, as you would expect.

Here is more information to show how the circuit responds to a full loop NFB connection, just as you would see in an ordinary amp. Refer to Figure 8, the Alternative Front End in the article. Set the input for differential operation. From the amplifier 8 Ω output a 10k resistor is connected to the grid end of R4, a 1k resistor at the amplifier front end. This includes the output transformer and results in about 12dB of NFB at 1kHz. The damping factor is increased considerably. I measured it to be 17 at 1kHz.

Frequency response when using alternative negative feedback

At 10W	1dB down	35Hz and 40kHz
At 1W	1dB down	10Hz and 60kHz

The low-frequency limit is set again by the iron in the output transformers.

However, they do seem to work well beyond the Hammond spec of 150Hz to 15kHz. I've not yet tried better-quality output transformers, but I intend to do that as soon as I can find time.

In all of the measurements, I found about a 0.5dB rise at something like 15kHz.

Thank you for your comments and interest in this project.

TRANSMISSION LINES

Rick Schultz is to be applauded for continuing his "TLs for Dummies" articles—exploring TL-space using George Augspurger's transmission line modeling software and distilling it so you can quickly come up with a reasonable—if not optimum—TL design for any given driver ("Alpha Transmission Lines," Aug. '03 *aX* and "Omega Quarterwave Reflex," Oct. '05 *aX*). I found the alphaTL very useful, and my brother is really enjoying the alphaTLs I built to replace the ones he has been using for the last 30 years (we didn't call them alphaTLs back then).

But Rick does a disservice to the community—and those whom he acknowledges (I am somewhat disappointed that I got no acknowledgment for executing the drawing of the omega QWR published in *aX*)—with his claim that his omega quarter-wave reflex is a unique design and subject to full intellectual property protection. As I pointed out to Rick when I reviewed his

work prior to publication, this TL is no different than what has, over the past four years, become commonly known as a Mass Loaded Transmission Line—ML-TL for short.

George Augspurger, in his presentation at the 1999 AES conference, suggested a straight TL with an offset driver and restricted terminus, i.e., an ML-TL. The popular Zigmahornet (an ML-TL) predates King's and Augspurger's work. Greg Monfret (a.k.a. GM) brought us the first highly visible ML-TL (and I believe coined the term to differentiate it from Martin King's TL-TQWT terminology) with his RS40-1354 ML-TL at Atlanta DIY 2001.

Martin King has published a number of incredibly well-documented ML-TL projects (from sims through builds with measurements) dating back to shortly after this, and the many, many (you could probably say hundreds) ML-TL configurations for specific drivers that GM has suggested (GM says he built his first one in 1969). I would guess that literally thousands of ML-TLs have been built by DIYers in the last few years. A number of representative designs can be seen at <http://www.planet10-hifi.com/boxes.html>. A number of the projects in Martin King's Transmission Line gallery are ML-TLs (<http://www.quarter-wave.com/Gallery/Gallery.html>), and Bob Brine's FT-1600s is one of the most widely implemented and successful ML-TLs (http://geocities.com/rbrines1/Pages/FT-1600_MkII/Main.html). The last two sites plus mine (<http://www.t-linespeakers.org/>) contain much information on successfully implementing quarter-wave designs.

Rick Schultz was a regular visitor to online forums over the evolution, discussion, and sharing of these designs,

so is well aware of them. His methodology to design them is certainly unique, but the design is not.

I would also suggest that any DIYer or commercial designer wishing to build an ML-TL could—using Augspurger's TLwrx software or Martin King's even more versatile MathCAD worksheets—design an ML-TL more fully optimized for a specific driver. Thanks to the generous efforts of many DIYers, including those already mentioned, many designs for many drivers are already freely available.

David Dlugos
planet10@mac.com

Rick Schultz responds:

David Dlugos certainly deserves credit for Fig. 14, along with my sincere apology. His credit improperly dropped during my edits of the manuscript.

I expected Xpress Mail would receive questions concerning adjustments and TSP-derived quarterwave design. I am gratified by the apparent acceptance of the essential portions of "Omega Quarterwave Reflex."

Mr. Dlugos' letter can be distilled into two good questions.

1. Is there naming precedent for the design I described?
2. What, if anything, is protected?

Mr. Dlugos understands the history and development of quarterwave design better than anyone I know. Mr. King's ML-TQWT naming credit is almost certainly accurate. I have every confidence in Mr. Dlugos' report of first use of "mass-loaded Transmission Line" (ML-TL) by Mr. Monfret® 2001.

