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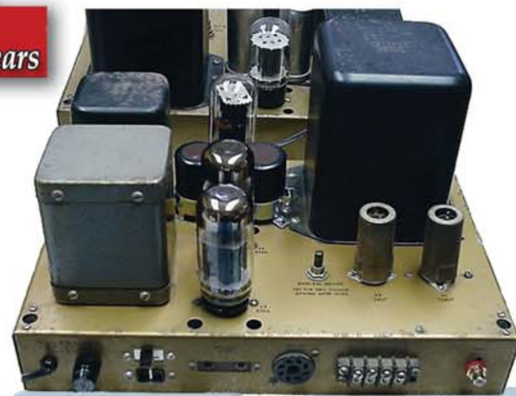

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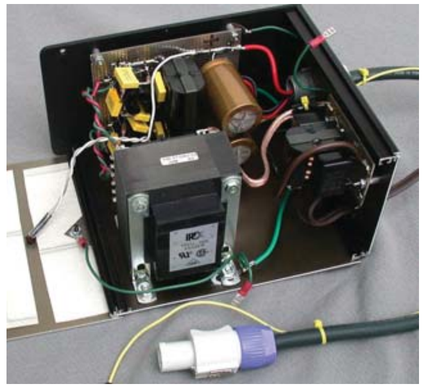
CONTENTS

VOLUME 35 NUMBER 12 DECEMBER 2004

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

FEATURES



page 26

DEPARTMENTS

page 20

IN EVERY ISSUE

THE STAFF

audioXpress (US ISSN 1548-6028) is published monthly, at $34.95 per year, $58.95 for two years. Canada add $12 per year; overseas rates $59.95 per year, $110 for two years; by Audio Amateur Inc., Edward T. Dell, Jr., President, at 305 Union St., PO Box 876, Peterborough, NH 03458-0876. Periodicals postage paid at Peterborough, NH, and additional mailing offices.

POSTMASTER: Send address changes to: ***audioXpress***, PO Box 876, Peterborough, NH 03458-0876.

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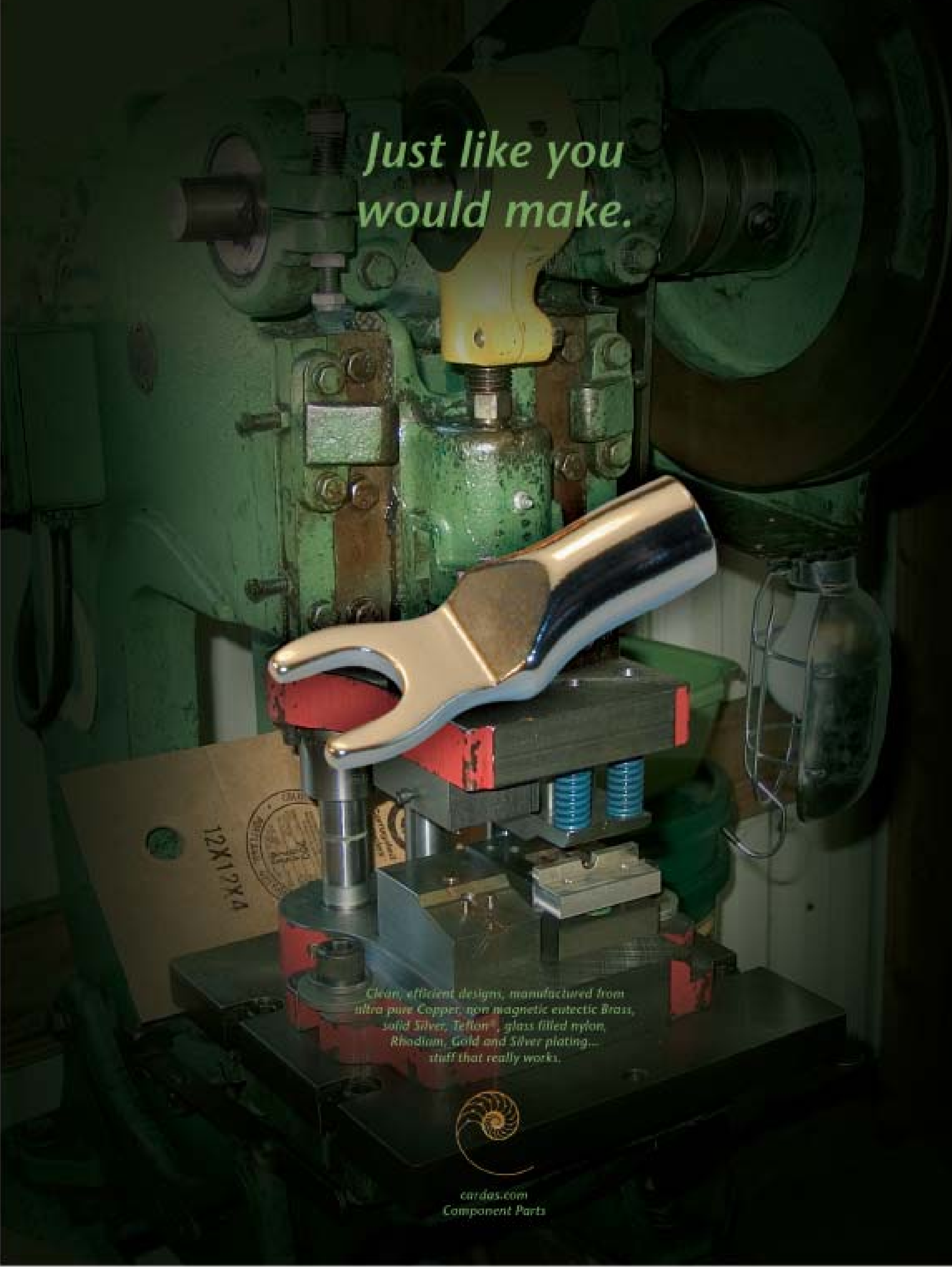
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Single-Ended to Differential Mode Made Easy

This article describes a very simple circuit for single-ended-to-balanced-mode conversion that, belying its simplicity, provides good performance. **By Norman Thagard**

PHOTO 1: Eight Citation IIs in one setting.

I had been intrigued for some time about the possibility of differentially driving a stereo amplifier in order to operate it as a monoblock at twice the single-channel power. As usual, curiosity alone was insufficient motivation. It was the acquisition of a number of 1960s vintage vacuum tube amplifiers along with the lack of a preamplifier with balanced output that finally provided the impetus to investigate the possibilities of such an operation along with suitable converters for its achievement.

SYSTEM SETUP

There might be other uses as well. For example, the A75[1] and DIFF 100[2] power amplifiers both accept balanced inputs. I have been unable to take advantage of this because I have no preamplifier with balanced output capability. This circuit, placed immediately at the output of a single-ended preamplifier, could be used with good results if some distance separated the amplifier and preamplifier. In such cases, balanced-line interconnection would take advantage of the balanced amplifier's common-mode rejection ratio (CMRR) to reduce hum and noise generated over the length of the interconnect run.


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The aforementioned DIFF 100 amplifier was overly conservatively rated. Using the circuit of this article, the monoblock-operated DIFF 100 delivered 243W into an 8Ω load at the clipping level, indicating a power level of greater than 120W per channel with both channels simultaneously driven. In general, a pair of similar amplifiers such as the two channels of a stereo amplifier may be driven differentially. Power will be twice that of a single member of the pair.

Thus, for stereo, you could set up a system with the configuration shown in *Fig. 1.* Each of the amplifiers numbered 1–4 should either be a monoblock or a single channel of a stereo amplifier. Interconnects are standard single-ended RCA types.

Note carefully that the amplifier's output ground terminals, labeled (–) in *Fig. 1*, would normally connect to the (–) loudspeaker connectors but are connected instead only to one another in differential-drive mode. Only the amplifier's "hot" output terminals, labeled (+), are attached to the speakers. The arrangement depicted will provide correct absolute and relative (left channel/right channel) phasing in the stereo setup unless either the preamplifier or amplifier—but not both—inverts the signal.

However, know thy amplifier! For example, the Pass Zen amp (*TAA* 2/94) not only inverts, but also to maintain correct absolute phase, the output ground and hot terminals were reverse-labeled (+) and (–), respectively. If two Zen amplifier channels are to be converted to a monoblock, you should connect one channel's (–) terminal to the (+) speaker terminal with the other channel's (–) terminal connected to the speaker's (–) terminal, tying the two (+) amplifier output terminals together in common. The Zen seems to tolerate a shorted output condition rather well, but there are amplifiers whose output would be destroyed by incorrect connections, so be careful.

Since the Zen does invert, you can maintain correct phase by reversing either the connection to the loudspeaker or from the phase splitter. Do not reverse both connections.

ABOUT THE AUTHOR

Five-time astronaut Norman Thagard was the first American to enter space aboard a Russian rocket for a 90-day mission to the space station Mir. With a total of 140 days in space, he became the most experienced US astronaut ever. In addition to an MS degree in engineering science from Florida State University, he holds a doctorate in medicine from the University of Texas Southwestern Medical School. He is currently Professor and Associate Dean for College Relations at the FAMU-FSU College of Engineering. An avid audiophile, he designs and builds audio amplifiers as a hobby. On May 1, 2004, he was inducted into the Astronaut Hall of Fame, located at the Kennedy Space Center Visitors Center, Florida.

BACKGROUND

In the mid-1980s, a former college roommate and longtime friend gave me his old Harman-Kardon (HK) Citation II tube amplifier. The unit was corroded, and the one working channel produced only about 50W instead of its rated 60W. I replaced every capacitor and resistor in the unit, which restored normal output in the one working channel. Unfortunately, the second channel's problem was a defective output transformer (OPT), the bane of the tube amplifier owner.

Eight years ago, a new friend gave me one of his two old Citation IIs in return for rebuilding the second unit as a gift for his son. Unlike solid-state amps, you can simply parallel the two channels of many stereo tube amplifiers for use as a monoblock possessing twice the per-channel power capability. The idea of placing two monoblock-configured tube amplifiers in my sound system excited me. I found a fellow who, at great expense, rewound the defective OPT.

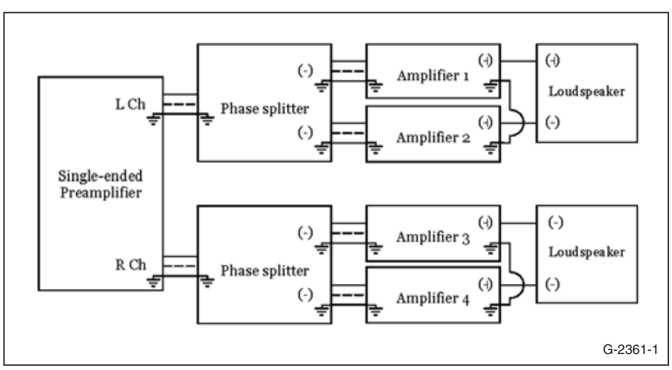

FIGURE 1: Single-ended-to-differential conversion in a stereo system.

The amplifiers these 120W monoblock-configured tube amps replaced were my 100W pure Class A solid-state monoblocks, whose designs were published in *Audio* magazine in 1995. I honestly heard no significant difference in the sound between the tube amps and the solid-state amps. However, the possibility of yet another power doubling intrigued me.

A visit to the Audiogon website revealed that a Citation II was available from a musician in Michigan who had at one time used it as a guitar amp. The unit was in a barn in Kentucky, and it was a couple of months before I actually took delivery. It was in reasonably good shape but did have some corrosion as well as a "Dirt Dauber" (sic) nest and spider webs inside. To compensate, the owner threw in a Citation I tube preamp for free.

Soon thereafter, I acquired a fourth Citation II from a seller on eBay. I completely rebuilt both amps, sanding the transformers to bare metal and repainting them satin black. They not only looked mean with black transformers, but, re-tubed with KT-90 output tubes, they would deliver about 65W per channel at 0.2% THD, beating HK's original specs of 60W per channel at 0.5% THD.

I simply paralleled the four 16Ω output taps of two stereo amplifiers to form monoblocks that could deliver 240W into 4Ω. This was perfect to drive my Martin Logan ReQuests with their 4Ω nominal impedance, and there was significant improvement in sound reproduction. Since the perception of improved sound persisted even after a year of frequent listening, I was inclined to believe it was real.

If four Citation IIs sounded so good, what might eight sound like? For one thing, simply continuing to parallel channels would not suffice. Amplifiers are ultimately power supply voltage-limited in their power delivery, and I had "maxed out" with the use of four parallel 16Ω output taps driving a 4Ω load.

As I've mentioned, you cannot safely parallel solid-state amps, and it is for this same reason that you should not parallel two voltage sources. However, you can safely place voltage sources in series. With amplifiers, you can realize this series arrangement of outputs through differential drive of their inputs, à la the monoblock-configured DIFF 100 bench test. Acquiring and rebuilding five more Citation IIs (you always need a spare), I was in a position to test such a configuration of four stereo amps per channel.

POWER

For initial testing, I used a LF411 op amp in unity-gain inverter configuration to generate the out-of-phase signals required for differential drive. Four channels of two Citation IIs were driven directly by a 1kHz sinusoidal test signal. I ran this test signal through the LF411 inverter to provide differential drive to the four channels of a second pair of amps.

I carried out the initial test with some trepidation. In 1964, I set a speaker on fire by foolishly using it as a dummy load in the engineering lab. While not worried about setting the resistive dummy load on fire in this case, I did have visions of smoke curling from the innards of one or more of the Citation IIs on which I had labored, and these power levels were well beyond my previous experience.

I also had never before seen such levels on the AC voltmeter of the distortion analyzer. With the Citation IIs' 16Ω output taps in parallel connected to an 8Ω resistive load, over 58V RMS registered, corresponding to about 425W.

Testing at this level was necessarily brief because the dummy load was a parallel-series arrangement of sixteen 20W noninductive resistors. Specifically to conduct full power, 4Ω load tests in this configuration, I constructed a second identical 8Ω, 320W dummy load. In the future, I will have the capability for unrestricted simultaneous testing of both channels of stereo amplifiers into 8Ω loads at power levels up to 320W/channel or single channel, 4Ω load testing up to 640W.

I conducted the next test with the two 8Ω dummy loads in parallel driven by paralleled 8Ω amplifier output taps. Again, about 425W were delivered at the onset of clipping. With eight 60W amplifiers, the expectation was at least 480W in both tests.

Due to the high output resistance of even large power tubes, most tube amplifiers transformer-couple the output to the load. The Citation II provides three output connections from separate taps on the secondary of the OPT—one each for 4-, 8-, and 16Ω loads. The rated power of 60W will be delivered only if the load impedance matches the output connection; e.g., if the 4- or 16Ω tap of the OPT powers an 8Ω speaker, less than 60W can be supplied.

Was an impedance mismatch the cause of the reduced power output? After all, this configuration was a little more complex than connecting one speaker to one amplifier connection. Perhaps I had failed to account for some nuance of the configuration.

IMPEDANCE MATCHING FOR OPTIMAL POWER DELIVERY

I reasoned that the test configuration was, in simplified form, as shown in *Fig. 2*. Each voltage generator of magnitude $\frac{1}{2}v_g$ represents the output voltage of four paralleled Citation II channels; i.e., all channels of two stereo amplifiers, with $R_g = \frac{1}{4}R_{tap}$ representing the effective output impedance. Thus, depending upon whether you use the 16-, 8-, or 4Ω output taps, R_g will effectively be 4-, 2-, or 1Ω, respectively.

I connected the ground (–) terminals of all eight channels together in common as shown. Note that it is one or another of the three output taps of four channels in parallel that attach to the top of load R_l with the second group of four channels attached to the bottom. One group of four channels is driven by a signal that is 180° out of phase with the drive signal to the second group, resulting in differential output of magnitude v_g. Thus, the already simplified schematic of the left side of *Fig. 2* can be modeled in the even simpler form of the right side.

I have omitted any reactance because we are interested in real power and because complicating the situation with complex variables adds little to understanding. Although obvious by inspection of the complex power expression, it is offered without proof that if $Z_g = 2R_g + jX_g$ and $Z_l = R_l + jX_l$, then $X_l = -X_g$ is optimum. Real power delivered to the load will be:

(1) $$i_l^2 R_l = \left[v_g / \left(2R_g + R_l\right)\right]^2 R_l = v_g^2 R_l \left(2R_g + R_l\right)^{-2}.$$

If this were graphed against R_l with a fixed value for R_g, the power would show a peak; i.e., its slope would be zero, at the point $R_l = 2R_g$.

To prove this mathematically involves some simple calculus. The slope of this graph is the derivative of the power expression with respect to R_l. At a peak or at a minimum, this slope is horizontal and consequently is zero. Thus, you take the derivative, set it to zero, and solve the resulting equation for the optimum value of R_l:

(2) $$\frac{d}{dR_l}\left[v_g^2 R_l \left(2R_g + R_l\right)^{-2}\right] = v_g^2\left[\left(2R_g + R_l\right)^{-2} - 2R_l\left(2R_g + R_l\right)^{-3}\right] = 0.$$

The solution to the algebraic equation on the right is $R_l = 2R_g$, verifying the graphical implication.

Since in my case I had a speaker whose nominal impedance was 4Ω and needed to determine which Citation II output tap, 4-, 8-, or 16Ω, to use for maximum power delivery, you might think that power should be differentiated with respect to R_g. However, there is no value of R_g for which the derivative would be zero because power is maximum at $R_g = 0$ and decreases continually as R_g increases from zero. This seems intuitive and can certainly be seen by substituting various values for R_g in the load power expression $v_g^2 R_l / (2R_g + R_l)^2$ for given fixed values of v_g and R_l. *Any* nonzero value of R_g lowers load voltage (and current) and therefore power delivered to the load.

In any event, the appearance here is deceptive and as you will see, the task really is to vary the (apparent) load resistance to the inherent, fixed generator (output) resistance of the amplifiers. Thus, the appropriate action is to solve equation (2).

The typical solid-state audio amplifier has inherently low output resistance even before application of negative feedback. Damping factors of 100 or more with respect to 8Ω loads can be seen that, given the definition of damping factor, means such an amplifier has $R_g \leq 8/100 = 0.08Ω$. Most solid-state amplifiers therefore operate almost like ideal voltage sources, and there is no possibility or need to match impedance. Imagine the current draw and power delivered if the loudspeaker had nominal impedance of 0.1Ω or less! Of course, there are few solid-state audio amplifiers capable of operating into such impedance.

Unlike a transistorized amplifier, a tube amp has inherently high output resistance, and via its OPT *does* operate on the basis of matched impedance. Therefore, optimum power delivery should occur when $2R_g = R_l = 4Ω$, meaning the output taps used in the present case should be $R_{tap} = 4R_g = 2R_l = 8Ω$.

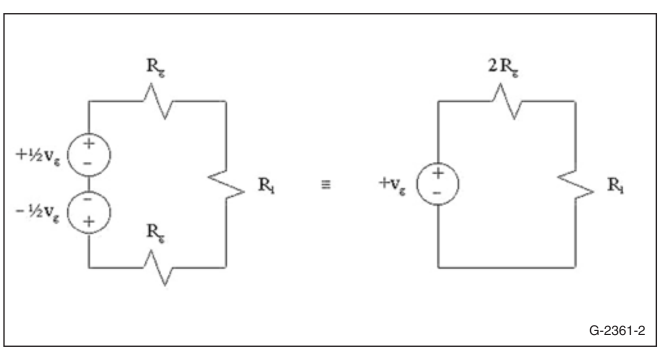

FIGURE 2: Simplified test configuration.

REFLECTIONS

While the foregoing discussion seems to imply that by selecting different OPT taps you are changing the amplifier's output impedance for optimal matching to a fixed load impedance, you are really changing the load impedance seen at the plates of the amplifier's output tubes. That impedance is $R_l\, n^2$, where n is the primary to secondary turns ratio from the utilized secondary tap of the OPT. By changing OPT taps, you seek to select n for which $R_l\, n^2 = R_p$. Here, R_p is used to represent the output impedance looking into the push-pull output stage of the Citation II amplifier. R_p is typically several kilohms (kΩ) for a tube-based audio power amplifier.

In summary, I really was varying load impedance for optimal matching to fixed output impedance in line with theory. The test bench results certainly indicated that this analysis applied to the configuration, but why, then, was the power lower than expected?

For all my misgivings about the validity of my assumptions concerning the configuration and this discussion notwithstanding, I am not such an expert on tube amplifiers to state unequivocally that they are completely valid. However, the apparent cause of reduced power output proved more mundane. Finally checking the power line supplying the test bench, I found that the poor 15A circuit supplying the four Citation IIs was sagging badly to 107V AC with the amps driven to clipping.

This is a caution to those who would place very high power amplifiers in their systems. A stock Citation II draws 350W from the power line. Even with the output stage bias reduced to 67mA/KT-90, my units draw almost 300W for a total of nearly 1.2kW *per channel*. One wag already jokes about buying stock in the local power company.

It is cold in Tallahassee as I write this, and I rather appreciate the 2.4kW space heater in the sound room. There is a dual outlet on a single 15A circuit and four single outlets on four independent 20A circuits. I had these latter outlets installed by an electrician soon after moving into my current home in anticipation of future folly, so there is no problem with line voltage sag in actual usage.

I ordered a 2kW Variac® to replace the old 500W unit on my test bench. This allowed line voltage to the amplifiers to be maintained at 117V AC during full-power testing per Harman-Kardon specifications. With this line voltage level, the amps squeezed out 481W at 0.5% THD, exactly what you would expect for eight channels based on single-channel specs.

While waiting for its arrival, I reconsidered the inelegant use of the LF411 inverter as a source of differential drive. There would be more circuitry in the inverted signal path than in the noninverted path. Although I am a practical engineer with a good measure of skepticism about the audibility of various topologies, types of interconnects, speaker cables, op amps, and the like, I nonetheless aesthetically disliked the asymmetry of the op-amp inverter approach. Was there a simple satisfactory alternative?

THE DIFF AMP

I know well and love the differential amplifier stage. All but one of my amplifier and preamplifier designs have used a dual differential input stage. The problem is application of feedback in a single-ended to balanced configuration. I played around with a two-stage differential design that yielded pretty good results.

This topology (*Fig. 3*) has the advantage that no capacitors are in the signal path. The disadvantages are several. For one thing, it requires six transistors, two diodes, and an op amp. Also, although differential output is provided, only the noninverting output is sampled and fed back to the input. With 100% feedback, it is guaranteed that the noninverting output will have a DC level very close to zero, but the symmetry of the circuit is not perfect, so the inverting output will not necessarily have a near-zero DC level.

This problem is minimized by incorporating a DC servo, implemented with an op-amp-based differencing integrator. At frequencies below $[2\pi(1M\Omega)(1\mu F)]^{-1} = 0.16Hz$–i.e., at DC–the servo will adjust current source current to the second differential amplifier stage to maintain the inverting output near zero.

Another problem with the circuit of *Fig. 1* is that inverting output distortion, while low, was higher than noninverting output distortion. In fact, noninverting output distortion was below the 0.003% THD floor of my Krohn-Hite distortion analyzer.

Were I to use the circuit, I would choose different transistors. The matched dual *n*-channel JFET is no longer available, and the TIP 30 *pnp* BJT is commonly used in power applications. Even so, while the devices and values shown worked well enough, was there a simpler solution?

THE CONCERTINA PHASE SPLITTER

When tube amplifiers ruled the audio world, it was necessary to provide differential drive to the push-pull output stage. This is not a requirement in complementary-symmetry push-pull output stages, but tubes do not come in complementary variants.

Not surprisingly, differential amplifier stages were often used to generate the two out-of-phase drive signals required for push-pull operation. There was an alternative that was also widely used because it offered good performance in a simple topology. It was called the concertina phase splitter. The tube-based version is shown in *Fig. 4*.

Obviously, this is simplicity itself. R_p was made equal to R_k, so neglecting any load effects and assuming grid current is zero, plate current, i_p, must equal cathode current, i_k, and both output voltages, $-i_pR_p$ and i_kR_k, must have equal magnitude. Gain magnitude is, in fact, unity, which is readily understood, given that the noninverted output is that of a cathode follower. The phase splitter is, in fact, both a cathode follower (common-plate) and common-cathode amplifier.

Exactly analogous to the behavior of common-emitter or common-source transistor stages, the output taken at the plate (common-cathode) is inverted from the input because increased input voltage produces increased plate current, greater voltage drop across R_p, and consequently lower voltage at the plate. The circuit therefore does its job of accepting a single-ended input and generating differential outputs at unity gain.

Not shown in *Fig. 4* are the two or three coupling capacitors required by the circuit. After all, neither the grid nor either of the two outputs is at ground potential. At the input, you need only a small capacitor because the grid resistance is so high. At the output, however, the input resistance of the thing being driven will dictate the necessary minimum capacitor value.

THE TRANSISTORIZED SOLUTION

The transistorized phase splitter circuit is shown in *Fig. 5*. The coupling capacitors should be large enough in value that response is flat down to 20Hz. For that to be the case, the cutoff frequency should be at least one decade below 20Hz or 2Hz.

For my case, each output drives all four channels of two Citation II amplifiers. The Citation II has an input resistance of 1MΩ, the phase splitter outputs see 250kΩ, and a capacitor of 1µF is sufficient for a low-frequency cutoff $f = (2\pi RC)^{-1} < 1$Hz. Conservatively, two 2µF film capacitors in parallel formed the output coupling capacitors.

I employed little conservatism at the input. There, the resistance posed to the driving source is $R_{in} = R_B \| [\beta (R_E + r_e)]$, where $R_B = R_1 \| R_2$ = 15k‖39.2k = 10.8kΩ, and R_E = 470Ω is the extrinsic emitter resistor. The intrinsic emitter resistance—the resistance "seen" looking into the emitter—is $r_e = V_T/I_{CQ} \cong$ 25mV/25mA = 1Ω. This expression is derived from the exponential Ebers-Moll equation that models the transistor's transconductance behavior.

Quiescent collector current, I_{CQ}, for this circuit will be seen to be nearly 25mA, hence the use of that convenient value in calculating r_e. For almost any value of extrinsic emitter resistor, it is apparent that r_e can be neglected, especially if I_{CQ} is more than a few milliamperes.

I purchased the particular 2SC2682 BJTs that I placed in the two-channel phase splitter several years ago from a lot whose devices were guaranteed to have $\beta \geq 280$. Therefore, circuit input resistance is expected to be around 10kΩ, which was the desired design value to avoid loading the preamplifier. The 10µF capacitor specified results in a cutoff frequency slightly less than 2Hz. With physically small but otherwise good-quality 5µF film capacitors in my parts bin, two in parallel couple the preamplifier signal to the phase splitter. Response at 20Hz was less than 0.1dB down from the 1kHz response, based on the 0.1dB resolution of the AC voltmeter in the distortion analyzer.