Mr. Dlugos reports it has taken four+ years of evolution to develop a generally accepted ML-TL. I fondly recall forum discussions with Martin, GM, Dave, Thorsten, Godzilla, ron, hitsville, James, and others. The difference between ML-TL evolution versus growth from "Rhinos" through "alpha Transmission Lines" to "Omega Quarterwave Reflex" is the difference between iterative modeling versus cookbook design.

The punctuated evolution of ML-TL design began shortly after "Rhinos" was published. "Rhinos" shows how an unstuffed aperture-loaded parallel cabinet (predating ML-TL) and an unstuffed, aperture-loaded reverse-tapered cabinet (a -tqwt) work.

"Rhinos" expressly discusses the quarter-wave and reflex actions of the pipes. At the

time, "I was astonished when I first saw the frequency response. [fQ shifts] about one-half octave below [pipe length resonance]!" That was news in 1999.

"Quarterwave reflex" is not used in the 1999 article. I had no concern with naming precedent then. The term ML-TL has come into general use since. I offer the new term to separate iterative modeling from my cookbook approach. Time will tell whether ML-TL and Quarterwave Reflex can stand side by side.

Mr. Dlugos says, "His methodology is certainly unique." This is a high compliment, especially considering the source. This also answers the second question above.

King and Augspurger are protected if a software program or loudspeaker design is derived using their computational models. Likewise, producing and selling software or loudspeakers derived from "Let's build one" and related graphs are protected.

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and therefore protected.

Because OQR contains the first discussion and explanation of the distinctive pulse-shaped impedance curve, I believe protection extends to that curve in the context of quarterwave loudspeakers.

David Dlugos responds:

To use Rick's term, the pulse-shaped impedance curve is simply the characteristic impedance curve of a box transitioning from a standard vented box to an aperiodic box. The bottom hump of the saddle-shaped impedance curve becomes smaller as more damping is added—to the port in the case of a BR or the whole line in the case of a TL (aka quarter-wave resonator). This was discussed and explained by Augspurger in his AES paper (see figure 7a) at AES 107, and I was also able to dig up a paper published in 1956 by Ted Jordan in *Wireless World* that shows and discusses essentially the same thing.

Regarding Rick Schultz's "Omega Quarterwave Reflex" article, I was pleased and excited to see a TL speaker design article in aX, having built sever-

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al TLs and modeled more potential TL designs than I can recall. I read it and its addendum on the aX website with great interest. The ΩQR design process is easy to follow and will likely allow many people to build a satisfactory design. Still, I have issues with many of the suppositions, conclusions, and claims presented in the article.

First, consider what the ΩQR is and is not. In my opinion, it's essentially a specific case of an ML-TL whose ultimate performance has been limited by restricting the use of three important design parameters: driver location, stuffing length along the line, and relative aperture size. There is nothing really unique about the ΩQR's design in terms of the resulting performance. And, the process does not necessarily—and, I believe, may infrequently—result in an optimally performing design as claimed. Its only real benefit is its ease of application; no computers are necessary.

The author placed much emphasis on the resulting impedance characteristic, comparing it to an over-stuffed TL and an un-stuffed TL and why it indicated superior performance to both. He went so far as to describe an EKG sinus rhythm waveform as being similar in shape to the impedance curve shape of the ΩQR. Why this EKG-waveform similarity is useful or important is not clear. From my viewpoint, it seems this analogy was used not for technical or scientific purposes, but simply to impress the reader.

As the author stated, all TL-type designs will have a pair of impedance peaks before stuffing is added. When stuffing is added, the magnitudes of both peaks will be reduced, with the peak occurring at the lower frequency decreasing more in magnitude than the other peak. If enough stuffing is added, the lower peak can be made to completely disappear, making the TL perform much like a sealed enclosure. Nevertheless, there is nothing special about the shape of the ΩQR's impedance curve; it shows only that the performance characteristics will lie somewhere between a typical vented system and a sealed system. Its impedance curve shape does not necessarily indicate superior or optimum perfor-

mance, just a different performance.

Locating the driver along the line according to the "20%" rule of thumb is generally a good rule *when you can't model the expected results*, but my extensive modeling with a variety of drivers and a variety of TL designs clearly shows a different location is often needed. Sometimes 25% is better, sometimes 15% is better, and sometimes somewhere in between or even outside of those values gives superior results. You just can't limit yourself to using 20%. With Internet access to readily available and easy-to-use modeling programs, such as those on Martin King's website, there's no reason to restrict a design's potential performance (or to just hope the performance will be good).

Restricting the stuffing length to the first 67% of the line's length will not ensure optimum performance. That may be too long or too short. Many of my modeled designs have worked better by stuffing just the first 50% or so of the line. Additionally, it's sometimes necessary to modify both the length and density of the stuffing to achieve optimum results. Again, it's not difficult to acquire and use modeling software, so why limit your design?