It is possible to eliminate the input coupling capacitor through the use of a MOSFET, as shown in *Fig. 6*. You could attempt this scheme with a BJT, but you would need to eliminate the 51.1kΩ ground reference resistor at the input. This would make the circuit ground reference depend upon the signal source, leading to a long circuitous ground path, which is probably not a good idea.

Then, too, any interruption in this path, such as can occur when the preamplifier is switched from CD to tuner, may momentarily lift the ground reference. This will upset the bias of the

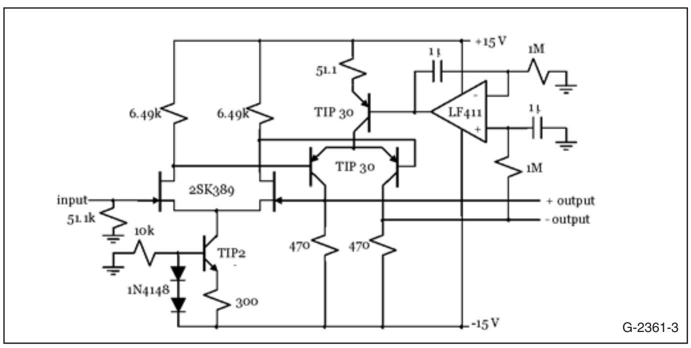

FIGURE 3: Single-ended to balanced converter using two-stage differential amplifier.

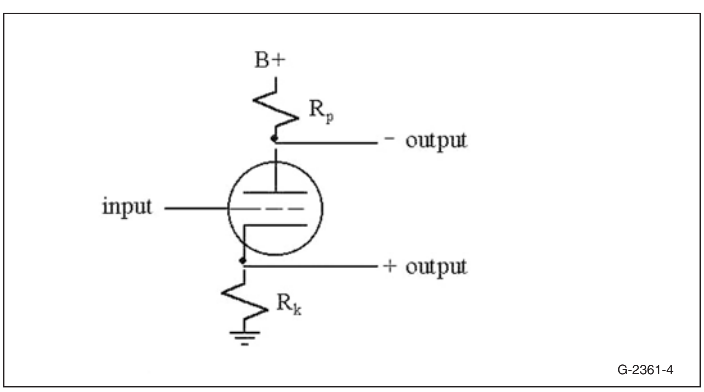

FIGURE 4: Concertina phase splitter.

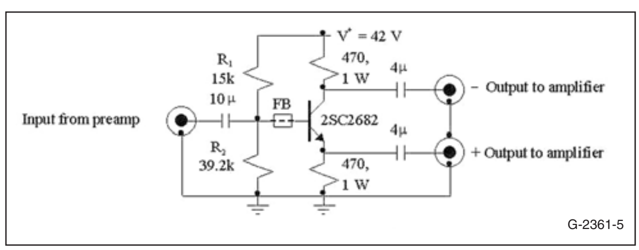

FIGURE 5: Bipolar junction transistor phase splitter.

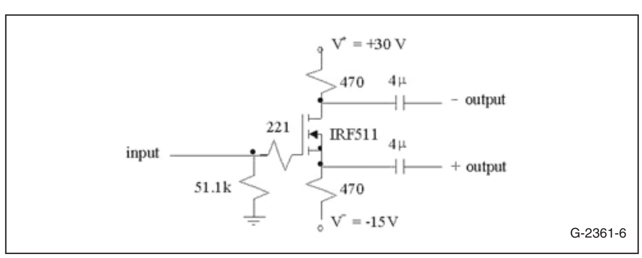

FIGURE 6: MOSFET phase splitter.

BJT, potentially producing a loud "pop." I know from trying this, but the scheme worked quite well as long as the input was unchanged. However, with eight 120W tube amplifiers waiting to exercise all 960W of their capability, pops were disallowed.

The advantage of eliminating the input coupling capacitor is obvious, because good-quality 10µF film capacitors are neither cheap nor physically small. One resistor is also eliminated.

The MOSFET-based splitter had significantly higher distortion than the BJT version. On the plus side, this distortion was nearly constant at about 0.02% over the entire 20–20kHz spectrum.

Another disadvantage is that a bipolar power supply is required. Quiescent source voltage will be about 3.5V, so a negative supply of 15V results in 11.5V across the 470Ω source resistor. This is very close to the desired value of $¼(V^+ - V^-) = ¼[30 - (-15)] = 11.25V$, which leads into the discussion of bias.

BIAS LEVEL

It is a simple matter to determine the proper bias conditions for the phase splitter. The extrema occur when the active device—be it MOSFET, BJT, JFET, or vacuum tube—is fully off or fully on. If fully on, both outputs—(neglecting BJT saturation voltage) will be—$½(V^+ - V^-)$. If fully off, inverting output will be at V^+ and noninverting output will be at V^-.

For symmetrical and therefore maximal undistorted output voltage swing, source or emitter voltage should be at $¼(V^+ - V^-)$, while drain or collector voltage should be at $¾(V^+ - V^-)$. This allows both outputs to swing as much as $\pm¼(V^+ - V^-)$ about their quiescent states. For simplification in the single supply case, note that $V^- = 0$.

To keep distortion reasonably low, you should limit swing to slightly less than $\pm¼(V^+ - V^-)$. A kind of standard sensitivity for audio power amplifiers is to produce full-power output at about 1V RMS input or so. The Citation II is designed to deliver 60W per channel into 4-, 8-, or 16Ω loads when input is 1.5V RMS.

There are amplifiers with significantly lower sensitivity. Again using Nelson Pass' Zen amplifier as an example, a source capable of 3.5V is said to be the requirement. The beauty of the concertina phase splitter is that almost any sensitivity can be accommodated if supply voltage(s) is raised sufficiently. A note of caution: choose devices whose $V_{CE\text{-max}}$ rating is not exceeded.

I desired some overload margin, primarily for design conservatism. A 42V power supply is sufficient for output voltage swings up to 21Vp-p, with good linearity up to 18Vp-p or so. This translates to 6V RMS for 12dB overload margin relative to 1.5V RMS. With 1V RMS output at 1kHz, the as-constructed amplifier had 0.003% THD at the noninverting and 0.005% THD at the inverting output. This increased little until 3V RMS was exceeded, reaching slightly less than 0.05% at 5V RMS. Again, you can control this almost at will by use of progressively higher supply voltage(s).

As for the choice of load resistors, I chose the value on the basis of the minimum that would not require transistor heatsinking. Even small-signal transistors can usually tolerate ½W dissipation. With the 42V power supply, $V_{CEQ} \cong$ 20V. $I_{CQ} \cong I_E \cong ¼(V^+ - V^-)/R_E = ¼$ (42 –

0)/470 ≅ 23mA, and transistor power dissipation is about 20 × 0.023 = 0.46W. The 2SC2682 is a medium-power transistor that can easily dissipate ½W without a heatsink.

A minimum value for load resistors is desired because inverting output resistance is essentially equal to the collector load resistor value, and keeping output resistance small minimizes loading effects. To drive four paralleled Citation II channels such small values are unnecessary, but amplifier input resistances can be as low as 10kΩ or even lower and a 1:10 ratio is usually recommended. Therefore, to give this design more general utility, I applied this criteria in calculating load resistor values.

STABILITY RULE OF THUMB

Refer to *Fig. 5* to calculate base bias resistor values. The requirements were $R_{in} = R_B \| [\beta(R_E + r_e)] \geq 10k\Omega$ and $V_B = V_E + V_{BE} \cong \frac{1}{4}(V^+ - V^-) + 0.7 = 11.2V$. To ensure that adequate base bias current would be available, I also applied the stability rule of thumb, $R_B = 0.1\beta R_E$.

A strict calculation of the divider ratio required to simultaneously ensure a value for R_B such that $R_{in} = \geq 10k\Omega$ and $V_B = 11.2V$ involves solving Kirchhoff's voltage law around the Thevenin-equivalent base-emitter circuit, which sounds worse than it actually is. The rule-of-thumb ensures the "goodness" of the voltage divider so that, to the accuracy that is needed, you can assume that currents through both resistors comprising the divider are equal.

The numerical conditions based on this are $[R_2/(R_1 + R_2)](V^+ - V^-) = 11.2V$ and $R_B \geq \{[\beta(R_E + r_e)]R_{in}\}/[\beta(R_E + r_e) - R_{in}]$, where $R_{in} = 10k\Omega$. The former equation ensures that R_1 and R_2 will be such that emitter voltage will be $\frac{1}{4}V^+$; i.e., $V_E = V_B - V_{BE} \cong 11.2 - 0.7 = 10.5\ V = \frac{1}{4}V^+$. Since collector voltage will then necessarily be about $\frac{3}{4}V^+ = 31.5V$, you can achieve maximum symmetrical voltage swing at both outputs. The latter equation constrains R_1 and R_2 to values that, in conjunction with $R_E = 470\Omega$, will maintain input resistance at or greater than 10kΩ per the design criteria. Of course, there is also the rule of thumb $R_B = 0.1\beta R_E$.

There are three conditions, but only two unknowns. While this might suggest that the values of R_1 and R_2 are over-determined, this is not the case. The rule of thumb is potentially in conflict with the input resistance criteria. If this sounds confusing, consider the actual design sequence:

1. Set power supply voltage to allow 5V RMS output.
2. Set collector load resistor to the lowest value consistent with no heatsink operation; i.e., to 470Ω, and let $R_E = R_C$.
3. Choose R_B consistent with emitter voltage $V_E = \frac{1}{4}V^+ = 10.5V$, and $R_{in} \geq 10k\Omega$.
4. Verify that $R_B \cong 0.1\beta R_E$.

The tendency will be for the rule of thumb to require values for R_1 and R_2 that are too low to meet the input resistance criteria. If, in step 4, $R_B >> 0.1\beta R_E$, then a possible solution is to choose a transistor with higher current gain. With $\beta = 280$, the 2SC2682 is already close to the highest available in a medium power transistor, but I tried the MPSA18 with $\beta \geq 500$ at the breadboard stage and it worked OK with slightly higher distortion than the chosen device, so it could be a suitable alternative.

On the basis of these equations, numerically for this design $R_2/(R_1 + R_2) = 0.27$ and $R_1R_2/(R_1 + R_2) \geq 10.8k\Omega$. This leads immediately to $R_1(0.27) \geq 10.8k\Omega$ or $R_1 \geq 40k\Omega$. The closest standard 1% value is 40.2kΩ, but I used 39.2kΩ because I had several 0.5% metal film resistors of this value in my parts bin. I accepted the fact that input resistance might be slightly less than the desired design value.

$R_2 = 0.27R_1 - 0.27R_2 \Rightarrow R_2 = 0.27R_1/(1 - 0.27) = 0.37R_1 = 0.37(40k) = 14.8k\Omega$.

The closest 1% value is 14.7kΩ, but I used 15kΩ resistors because I already had them on hand. Input resistance with the schematic values should be at least 10,023Ω, so the input resistance criterion is met despite the deviations from the calculated values. Emitter and collector voltages differ a little from ideal due to the resistor values, which slightly reduces the overload margin but otherwise is of little significance.

Checking the rule of thumb, $R_B = R1\|2 = 39.2k\|5k = 10{,}849\Omega$ and $0.1\beta R_E = 0.1(280)(470) = 13{,}160\Omega$. This indicates that input resistance could have been raised a bit without violating the rule-of-thumb. After all, the rule-of-thumb is an approximation, not a rigidly fixed value, so the energetic builder might choose to raise R_1 and R_2 values to reach an input resistance of 15kΩ.

PERFORMANCE SUMMARY

I've already partially covered the circuit performance parameters in the text. To summarize, at 1V RMS output, THD was 0.003% at 1kHz for the noninverting and 0.005% at the inverting outputs, rising to 0.006% and 0.009%, respectively, at 20kHz. Response was flat within 0.1dB from 20–20kHz. These results were identical for both channels that were constructed.

Matching between inverting and noninverting output levels was 0.1dB, even though emitter and collector resistors were only 2%-tolerance metal film power resistors, which were not selected but were from the same Digi-Key lot. By placing higher-value resistors in parallel with either the emitter or the collector resistor as appropriate, you could achieve any degree of desired matching.

With the 2SC2682 BJT, the circuit oscillated unless power supply bypassing was carefully attended. The ferrite bead shown in the BJT circuit eliminated this oscillation without regard to power-supply considerations. I saw no oscillation with the MPSA18 transistor. The gate resistor for the MOSFET circuit is recommended for the same reason the ferrite bead was employed in the BJT circuit. It may seem strange that a unity-gain circuit can oscillate, but it can[3].

For the lawyers among you, let me state that **I assume no responsibility for damage to amplifiers or humans in the use of the phase splitter**. I bench-tested the variants shown, and the recommended version has been in the 480W per channel system for six months with no problems as of this writing. However, it was pointed out that amplifiers differ in their internal configurations and if not taken into account, this can lead to improper interconnections that can cause amplifier damage. Inadvertent failure to connect inputs to one Citation II resulted in a dull red glow of the KT-90 plates at high volume as the working amplifier attempted to drive the idle amplifier's outputs. Fortunately, this was quickly corrected with no damage.

This article describes one circuit. Two are required for stereo as shown in *Fig. 1*. All parts including the specified transistor should be readily available. I found an Internet source for the 2SC2682 at a unit price of $0.97. You can use many transistors as substitutes. The SK9041 and NTE373 are replacement series devices roughly equivalent to the 2SC2682.

Of all the transistors I used, the 2SC2682 provided the lowest distortion, but did not beat the MPSA18 by much in this regard. The MPSA18 has the advantage that typical current gains are 1,000, permitting the input resistance of the circuit to be raised to nearly 50kΩ. On the other hand, the MPSA18 is not usually operated at 23mA collector current, although this does not exceed Motorola's rating.

FINAL COMMENTS

I guess that there are those who would like to read some comment about the 480W per channel, tube amplifier performance. It has been my pleasure to listen to some pretty good systems, including the one at Pass Labs used for auditioning some of the finest solid-state preamplifiers and amplifiers to be had.

First, the system includes a TNT 3.5 turntable, JMW 10 tonearm, and Audio Technica OC9ML/II moving-coil (0.4mV) cartridge. The phono pre-preamplifier is a version of a design published in *audioXpress*[4] modified for greater gain to accommodate the low-output cartridge. The output of the phono pre-preamp directly feeds the input of a Krell PAM-5 preamplifier, which feeds the phase splitter, which, in turn, drives the eight Harman Kardon Citation II tube amps in a four and four, 480W per channel arrangement for stereo. Loudspeakers are Martin Logan ReQuest hybrids. For SACD/CD/DVD source material, I used a Philips 963 player.

There is no question in my mind that power and lots of it is required for accurate reproduction of sound. Even for those with very efficient speakers, I believe the sound would benefit from higher-powered amplifiers. Power-handling capability of the ReQuest is 250W and few loudspeakers can handle 480W, even on a short-term basis.

Obviously, the idea is not to achieve speaker-damaging power levels. A circuit operating closer to its small-signal ideal is inherently more linear, and I believe that this at least contributes to the improved sound. Then, too, sound sensitivity is logarithmic, so squeezing out those last few dBs of sound pressure level at loud volume is not possible without power reserve.

There is—I have heard—a tendency to turn up the volume to just below the point where distortion becomes objectionable. This notion certainly coincides with my experience. For realism, it seems that levels must be close to live levels. Based on sound-pressure levels, the volume is now significantly higher, but it paradoxically sounds no louder than before. Some of you may have wondered why a live orchestra sounds good, while the reproduction of an orchestral performance in a sound system can sound unpleasantly loud even though the actual sound pressure levels are less in playback than in performance.

THE BIG LEAGUE

With the current system, the effect is almost visceral. Joy knows no bound when the reproduced sound of an instrument is suddenly so real that it evokes involuntary laughter. Every aspect of the listening experience that I can think of is better now, and I believe that I have moved into the big league of sound reproduction.

I saw an Internet ad for a used 90-tube, 900W per channel amplifier recently. The price was in excess of $60,000. Although the assemblage of Citation IIs described here has only about half that power rating, the total cost including replacement of every resistor, capacitor, and tube was less than $8,000 spread over many years. With good, used tube amplifiers often advertised on eBay and Audiogon, this approach is a relatively inexpensive way to join the big league.

Before motivating you to do something foolish, I should point out that this method of power augmentation is not for the faint-of-heart. The amplifiers occupy a lot of real estate, weigh a total of 500 lb, and the idle power consumption makes the amplifiers expensive to operate, a factor exacerbated by a requirement for increased air conditioning in warm weather.

Interconnect cost can be enormous if you purchase audiophile cables. Because I placed the phase splitter in an old SQ quad converter (how appropriate) utilizing the ten RCA jacks already mounted on it, four 2-jacks-to-1-plug adapters were required along with a total of 13 1m stereo interconnects. Although I used some inexpensive (but nonetheless *Stereophile*-recommended) AR interconnects at about $20 each, total cost came to nearly $300. Similarly, there are 18 speaker cables to manage. For this purpose I built two adapter boxes, one for each channel.

I beg audiophiles' forgiveness because 1′ lengths of 18-gauge zip cord connect the Citation output terminals to the adapter box. Internally these are connected appropriately to two heavy-duty, five-way binding posts. Connection from these binding posts to the terminals on the ReQuests is via 12-gauge Monster Cable terminated in dual banana plugs. You must take great care in the management of so many connections.

On the input side, some of the complexity would be removed by using a balanced preamplifier, although I would still have to construct XLR-to-RCA adapters. Even so, at some point, I intend to replace the phase splitter and the Krell with a balanced preamplifier. For now, I'm too busy listening.

I would like to thank Mr. Jim McShane for his useful information as well as parts for Citation II restoration. ❖

REFERENCES

1. Norman Thagard and Nelson Pass, "Build the A75 Power Amplifier," Parts 1 & 2, *TAA*, 4/92 & 1/93.
2. Norman Thagard, "A 100W/Channel AB Differential Input Amplifier," Parts 1 & 2, *audioXpress*, Nov. and Dec. '02.
3. Thomas C. Hayes and Paul Horowitz, Student Manual for *The Art of Electronics*, p. 221, Cambridge University Press, 1989.
4. Norman E. Thagard, "A Phono Pre-Preamplifier for the CD Era," (*audioXpress*, Jan. and Feb. '01.)

H.V. Differential Amplifier

This low-cost diff amp has good precision and allows you to accurately measure different signals in the presence of high common-mode voltages, such as those you might find in tube equipment or bridged tied output power amplifiers. **By Charles Hansen**

PHOTO 1: Front view of high voltage differential amplifier.

An oscilloscope is a very handy tool in audio work, and a differential plug-in allows you to view waveforms that are not referenced to ground. The 7A22 plug-in for my Tek 7603 can work at common-mode voltages as high as ±1000V; or when used at a more modest common-mode voltage of ±1V, it can make differential measurements down to 10μV/div. A high-quality differential amp is fairly costly—about several hundred dollars on eBay.

You can build this more modest high voltage differential amplifier (HV diff amp) for about $60. It is based on the Analog Devices AD629 instrument amp IC. This is a monolithic difference amplifier with a high input common-mode voltage range. The AD629 will operate over a ±270V common-mode voltage range, and its inputs are protected from common-mode or differential voltage transients up to ±500V.

Specifications for the prototype unit are shown in *Table 1*. The parts list is in *Table 2*.

HOW IT WORKS

The schematic diagram for the HV diff amp is shown in *Fig. 1*. The main power source is an 18V AC plug-in AC adapter, or optional power by two 9V batteries. Battery power will reduce the common and differential input range, but allows portability if that is a requirement.

S2 is a DPDT center-off switch that can select either the battery power or the 18V AC input from the plug-in AC adapter at J4. When you select battery power, C5 and C6, in conjunction with ferrite beads FB1 and FB2, low-pass filter any HF noise, while C7 and C8 are the main reservoir caps. C9–C12 provide local supply bypass for U1 and U2. LED1, R6, and R7 function as the power-on indicator.

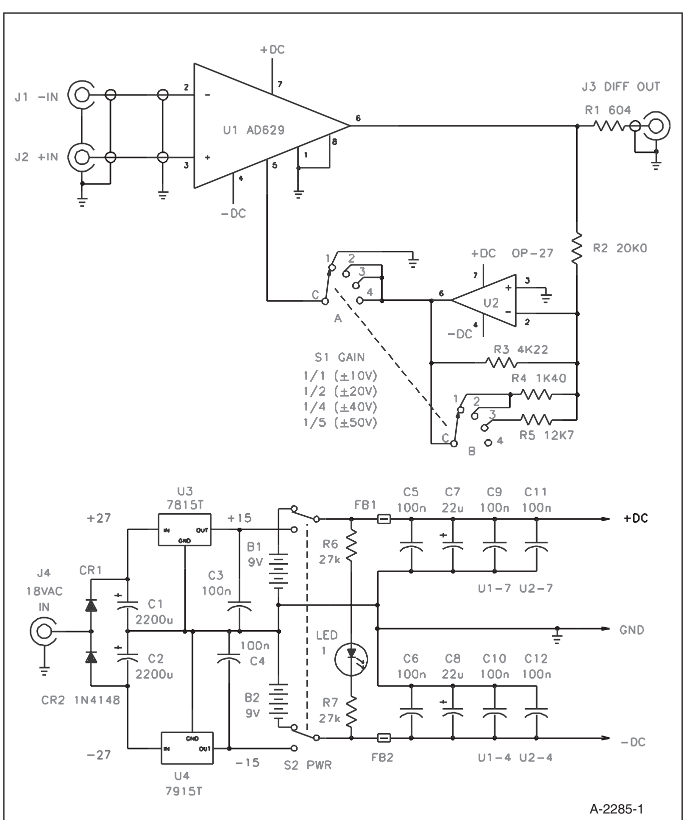

FIGURE 1: HV diff amp schematic.

TABLE 1
SPECIFICATIONS

DC coupled
Common-Mode Input Range
- ±270V (AC power)
- ±150V (9V batteries)
- ±500V transient protection (10ms)

Differential Mode Input Range (AC power)
- Unity Gain (1/1), ±10V type (13V max)
- ½ Attenuation, ±20V
- ¼ Attenuation, ±40V
- 1/5 Attenuation, ±50V

Differential Mode Input Range (battery power)
- Unity Gain (1/1), ±6V
- ½ Attenuation, ±12V
- ¼ Attenuation, ±24V
- ⅕ Attenuation, ±30V

Gain–Bandwidth DC–200kHz (–3dB)
Slew Rate 2.1V/μS
Settling time
- 0.1% 10V step; 12μS
- 0.01% 10V step; 15μS

Input Impedance (±25%) 800K differential, 200K common–mode
Common–mode rejection
- DC 86dB, Rsource <75Ω
- AC (60Hz) 66dB, Rsource <75Ω

Output swing ±12V (AC power), ±5V (batteries)
Output impedance 600Ω
Distortion 0.005% at 1kHz
Output Noise
- 25μVp–p, 0.01Hz to 10Hz
- 550nV/√Hz

In order to minimize power drain on the batteries, I chose an LED that has noticeable brightness at only 0.3mA.

When you select AC power, CR1 and CR2 half-wave rectify the 18V AC into roughly ±25V DC. C1 and C2 are the main reservoir caps, and U3 and U4 are linear regulator ICs that provide ±15V DC at the supply rails for U1 and U2. C3 and C4 ensure stability for the regulators.

The AD629 does most of the work, since it has 380k resistors in series with each input lead. The differential gain is Av = 1 (0dB). Differential input signals are applied to U1 through J1 and J2. The ground-referenced differential output is coupled to J3 via current-limiting resistor R1, which also sets the output impedance at about 600Ω.

The differential input voltage is ±13V maximum for ±15V supply rails, and proportionally less when operating on 9V batteries. There are instances when you may prefer to trade gain for a wider differential input range. I came across a method for doing this in a Burr-Brown applications note (Application Bulletin AB-001) for the INA117 difference amplifier IC, which is an earlier implementation of a high-voltage in amp. The AD629 is pin-compatible with the INA117.

Precision op amp U2 inverts a fraction of the output signal by means of R2 and one of the resistors selected by S1 (R3, R4, or R5). The inverted voltage is fed back to U1 REF+ pin 5, which reduces the gain of U1 in proportion to the resistor ratios. In the unity gain position (1/1), pin 5 is grounded to return to the "data sheet" configuration.

S1 can select from unity gain (±10V nominal differential range, ±13V maximum) to ⅕ gain with a ±50V nominal differential range (±65V maximum). The INA117 is limited in its gain reduction capability because it is unstable for gains less than 0.2 (⅕). I assume that this same limit applies to the AD629. An added advantage of the increased differential range is that the output noise is reduced by the same factor. The common-mode reduction ratio (CMRR) is preserved regardless of the gain reduction.

The DIP version of the AD629 is no longer listed in the latest Newark Electronics catalog. You may be able to obtain it from Analog Devices directly (www.analog.com), or use the SOIC version. The DIP version of the INA117 is available from Digi-Key, but observe the reduced common mode and differential input ranges as compared with the AD629.

CHASSIS ASSEMBLY

I built the HV diff amp into an aluminum enclosure. The front view is shown in *Photo 1*. I put a label on the top of the box showing the input and output operating limits, for easy reference.

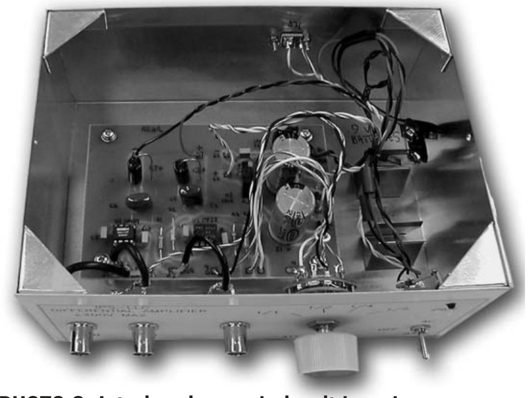

PHOTO 2: Interior view and circuit board.

Photo 2 shows the interior view of the HV diff amp. The shells of the BNC jacks are grounded to the circuit board ground plane. The LED and power-supply wires are twisted to minimize pickup or noise radiation. The two battery holders are on the right side of the chassis, with their battery clips.

CIRCUIT BOARD LAYOUT

I built the circuit on a ground plane PC board. *Photo 2* shows the finished PC board, and *Fig. 2* shows the parts placement. I used low-profile gold-plated 8-pin sockets for the DIP ICs.

The PC board itself was a fairly simple layout, so I routed it with a 0.046″ router bit in my Dremel tool rather than use an etched PC board. This maximized the ground plane area. A few instances required that I wire by hand between components.