The author made a comment on the shape of the aperture's SPL response on either side of its peak stating, in so many words, that because the response sloped up or down at 12dB/octave on both sides, this defined the *system* as being 2nd order! An aperture response is only part of the system response. All TLs, no matter their configuration, are 4th-order systems. That's not to say the *system* response in the first octave or two below f₃ can't or won't be more gradual, even 2nd order, but the SPL system responses of all TLs will become 4th order as frequency approaches zero.

Making the aperture's area fairly small with respect to the line's area is not the only way, nor is it necessarily the best way to make the system response be flatter and deeper. You can make the line longer or taper it, both of which will deepen the response because the line's ¼-wavelength resonant frequency will be lowered, while tapering will also flatten the response. Mak-

ing the enclosed volume larger will also deepen the response.

While reading the author's analogy between TLs and the movement of the radio antenna on his car as he drove along a highway, I found it contrived yet vaguely familiar. In a quick check on Martin King's website I re-read his paper describing the enclosure design of an ML-TL for a Lowther driver.

In that paper Martin visualized using a yardstick with one end clamped to a table to illustrate what happens when the free end of the yardstick is plucked. The yardstick's vibration was analogous to the air velocity in a TL. Further, Martin discussed how the yardstick's vibrational patterns would change if its length was increased or decreased, or if a weight was applied to the loose end. The similarity between Mr. Schultz's antenna analogy and Martin's yardstick analogy is curious, to say the least, *and Martin's was written in 2003*.

Never one to not at least try yet another documented TL design process, I designed and modeled an ΩQR for my favorite 8" woofer, a Peerless 831709, following the author's design process. I then compared the modeled performance to the modeled performance for my current TLs using the same woofer. I used the same Mathcad-based programs from Martin King for both models.

The T/S parameters I used are the average values for the pair of Peerless drivers I'm using, which were broken in and measured for me by the good folks at Madisound. The parameter values are:

fs = 21.8Hz	Qms = 2.36
Qts = 0.334	Re = 5.52Ω
Vas = 8139 in ³	Le = 0.56mH
Sd = 33 in ²	Bl = 7.04
Qes = 0.389	

For the ΩQR cabinet, stuffed with AcoustaStuf, the calculations resulted in a line length of 100", Sx of 106.4 in², Sa of 10.64 in², a stuffing density of 0.56 lb/ft³ for the first 67" along the line, and the driver's center located at 20" from the closed end of the line. For convenience I modeled the cabinet with an Sx of 105 in² and an Sa of 10.5 in², making the cabinet's internal dimen-

sions 10.5" W × 10" D × 100" H.

Three graphs of the modeled ΩQR performance are shown in **Fig. 1**. (Figures are located at



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Once there, click on the current issue and follow the link title "articles and addenda.") The red line in **Fig. 1A** is the system's SPL output (ignore the dashed blue line). The red line in **Fig. 1B** is the woofer's SPL output, and the dashed blue line is the aperture's SPL output. Finally, the red line in **Fig. 1C** shows the system's impedance curve (again, ignore the dashed blue line).

Now consider how the performance of the same woofer in my TL-based 3-way speaker compares. For convenience, just call my system the SigmaTL, which is a three-way system with the Peerless 831709 working into a 12:1 tapered TL. At the closed end of the line the cross-sectional area is 70 in², and it tapers to 6 in² at the open end of the line. The line is 69" long and stuffed with AcoustaStuf for the first 36" from the closed end at a density of 1 lb./ft³. The driver is located 16% of the line's length from the closed end. **Figure 2** contains the three performance graphs that resulted from the modeling process.

Now, consider the comparable graphs, one pair at a time. For the ΩQR you can see from **Fig. 1A** that f₃ is about 27Hz, whereas for the SigmaTL (**Fig. 2A**), f₃ is about 37Hz. Clearly, the ΩQR has a performance advantage over the SigmaTL in this response area.

Looking higher in frequency, above 300Hz both response curves are pretty much identical with very small-amplitude "ripples," but from the knees in their response curves up to 300Hz, the response of the SigmaTL is much flatter than that of the ΩQR. Staying with **Figs. 1A** and **2A** and looking at the slopes of the curves below their knees, the ΩQR rolls off at a faster rate than the SigmaTL. Neither the ΩQR nor the SigmaTL can be called or is a 2nd-order system, however.