Figure 3 shows the top and front view of the chassis with key dimensions, with the panel lettering designations for the front panel. I made a full-size copy of this lettering on drafting appliqué film, which is an adhesive-backed transparent plastic. A coat of light color paint on the enclosure works well with the black photocopy lettering.

The circuit board mounting holes have clearance for a 4-40 screw with flat washer. I used four tapped spacers to mount the board to the chassis. The labels for the top and front panels of the enclosure are shown in *Fig. 4*.

FIGURE 2: Circuit board parts placement.

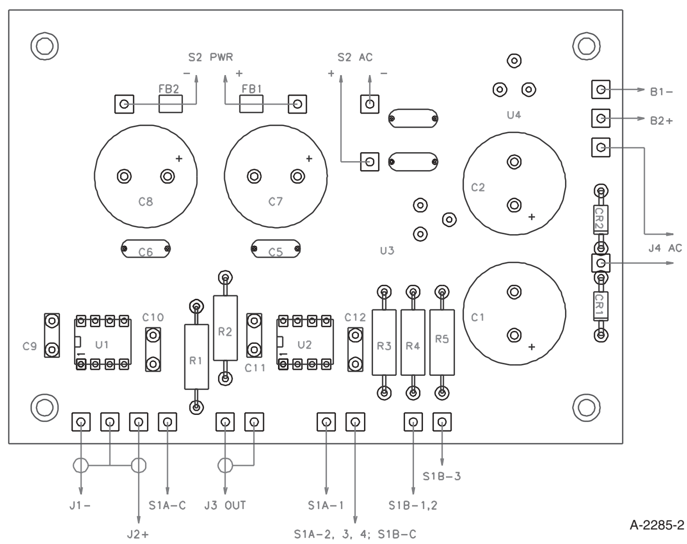

TABLE 2
HV DIFFERENTIAL AMP PARTS LIST

SYMBOL	VALUE	DESCRIPTION	VENDOR	PART NO.	QTY
B1, B2	9V	Battery			2
C7, C8	22µF 50V	Aluminum	Mouser	140–XRL50V22	2
C1, C2	2200µF 35V	Aluminum	Mouser	140–XRL35V2200	2
C3,C4,C9–C12	100nF 50V	Ceramic X7R	Mouser	21RX310	6
C5, C6	100n 100V 10%	Polyester	Mouser	140–PF2A104K	2
CR1, CR2		1N4148	Mouser	78–1N4148	2
FB1, FB2		Ferrite Bead	DK	M2310–ND	2
J1–J3		BNC, Panel Mount	Mouser	523–31–221–RFX	3
J4		Jack 5.5mm male	Mouser	163–5006	1
J5, J6		Battery Snap, 9V	Mouser	121–2224	2
LED1		Red, High Eff.	Mouser	604–L7113SECH	1
R1	604 1%	Metal Film	Mouser	271–604	1
R2	20K0 1%	Metal Film	Mouser	271–20.0K	1
R3	4K22 1%	Metal Film	Mouser	271–4.22K	1
R4	1K40 1%	Metal Film	Mouser	271–1.4K	1
R5	12K7 1%	Metal Film	Mouser	271–12.7K	1
R6, R7	27K 5%	Carbon Film	Mouser	291–27K	2
S1	3P–4 Position	Rotary Sw	Mouser	10YX034	1
S2	DP ON–OFF–ON	Mini Toggle	Mouser	108–MS550H	1
U1	AD629AN	Diff Amp, HV	Newark	see text	1
U2	OP–27EP	Op Amp	Newark	05F8607	1
U3	78T15	Reg, +15V 20W	DK	LM78M15CT–ND	1
U4	79T15	Reg, –15V 20W	DK	LM79M15CT–ND	1
		Batt holders, 9V	RS	270–326	2
		Blank 3.5×5 pcb	DK	PCB–46	1
		Enclosure, 7×5×3	DK	L191–ND	1
	18VAC 0.5A	AC adapter	Mouser	412–218054	1

Drafting appliqué 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 (beldecal@bellsouth.net), or Worth Poly™ polyester laser labels from Worth Data (barcodehq.com).

USING THE HV DIFFERENTIAL AMPLIFIER

Signals applied to input jacks J1 and J2 should use standard X1 oscilloscope probes, and the probe end ground leads

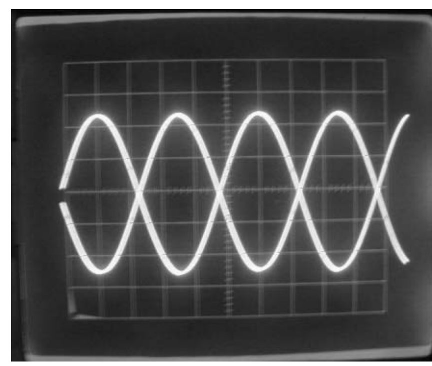

PHOTO 3: Grid signals from tube phase inverter.

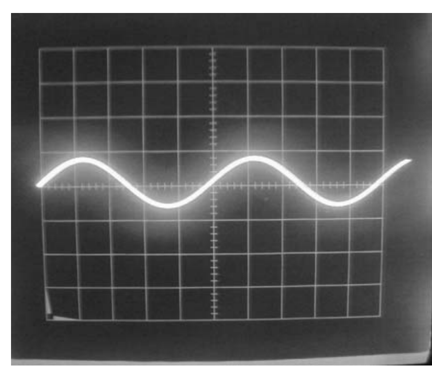

PHOTO 4: Plate voltage using 10× probe.

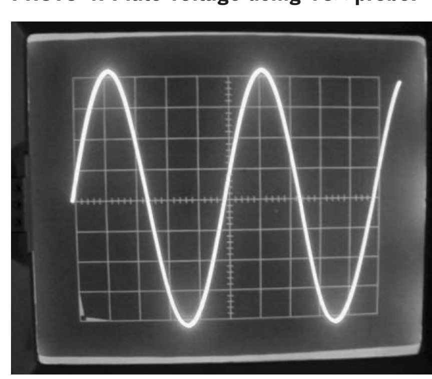

PHOTO 5: Differential voltage plate–cathode using HV diff amp.

should be connected together and kept from coming in contact with any ground. The probe grounds will provide signal shielding by way of the HV diff amp chassis.

You can connect the output jack to any high impedance scope or voltmeter input by BNC terminated coax. When you select any of the reduced gain settings, you must multiply the scope or voltmeter readings by the inverse of the gain reduction. If you read 5V at the $1/5$ gain setting, the actual differential voltage is 25V.

Be sure to keep the common-mode voltage limits in mind when making measurements. High voltages do not appear on the chassis of the HV diff amp, so safety is not an issue as it would be if

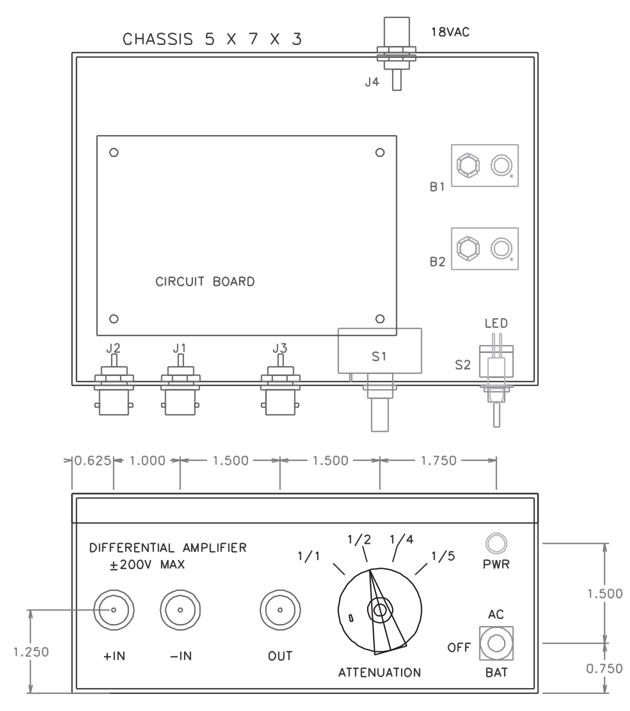

FIGURE 3: Differential amp chassis. A-2285-3

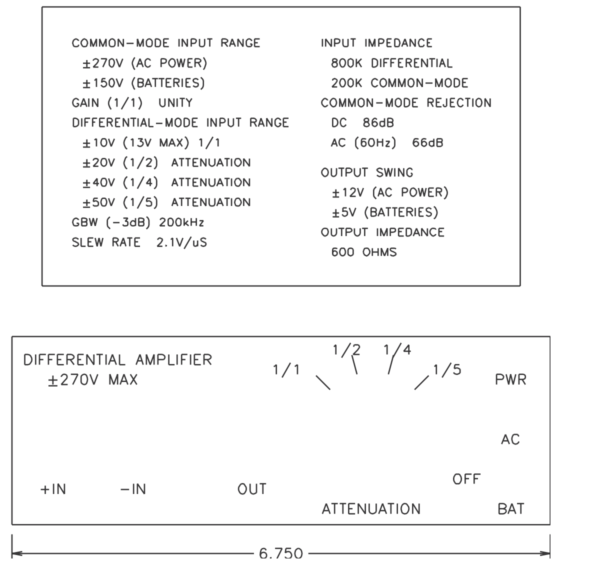

FIGURE 4: Label designations for chassis. A-2285-4

you used an AC cheater plug to "float the scope" in order to make measurements not referenced to ground. This is a very dangerous practice because it places the entire chassis of the scope at the common-mode voltage level where the measurement is being made (the plate of a tube, for instance).

In order to check the performance of the HV diff amp, I breadboarded the input section of a push-pull tube amplifier. I connected a 12AX7 as a direct-coupled split-load phase inverter. I coupled the plate and cathode through 470nF caps to grounded 470k resistors to simulate the grids of the output tubes. *Photo 3* shows the two-channel scope photo of the equal and opposite "grid" signals across the two 470k resistors.

The voltages are 18V RMS, but they are loaded down by the 1MΩ scope inputs. This is a factor you must keep in mind when working with high impedance circuits. Note that this HV diff amp has an even lower input impedance than a scope, so calculation of the amount of attenuation based on parallel impedances may be necessary in high impedance circuits.

Photo 4 shows the phase inverter plate voltage referenced to ground using a 10× scope probe. The AC signal is riding on the nominal DC plate voltage as it swings between 230V and 282V. This AC component represents the voltage that is coupled to one of the output tube grids.

Photo 5 shows the waveform of the plate-cathode voltage at the phase inverter using the HV diff amp. I set the gain to ⅕. The output of the HV diff amp is ground referenced, so you can do what you wish with it. You could get a similar presentation on

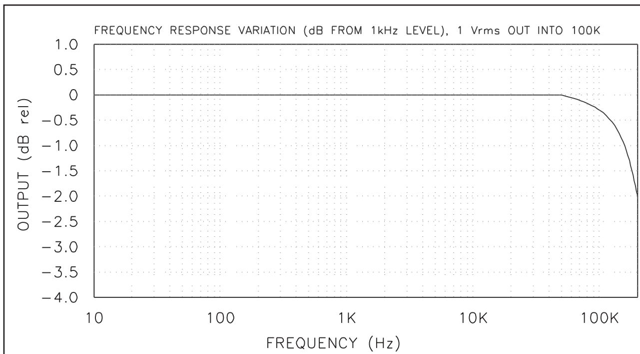

FIGURE 5: Frequency response. A-2285-5

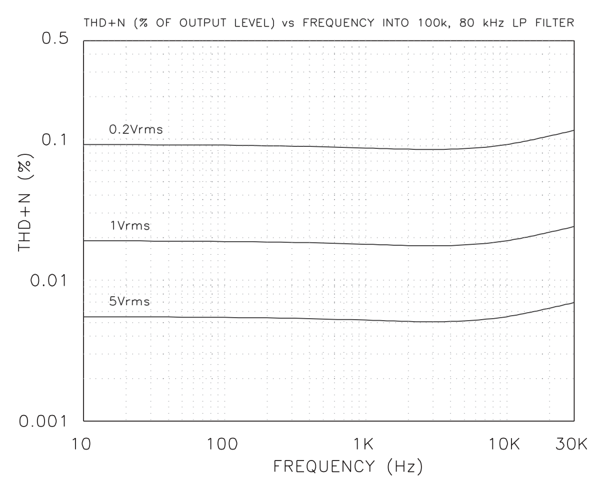

FIGURE 6: THD+N vs. frequency. A-2285-6

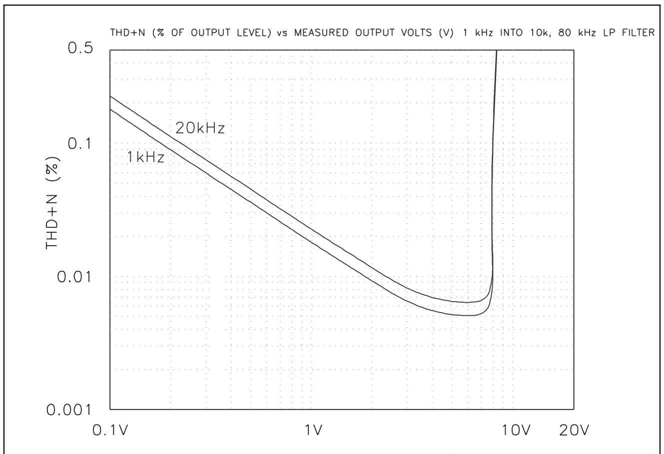

FIGURE 7: THD+N vs. output voltage. A-2285-7

the scope using the 10× probe with AC coupling, or with a differential amplifier plug-in.

However, the signal from the HV diff amp can be sent to a distortion analyzer or other low voltage instrument without concern for the high voltage present in the tube circuitry. I measured a THD+N of 0.45% across the 12AX7. This fairly high level is probably due to the open-loop connection of the 12AX7 circuitry, and would improve in a close-loop power amplifier.

You might think you could use two 10× probes to increase the common-mode voltage even further. However, this will probably seriously degrade the CMRR of the AD629, whose internal resistors can vary as much as 25% from the nominal values. Refer to AB-001 for details concerning the accuracy and CMRR degradation due to external components.

MEASUREMENTS

Figure 5 shows the frequency response of the HV diff amp at the unity gain (1/1) setting, from 10Hz to 200kHz. It is ruler flat from DC out to 50kHz, making it very useful for audio work.

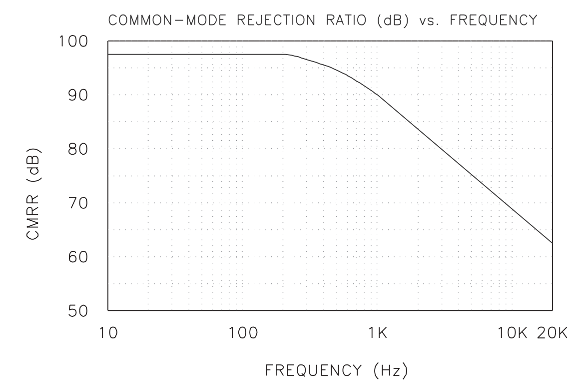

FIGURE 8: CMRR vs. frequency.

Figure 6 shows the THD+N versus frequency for three differential voltages, again at the unity gain setting. The input-referred noise is limited by the Johnson noise of the 380k input resistors in the chip. I engaged the distortion test set 80kHz LP filter to limit out-of-band noise.

The THD+N versus output voltage at unity gain is shown in *Fig. 7*. The THD is a bit higher at 20kHz due to the usual gain-bandwidth limitations.

Finally, *Fig. 8* shows the common-mode rejection versus frequency. Even at 20kHz, there is 65dB CMRR available. This requires that both inputs be at the same impedance, as low as possible to prevent CMRR degradation. ❖

SOURCES

Digi-Key Corp.
701 Brooks Ave. South
Thief River Falls, MN 56701–0677
1-800-344-4539
www.digikey.com

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

Newark Electronics
4801 N. Ravenswood Ave
Chicago, IL 60640–4496
1-800-463-9275
www.newark.com

Radio Shack–Local Store

GFP-565 Preamp Follow-Up Mod

This follow-up to Gary Galo's four-part Adcom GFP-565 remake offers some additional options, including a new line stage op amp and an alternate approach to the external power supply. **By Gary Galo, Regular Contributor**

By now, interested readers will have had a chance to digest my four-part remake of Adcom's GFP-565 preamp.[1] In this follow-up, I offer some additional refinements for that project.

In Part 3, I mentioned the possibility of using Analog Devices AD825AR op amp in the line stage, in place of the AD744KN. After working with the AD825 for some time, I find it to be a worthwhile improvement over the AD744, particularly in the area of inner detail and clarity. By comparison, the AD744 puts a slight veil over the sonic picture, though I would never have described the AD744 in such terms without a direct comparison to the AD825.

The AD825 isn't available in a DIP package, only SOIC. The "AR" version is eight-pin, so you can use the same Aries adapter that I recommended in Part 2 of the series (*Photo 1*). Note that the output and local feedback for the AD825AR *must* be taken from pin 6.

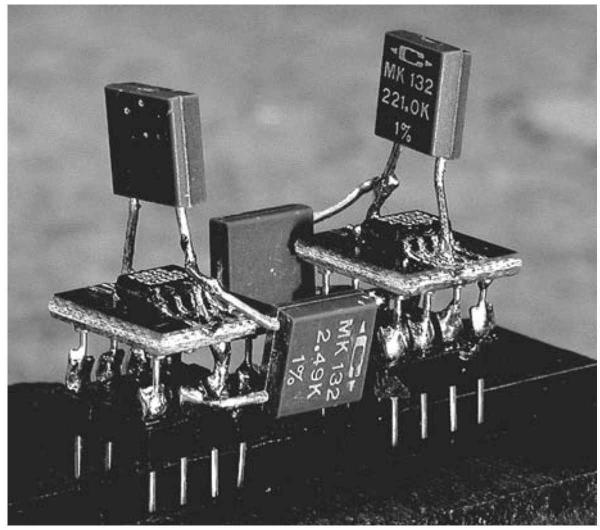

PHOTO 1: Two views of the AD825AR op-amp modules using Caddock MK132 resistors. The SOIC chips must be soldered to an Aries DIP adapter. Otherwise, construction is similar to Photo 28 in Part 3 of the original series.

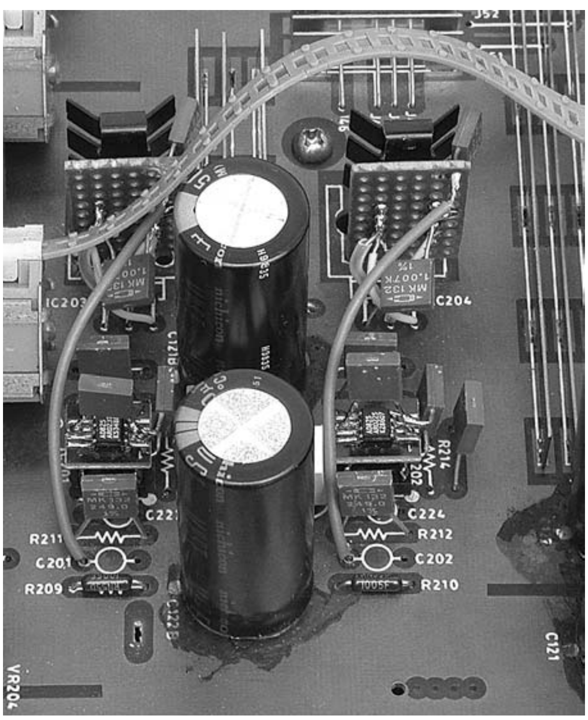

PHOTO 2: The completed line stage using AD825AR op amps and Caddock MK132 resistors.

- Solder the AD825AR to an eight-pin SOIC-to-DIP adapter, just as you did during construction of the regulators in Part 2. Note that pin 1 is on the side with the beveled edge. You can download a datasheet for the AD825 from www.analog.com for further clarification.
- Cut off pin 2 of the SOIC-to-DIP adapter.
- Solder the SOIC-to-DIP adapter to a Tyco eight-pin plug adapter, except for pin 2. You can also use the Aries 8-pin DIP header in the parts list (*Table 1*), but I prefer the gold-plated Tyco because it's easier to solder.
- Solder R251 (R252)—221k—between pins 2 and 6 of the SOIC-to-DIP adapter, on top of the adapter.
- Solder R253 (R254)—2.49k—between pin 2 of the component carrier and the resistor lead already soldered to pin 2 of the SOIC-to-DIP adapter.
- Solder the two assemblies in the IC203 and IC204 footprints. Pay careful attention to orientation, which is the same as the original op amps.

The completed line stage with the AD825AR and Caddock MK132 resistors is shown in *Photo 2*. Measurements will be essentially the same as those given in Table 1 of Part 3. Output DC offset will normally be 5mV or less. Typical input offset voltage for the AD825AR is 1mV, with a maximum of 2mV. I have never encountered anything close to the worst-case situation (10mV in this gain-of-five line stage) with the AD825.

ALTERNATE EXTERNAL SUPPLY

In Part 1, I mentioned an alternate approach to the external power supply.

This external supply houses the AC line filter, power transformer, dual rectifier bridges, DC common-mode chokes, and raw DC filter capacitors in an external chassis. *Figure 1* shows the schematic for alternate supply and interfacing with the main preamp chassis. Anyone who has made it through the four parts of the original series probably won't need step-by-step instructions for building this supply, so I'll offer some general guidelines.

- The Bud 6 × 5 × 4″ minibox recommended in Part 1 is too small for the alternate supply. I recommend the Sescom MC-8A metal cabinet, which measures 7 × 6 × 4″. I opted for the black cabinet. The anodizing can prevent a good ground connection between the various pieces of these modular enclosures. I used small ground lugs, slip-on crimp connectors, and 18AWG hookup wire to ensure a solid connection between the six pieces.
- You can re-use the four HexFred rectifiers installed in the main preamp chassis in Part 2, plus the four sets of R/C snubbers. You'll need four more of each for the new supply.
- I mounted the rectifiers, R/C snubbers, DC common-mode chokes, and raw-DC filter capacitors on a piece of Circuit Specialists RF Prototyping Board (*Photo 3*). Use 16AWG solid bus wire to make connections on the board. You can use the same type of bus wire you used to connect the regulator board in Part 2—just remove

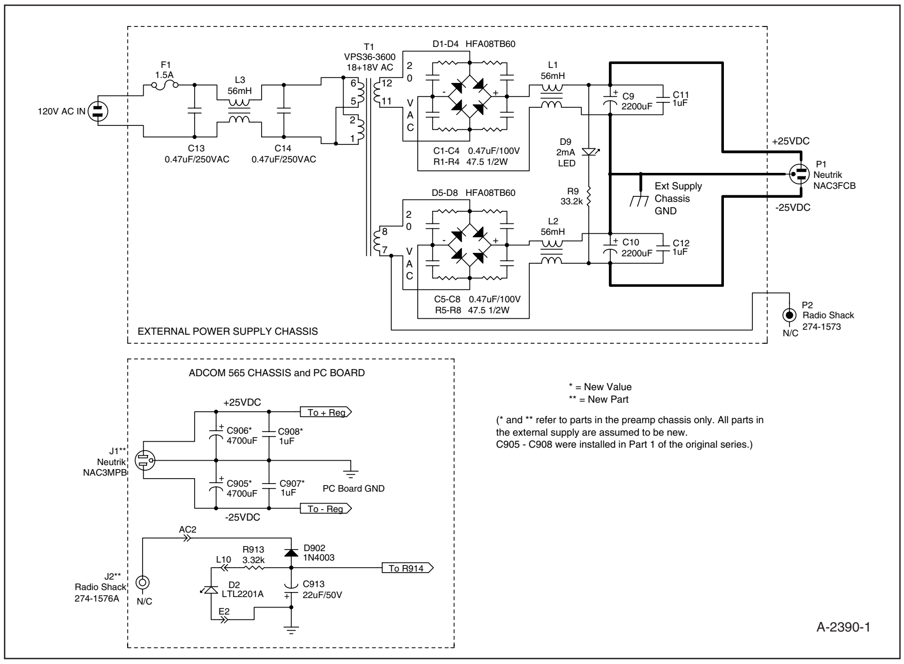

FIGURE 1: Circuit diagram of the alternate external supply and the connections to the main preamp chassis. This supply houses the AC line filter, power transformer, dual rectifier bridges, common-mode chokes, and raw DC filter capacitors in the external chassis.

the insulation. Slip a piece of ⅜″ heatshrink tubing over each HexFred rectifier diode. You don't need to shrink the tubing—it will be a snug fit without shrinking.

- You can re-use the 80VA power transformer recommended in Part 1, or try the Parallax/Triad 130VA transformer shown in *Fig. 1*. I found that a 130VA transformer makes a noticeable improvement in dynamics with this preamp. As I indicated in Part 1, the 130VA transformer has higher secondary voltages under the load encountered by this preamp. But, having separate rectifier bridges for the positive and negative supplies adds an additional forward drop for each half of the cycle. This eliminates the potential problem of excessive regulator heat due to higher raw DC voltages.
- The AC line filter built for the outboard supply in Part 1 can be re-used. I used a 56mH choke for the new external supply.
- Three-conductor, 14AWG SJT cable is fine for the raw DC cable to the preamp. I chose D.H. Labs Q10 speaker cable, which is Teflon-insulated, silver-plated wire. The Q10 cable is

THE GALO-MODIFIED GFP-565 PREAMPS—A SONIC EVALUATION BY LORELEI MURDIE

The three preamps auditioned by Lorelei Murdie were not identical. Preamp #1 is the unit I modified for Victor Campos, and the same one auditioned by Ed Dell for his report in Part 4 (Feb. 2004). It contains the outboard power supply described in Part 1 (Nov. 2003) of the series, AD744KN op amps in the line stage, with the output and feedback taken from pin 6, and Holco resistors in the line stage. Preamp #2 is my own reference preamp. It has the alternate outboard supply described in the accompanying article, with the AD825AR op amps and Caddock MK132 resistors in the line stage. Preamp #3, my secondary preamp, is identical to #1 except for AD825 op amps in the line stage. Lorelei's own preamp is an Adcom GFP-565 with one minor modification: over 12 years ago I replaced the LT1056 op amps in the line stage with Linear Technology's LT1122. This was an incremental, rather than a dramatic, improvement.—G.G.

REFERENCE RECORDINGS:

(All listening was done using compact discs).

Moussorgsky/arr. Ravel: *Pictures at an Exhibition*. Chicago Symphony Orch., Fritz Reiner, cond. JVC JMCXR-0016 (XRCD2).

Rimsky-Korsakov: *Scheherazade*; *Stravinsky: Song of the Nightingale*. Chicago Symphony Orch., Fritz Reiner, cond. RCA Victor Living Stereo 68568-2 (UV22-Encoded Gold Edition).