Comparing **Figs. 1B** and **2B** shows the beneficial effects of the small aperture size in the ΩQR in reducing harmonic ripples in the system response shown in **Fig. 1A**. Still, both system responses

are quite good, and almost identical, above 300Hz or so.

The graph of system impedance in **Fig. 1C** shows the "sinus" waveform discussed by the author for the ΩQR. In **Fig. 2C** it's clear the lower-frequency impedance peak has been almost completely suppressed as compared to **Fig. 1C**, but this only means that the rolloff rate of the SigmaTL's response below its knee will be slower.

Which system is better? Well, if your only concern is how low f₃ is, clearly the ΩQR is better, but think about cabinet size for a moment. The ΩQR's cabinet would enclose 6ft³ of volume and be over 8' tall! Even if the line were folded once, the cabinet's dimensions would end up around 1' wide by almost 2' deep and would be over 4' tall. That's a really big box!

How big is the actual cabinet for my SigmaTL? I built this as a two-box system, so the woofer has its own dedicated cabinet for the TL. Its volume is only ~1.5ft³ with external dimensions of 11.5" W × 27" H × 14.75" D (the SigmaTL box does not have a rectangular shape and has internal baffles to fold the line several times, so using these dimensions to estimate its internal volume would not be appropriate).

Except for sound effects, such as the tromping of dinosaur feet through the jungle, or pipe organs perhaps, how much of the music we listen to requires an *anechoic* f₃ much below 40Hz? I can't speak for anyone else, but my musical tastes certainly don't. And don't forget, room gain effects usually add in quite a bit of boost in this region.

If I wanted a much lower system f₃ and could live with a somewhat larger cabinet, I would stick with a tapered TL. For instance, I'd increase the SigmaTL's cabinet volume by about two-thirds to ~2.5ft³, while keeping the line length, line taper, and stuffing distance/density the same.

Call this one the DeltaTL. Its modeled system SPL response is shown in **Fig. 3**. Not only is the f₃ of 30Hz comparable to the 27Hz f₃ of the ΩQR response shown in **Fig. 1A**, the DeltaTL's overall system SPL response is much smoother and flatter. And, the DeltaTL's cabinet volume and size would still be much smaller than those of the ΩQR.

While I believe the performance abilities and uniqueness of the ΩQR were overstated, I was bothered even more by the self-important tone of the article and the author's failure, in my opinion, to fully acknowledge the work of others before him. Anyone reading the article who does not generally know much about transmission line speaker systems or their established designers might easily conclude the author did much more original thinking and work than I believe he actually did. His acknowledgments of the extensive contributions from George Augspurger and Martin King in this field, from which the ΩQR design was "derived," were grossly inadequate.

In all fairness, I must give Mr. Schultz credit for developing the mathematical expressions and graphs that define the ΩQR design and defend his proprietary rights to them, but I can't imagine he honestly believes any of this has enough originality to be patented.

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Rick Schultz responds:

It pleases me to see reference to Mr. King. He is a pioneer in quarter-wave design. His MathCAD-based modeling tool is the best freeware for pipe design.

"Omega Quarterwave Reflex" is written as a follow-up to "alpha Transmission Lines" (*aX* 9/2003) and "Rhinos" (*Speaker Builder* 5/99). Please consider them together. I acknowledge Mr. King and Mr. Dlugos in "alpha Transmission Lines" and do not repeat the acknowledgment in the current article. I consider them both cyber-friends. See more at <http://www.quarter-wave.com/> and <http://www.t-linespeakers.org/>, respectively. Sorry, I don't have a website.

Mr. King and Mr. Augspurger are engineers by trade, and this brings great value to quarterwave design. Mr. King's articles and discussions rely heavily on calculus. Mr. King says on his site, "If a new user invests a few hours working with one of the simple transmission line worksheets, I believe they become proficient MathCAD users very quickly. Unfortunately a few users are never comfortable with MathCAD and abandon the worksheets."

A user needs nine TSP or similar engineering parameters, not all of which are

readily available. I re-examined his "Classic Transmission Line Alignment Tables," which help the beginner determine length, cross-section using several tapers, and driver placement based on $\frac{1}{3}$ length. The user changes them and speculates over aperture size until a good pipe is revealed. Every iteration yields several new graphs.

The writer modeled a nice design for his speaker. I have no idea how much iteration it took to come up with his design.

My OQR "cookbook" model requires three TSP and three readings from one simple graph. With a little algebra, OQR quickly delivers a complete aperture loaded pipe that can be constructed with a good performance. OQR reduces guesswork and predicts cabinet dimensions.

The King and Augspurger models produce graphs. With less input, my model delivers recommended pipe characteristics. Because King's and Augspurger's goals and outcomes are different from mine, comparing them as the writer does is imprudent. Apples and oranges.