The French Touch. Boston Symphony Orch., Charles Munch, cond. RCA Victor Living Stereo 68978-2.

Verdi: *Requiem*. Joan Sutherland, Marilyn Horne, Luciano Pavarotti, Martti Talvela, Vienna Philharmonic Orch., Georg Solti, cond. London 411 944-2.

Honkytonkville. George Strait. MCA Nashville. B000011402

Red Dirt Road. Brooks & Dunn. ARISTA. 67070-2

LISTENING CRITERIA:

1. **Comparisons:** When listening to each preamp, my husband Randy and I were mostly comparing the new preamp to our very slightly modified preamp, which we have used for 14 years. While previewing these preamps, there were times—especially with the classical CDs—that I thought about the natural sound of the instruments. As a concert hall manager in a complex that supports over 300 events per year, I hear so much live music that it is hard for me not to compare this sound as well. For example, I know the sound of a set of cheap tympani and the sound of a set of professional-grade timpani. I also know how a piano sounds under the hands of Leon Fleisher versus a college freshman piano major.
2. **Soundstage:** For my own enjoyment, I like to close my eyes while listening to music and pretend I'm in the concert hall listening to the music live. The clarity of the soundstage, width, and depth make a big difference to my listening pleasure.
3. **Volume:** We listened to every CD with the volume control pointing to ten o'clock. We rarely listen to music over that level. When listening at 10:00, it is difficult to have a conversation (especially when listening to a country music CD). I think ten o'clock is as loud as our system can go without distortion.
4. **Dynamics**: When listening to live music, you realize that sound level is never dynamically flat. There are always subtle crescendos and decrescendos being played as well as the more obvious dynamics when the piece as a whole becomes louder and softer. To hear them more fully on a system increases the listening enjoyment and makes the recordings sound more musical and natural.
5. **Detail & Definition**: This includes hall acoustics, articulation, low-end punch and cleanliness, high-end brightness, and how well the instruments sound together.
6. **Overall**: Warmth, tone quality, and tonal balance can make a big difference to listening enjoyment and naturalness of sound. For example, if the sound level of the bass, midrange, and treble are not similar and the sound system favored one over the others, the recording would not be as natural.
7. **Our system:** Vandersteen 2ci Loudspeakers, Cambridge Audio D500 CD Player, AudioQuest speaker cables, D.H. Labs BL-1 interconnects, Adcom GFA-545 power amplifier.

LISTENING TO THREE PREAMPS

PREAMP #1

We have been lamenting the fact that we are using our back-up power amp (an Adcom GFA-545). Our old reference amp (an Adcom GFA-585) bit the dust a few months ago. Adding this preamp (#1) to our system has changed the way we feel about our system as a whole. We were no longer lamenting our plight but are excited to listen to our recordings again. Our CDs sound more like our old system and in many ways even better!

This preamp has much more detail. In Solti's Verdi *Requiem* (this happens to be one of my favorite recordings, which I have listened to many, many times), there is a section at the end of the "Lacrimosa" where the basses are singing low notes with the orchestra. I have never heard the bass drum playing along with the men the way I heard it with this preamp. I remember being surprised to hear this with such clarity and definition. The soloists also stood out more and their articulation was more pronounced.

In Reiner's *Pictures at an Exhibition* the first thing I heard was the sound of the recording equipment. I had never heard that before in this CD even with our old reference amp. The hall ambience was more pronounced and woodwind sound was warmer. The trumpet sound at the beginning was cleaner and clearer. The sound stage opened up with this preamp, and with greater depth. What I kept hearing over and over again—and I think what impressed me the most—were the dynamics.

All of my recordings in general sounded more musical with this preamp, which greatly increased my listening pleasure. The subtle crescendos and decrescendos were much more pronounced. The low end was tighter and more prevalent. We were missing this with our GFA-545 because it doesn't have the low-end punch and control of the GFA-585. Also, I noticed that the sound level in general was louder at ten o'clock than what we were used to. This is surprising, since Gary says that the modified preamp has less than half the gain of the Adcom original.

My husband listened to this preamp for five minutes before saying, "How do we get one of these?" He was listening to his favorite rather stiff and more difficult to work with than SJT. This cable has four conductors, two 14AWG, and two 16AWG. I tied the 16AWG conductors together for the ground wire. I suggest limiting the length of this cable to two feet. The power supply can be situated on a shelf below the preamp.

- Use a Neutrik PowerCon NAC3FCB cable connector, which mates with the NAC3MPB mounted on the preamp chassis. These are the blue/gray Neutrik connectors. Since I used the all-blue Neutrik pair for the AC connections in Part 1, I decided to use the blue/gray type for DC. They aren't mechanically interchangeable, which prevents accidental connection of the

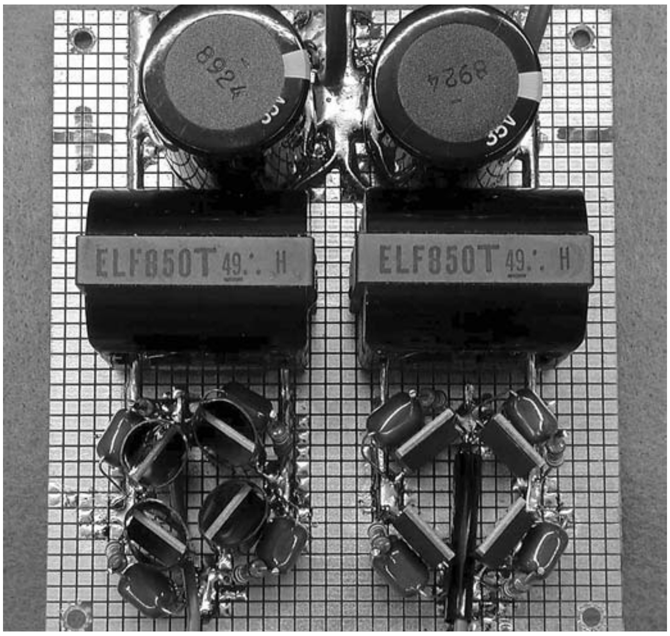

PHOTO 3: The dual rectifier bridges, R/C snubbers, DC common-mode chokes, and raw DC filter capacitors are mounted on a piece of Circuit Specialists RF Prototyping Board.

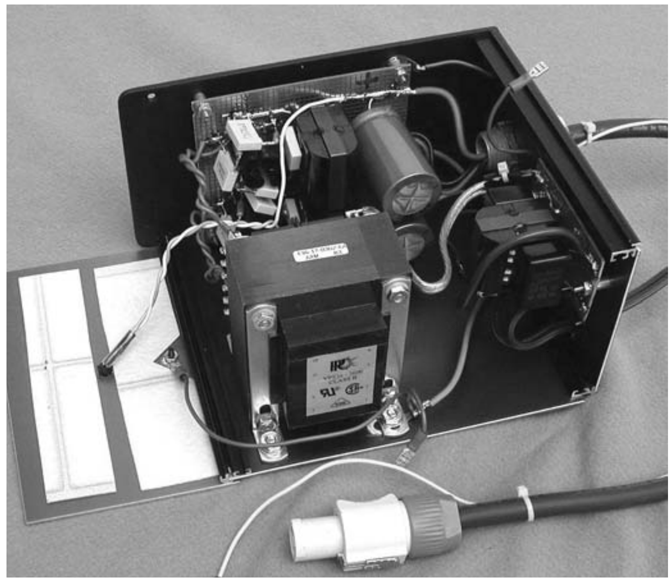

PHOTO 4: Inside view of the external DC supply. The power transformer is in the foreground, with the rectifier board behind it, mounted on the side panel of the chassis. The AC line filter and line fuse are mounted on the rear panel. A separate 18AWG wire provides the connection to the muting circuit bias supply.

wrong type of supply. I have two preamps, one with each type of outboard supply, so I don't want to accidentally mix them up. You could re-use the all-blue pair

PHOTO 5: Rear panel of the external supply, showing the main DC cable, bias supply wire, AC fuse holder, and AC power cord.

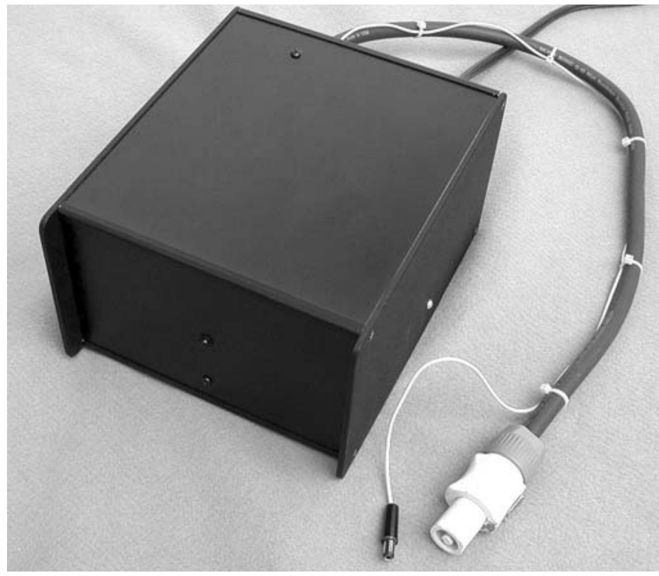

PHOTO 6: The assembled external DC supply. Nylon cable ties are used to bundle the bias supply wire and the main DC cable.

recording of George Strait. Don't cringe just yet.

Now, I realize that it is very hard to reference the sounds, as no one really knows exactly what it is supposed to sound like. We did attend a George Strait concert and sat in the third row last year. I remember having my fingers in my ears most of the time because it was so loud. I also remember commenting to my husband that it sounded better at home.

Anyway, my husband said this preamp made George sound even better. He enjoyed hearing instruments in the recordings that weren't as noticeable with our old preamp. The background instruments stood out more with this preamp. His comment was that he could hear more detail and clarity, especially in the soundstage.

PREAMP #2

If our old preamp was a Chevrolet, preamp #1 would be a Cadillac, and this preamp would be a Rolls Royce. We were blown away by the sound of our system with this preamp installed. Everything we played sounded great. Our system has never sounded this good.

We have a few CDs (most of them are in my husband's collection) that I considered unplayable on our system at 10:00. I think they were recorded for cheap CD players and the recordings are "hot" or very bright. This preamp made them sound good!

One such example (and I don't recommend that you go right out and buy this CD) is a country CD of my husband's called "Red Dirt Road" by Brooks and Dunn. It is so bright that it hurts to listen to it at ten o'clock. Your ears fatigue quickly listening to this recording with our old preamp, and to a lesser degree but still problematic with preamp #1. With this preamp, the whole system was able to play this recording and it sounded good (well, if you like country music)! My husband's comment about this preamp was that even though he had been listening to these same country music recordings many, many times, he believed he was listening to different recordings now because with this preamp #2 he could hear things he had never heard before.

Listening to *The French Touch*, I was impressed by the volume of sound at ten o'clock. This preamp is louder than preamp #1 and without distortion. It is so smooth sounding. The low end is amazing. You get more low end but it is controlled and tight. The balance of high, mid, and low end is wonderful. You don't get a sense that any of them overpowers the other. You just get more.

While listening to Verdi's *Requiem*, I couldn't help but notice how much more I could hear the singer's articulation. The soundstage with the preamp is bigger and has much more depth than even preamp #1. This is a most impressive preamp. We did not want to see it go. I couldn't believe how wonderful our system sounded even with our backup power amp.

PREAMP #3

Well, I'm sorry to say that we heard preamp #3 after preamp #2, because all we could think about was how wonderful preamp #2 sounded. I would say that this preamp is slightly better than preamp #1, but a far cry from preamp #2. I noticed that the sound is a little warmer than preamp #1. There is a slightly larger and deeper soundstage than preamp #1.

To give an example, compared to preamp #1, the trumpet sound in *Pictures* is slightly better (fuller and warmer), and I noticed that the snare drum sounded tighter and clearer than with preamp #1. This is a wonderful preamp similar to preamp #1. I would say it is 25% better than preamp #1, but about 75% away from preamp #2.

OUR PREAMP

It has been three weeks since we last heard our preamp. Our first reaction to the switch back was, "The grunge is back!" The sound wasn't as clean. The soundstage was smaller and shallower. The volume level wasn't as great at ten o'clock.

Also, I missed not hearing the subtle crescendos and decrescendos of the borrowed preamps. This preamp just doesn't have the musicality of the other preamps. We're back to lamenting our fate and lack of our reference power amplifier. We both considered it a privilege to hear these preamps and thank Gary Galo for sharing them with us.

NEW POWER AMPLIFIER

Another three weeks have passed. We purchased a new power amplifier—an NAD S200. It is wonderful (and at $1750 plus tax it should be). Our system has power again. We hear wonderful dynamics, more low and high ends, larger soundstage, and more detail.

I asked my husband last night—after listening for four to five hours each night for three days—one question: "You have heard our system with preamp #2 and our backup power amplifier. You have now heard our system with our unmodified Adcom preamp and NADS500 power amplifier. If you had to choose between the two systems, which system would you rather listen to?" My husband's response after a brief pause was, "I'd rather listen to preamp #2 and our backup power amplifier." I agree with him. We both decided (independent of the other) that preamp #2 patched into our system just sounded better and was a more enjoyable listening experience! Bravo Gary Galo!

listed in Part 1, if you like.