A pipe builder could start with OQR to get a good pipe and then move on to use Mr. King's or Mr. Augspurger's modeling software. Speculate on tweaks and adjustments until the definitive, ultimate pipe is designed.

Naming or categorizing quarterwave pipes is the source of much confusion. Before I proceed to address some of the writer's other issues, I need to explain the state of the art as I perceive it. As you can probably tell, I like to organize things into a

+-----+-----+		
	parallel sides	tapered
+-----+-----+		
open	Transmission Line	TQWT
+-----+-----+		
aperture loaded	Quarterwave Reflex	ML-TL, -tqwt
+-----+-----+		

useful system.

I believe quarterwave pipes should be categorized by two characteristics: sides and loading.

Driver placement is important but not crucial.

My two articles discuss Transmission Lines and Quarterwave Reflex. For the rest of the readers, ML-TL means "mass-loaded Transmission Line." A TQWT is a "tapered quarterwave tube." It is narrow at the closed speaker end and wide open at the other end.

I started using the term "-tqwt" on fullrangedriver.com several years back. It means the taper of the TQWT is reversed. The closed end is wide while the open end is small.

I have clearly and repeatedly said TL and QR have parallel sides and are not tapered. My comments and assertions pertain to a parallel-sided quarterwave design, whether a TL or QR. I stand by what I have written.

The writer describes a -tqwt in his letter. His pipe is very good. Many of his comments are correct with respect to his -tqwt. IMHO, -tqwt is the best quarterwave design possible. I think -tqwt and aperiodic speakers are the two best cabinet designs overall.

Challenging the value of OQR by presenting a -tqwt that is tapered, shorter, and more densely stuffed is inequitable. It would take an entire article to explain how and why his -tqwt is different and better than an OQR. (I have only sketched an outline for an article in my head. It will have to wait until after an article concerning stuffing adjustment.)

I will address three tapered system characteristics briefly: impedance curve, stuffing location, and driver placement. I was not explicit when describing stuffing and driver placement in OQR, but believe it is a good time to get the full explanation into print.

Driver placement is a function of cabinet volume above and below the speaker. I mean volume from driver to open end versus driver to closed end. The rule for parallel pipes applies to tapered pipes. A 5:1 volume ratio sets driver location.

This immediately implies two things. If the closed end is larger than the open end (the writer's -tqwt), the driver must move toward from the closed end. Second, the steeper the taper, the more the driver moves. A steep 12:1 taper requires more movement than a 3:1 taper. Mr. Kittinger's 16% location seems on target, though I did not do the calculation.

Next, the same basic principle applies to stuffing. The volume of stuffing at the closed end is about twice the volume of empty space at the aperture end. Denser stuffing may allow for some variation, but the 2:1 volume ratio is generally reliable.

I hope these two pointers save time for iterative modelers.

The impedance curve of the OQR is a telltale electrical characteristic. It does not indicate superior performance. It does indicate optimum design. You can over-stuff an

OQR to flatten the curve or under-stuff to be more like a bass reflex, but why would you? When the pulse-shape curve is present, the design is right. There indeed is something special about the OQR impedance signature.

Mr. Kittinger flames, I fail "... to fully acknowledge the work of others before [him]." I went to great lengths in "alpha Transmission Lines" to properly credit the long history of quarterwave design. The history came from Mr. Dlugos, Steven E. Schoenherr (<http://history.acusd.edu/gen/recording/loudspeaker.html>), and my own reading of patents and articles. Mr. Kittinger finds my acknowledgments "grossly inadequate."

Flames continue, "... [readers] might easily conclude the author did much more original thinking or work than I believe he actually did." My first paragraph claims this is the first quarterwave design based solely on TSP and not rules of thumb. As one example, I assert f_s/Q_{ts} is a more reliable predictor of length than the old rule of thumb. If one driver has $f_s = 40$ and $Q_{ts} = 0.5$ and another has $f_s = 80$ with $Q_{ts} = 1.0$, the pipe for each should be the same length?

That is original, specific, and new. Before last month, this thought did not appear in print or any form of public discussion. The placement-volume and stuffing-volume relationships for both tapered and parallel pipes within this letter are news.

My article ends by stating the individual adjustments are nothing new. What is new is bringing the adjustment together with TSPs into one design model. I welcome a discussion of prior quarterwave designs derived from TSPs.

P.S. Yes, a yardstick with a lump of clay is similar to an antenna with a golf-ball thingy. Yes, to some degree we have ripped off prior ogies. Of course mine was partially contrived. *aX*

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