- The connection for the muting circuit bias supply must be made with a separate wire connected to the power transformer (*Fig. 1*). I used a Radio Shack 274-1573 cable-type DC power connector for the external supply, which plugs into their 274-1576A chassis connector. Note that only the center pins of these connectors are used—there isn't a separate ground connection for this supply. If the muting circuit bias supply is not connected, the preamp will be in a permanent state of mute.
- *Photo 4* shows the inside view of the external supply. In his sidebar to Part 4, Ed Dell correctly noted a slight mechanical buzzing in the case of the external supply. I suggest Armstrong self-stick floor tile to damp chassis vibration from power transformer, and I placed a piece of 1″-thick foam between the top of the transformer and the chassis top plate. This pretty well eliminates audible buzzing from the transformer.

After the external supply is completed, you should test it before connecting it to the preamp. Raw DC at the supply output should be ±27V with no load. The 2mA LED is the pilot lamp for the external supply. It will also drain the raw supply if it is powered up without being connected to the preamp. Discharge takes about two minutes after AC power is disconnected. *Photos 5* and *6* show two views of the completed external supply.

Now, a few comments about the main preamp chassis modifications:

- No rectifiers, snubbers, or DC common-mode chokes are mounted inside the preamp chassis. They have all been moved to the external supply.
- Mount the Neutrik PowerCon chassis connector between the fuse holder and the AC line cord, as in Part 1. Mount the Radio Shack DC power connector for the bias supply next to the switched, fused AC outlet on the GFP-565 preamp chassis (*Photo 7*).
- Use 14AWG wire—or individual conductors from the D.H. Labs Q-10 speaker cable—between the Neutrik chassis connector and the main PC board. If you used the D.H. Labs cable, parallel the two 16AWG conductors for the ground connection (*Photo 8*). *Carefully observe DC supply polarity!*
- Connect the muting circuit bias supply with a short length of 18AWG hookup wire between the Radio Shack DC chassis mount connector and the AC2 hole on the main preamp PC board (*Photo 9*).

After completing the preamp chassis modifications, recheck all wiring, making sure that you have DC polarities correct from the external supply to the main preamp PC board.

Connect the new external supply to the preamp, power up, and recheck the raw DC voltages and all regulator volt-

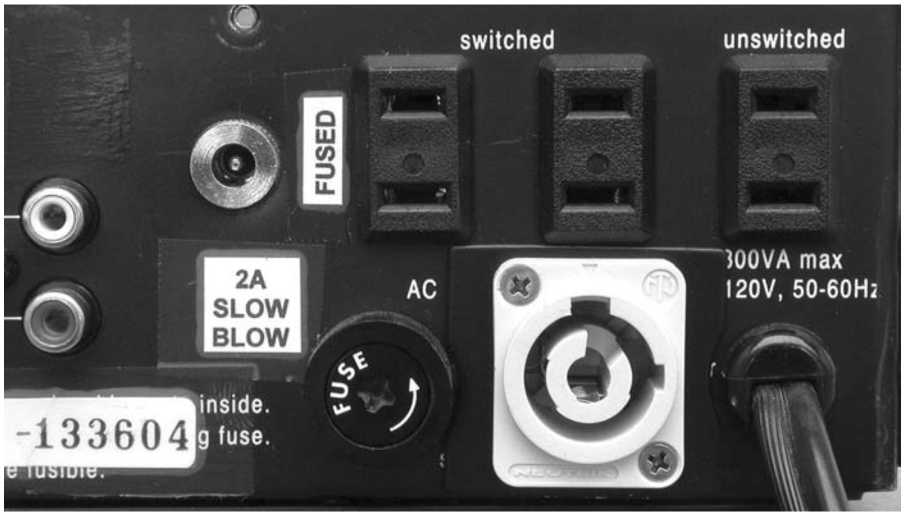

PHOTO 7: Rear panel of the GFP-565 preamp modified for the alternate external supply. The raw DC connections are made with the Neutrik PowerCon chassis connector mounted between the AC line cord and the fuse holder. A Radio Shack DC chassis type power connector, mounted to the left of the switched and fused AC outlet, is used for the muting circuit bias supply connection.

ages, as described in Part 2. Under load, the raw DC voltages will be ±25V. If the preamp pilot LED glows dimly, the muting circuit bias supply probably isn't connected. When you install the preamp in your system, plug the AC line cord from the external supply into the switched, fused outlet on the back of the preamp, or directly into a switched power line filter.

TABLE 1

PARTS LIST

Line Stage Op-Amp Change

DESCRIPTION	QUANTITY	SOURCE	NOTES
Analog Devices AD825AR OpAmps	2	Analog Devices AD825AR Digi-Key AD825AR-ND, Newark 83F3403 (IC201, IC202)	
Aries 8-pin SOIC/DIP Adapters for the AD825AR Op-Amp	2	Digi-Key A724-ND or Accutek Microcircuit AK08D300-NSO	
Tyco 8-pin Plug Adapter Aries 8-pin DIP Header	2	Allied 905-3114 or Digi-Key A101-ND	
221k (R251, R252)	2	(Re-use from AD744 modules; see Part 3)	
2.49k (R253, R254)	2	(Re-use from AD744 modules; see Part 3)	
ALTERNATE EXTERNAL POWER SUPPLY			
Sescom MC-8A Metal Cabinet	1		
Parallax/Triad VPS36-3600	1	Allied 967-8031 (T1)	18+18 VAC/130VA
or Parallax/Triad VPS36-2200		Mouser 553-VPS362200 (T1)	18 + 18V AC/80VA
International Rectifier HFA08TB60 HexFred Diodes	8	Digi-Key	HFA08TB60-ND (D1-D8)
47.5Ω, ½W resistor Vishay-Dale CMF Type RN60	8	Mouser	71-RN60D-F-47.5
or Roederstein Resista			MK3 (R1-R8)
0.47µF, 100V DC capacitor Xicon MEB	8	Mouser	1430-1474
or Panasonic V-series		Digi-Key	P4733-ND (C1-C8)
Panasonic V-series Line Filter, 56mH, 1.1A	3	Digi-Key	PLK1017-ND (L1, L2, L3)
Nichicon 2200µF, 100V DC Nichicon KG-Gold	2	Michael Percy Audio	(C9, C10)
1.0µF, 50V DC Mallory 168-series	2	Newark	89F1692 (C11, C12)
Panasonic ECQ-UV, 0.47µF, 250V AC	2	Digi-Key	P4614-ND (C13, C14)
1.5A Fast-Blow Fuse	1	Radio Shack	270-1006 (F1)
Panel-mount fuse-holder	1	Digi-Key Radio Shack	283-2344-ND, 270-367
Panasonic 2mA Red LED	1	Digi-Key	HLMP4700-ND (D9)
Yageo 33.2k ¼W Metal Film Resistor	1	Digi-Key	33.2KXBK-ND (R9)
Neutrik PowerCon NAC3FCB Cable Connector	1	Mouser MCM Electronics	568-NAC3FCB, NAC3FCB (P1)
Radio Shack 274-1573 In-Line DC Power Connector	1		5.5mm O.D. × 2.5mm I.D (P2)
MISCELLANEOUS			
16AWG, 2-cond. AC Power Cord			
Strain Relief for above		Mouser	561-MP4N4
3-cond. 14AWG SJT wire or D.H. Labs Q-10 Speaker Cable			
Strain Relief for above		Allied	534-9176
18-AWG hookup wire			
Strain Relief for above		Mouser	561-MP3P4
16 AWG solid bus wire		Digi-Key	A3057B-100-ND
Prototyping Board, Circuit Specialists IF-RFB			
⅜″ Heat Shrink Tubing		Radio Shack	
PREAMP CHASSIS/PCB PARTS:			
Neutrik PowerCon NAC3MPB Chassis Connector	1	Mouser MCM Electronics	568-NAC3MPB, NAC3MPB (J1)
Radio Shack 274-1576A Panel-Mount DC Power Connector		5.5mm O.D. × 2.5mm	I.D (J2)
MISCELLANEOUS			
3-cond. 14AWG SJT wire or D.H. Labs Q-10 Speaker Cable 18-AWG hookup wire			

A while back, I asked Lorelei Murdie, my audiophile friend and colleague at The Crane School of Music, SUNY Potsdam, whether she would be interested in listening to three of the modified preamps and offering her observations in print. Since she has owned an Adcom GFP-565 preamp for 15 years, I thought she would be a good person to offer a third-party evaluation. Lorelei is the facilities manager at Crane, a musician by training, and, like myself, she hears live, unamplified music nearly every day. She has a very revealing audio system, and great ears. Her comments appear in the accompanying sidebar. ❖

REFERENCE

1. Galo, Gary, "Adcom's 565 Preamplifier" Parts 1 through 4, *audioXpress* 11/03, 12/03, 1/04, and 2/04.

SOURCE

MCM
650 Congress Park Drive
Centerville, OH 45459
1-800-543-4330 (Voice)
1-800-765-6960 (FAX)
www.mcmelectronics.com
(All other vendors are listed in Parts 1 and 3 of the GFP-565 preamp series)

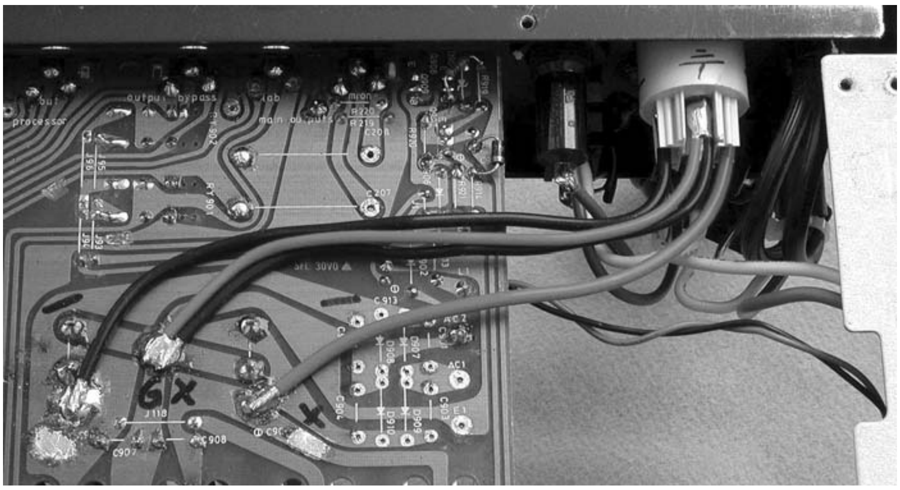

PHOTO 8: Inside the preamp, bottom view, showing the DC supply connections to the main PC board.

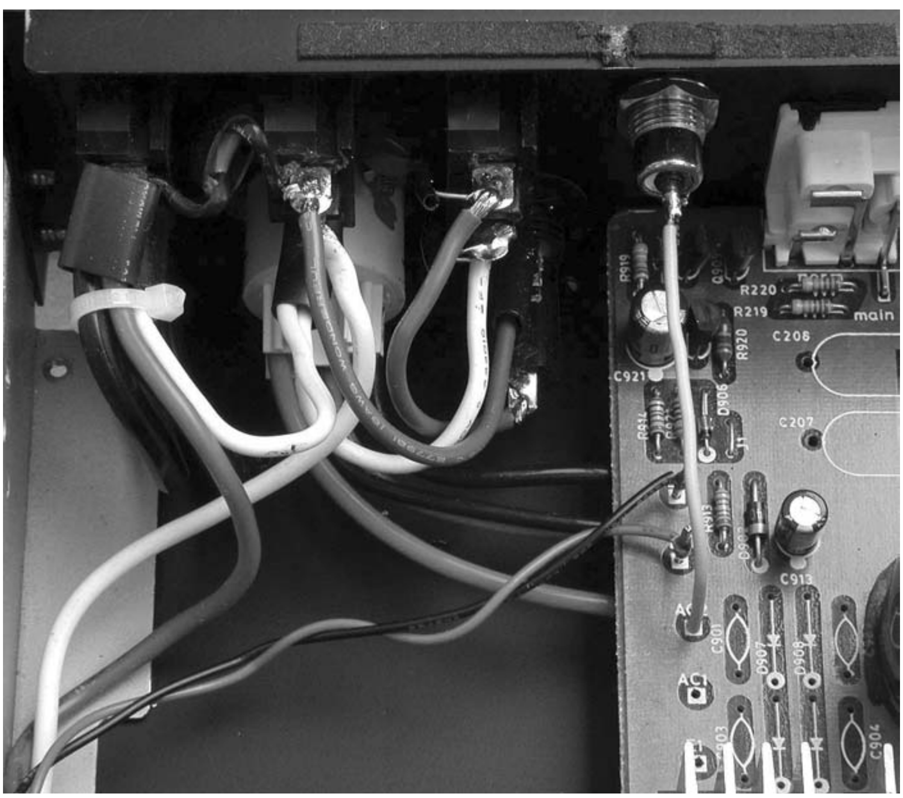

PHOTO 9: Inside the preamp, top view, showing the muting circuit bias supply connection to the AC2 hole on the main PC board.

Rebuilding a Classic: Heath's W-5M

With these modifications and parts replacement, you can bring this sweet-sounding amp to life.

By Bruce W. Brown, RPh.

Most *audioXpress* readers will recognize the Heath W-5M amplifier, which Heath first sold in the 1950s as a high-quality kit for the "audiophile," a new term for that time. The W-5M was a Williamson-type amplifier built around a very high quality Altec/Peerless output transformer, utilizing KT-66 output tubes.

I have purchased a number of W-5s over the years and am always amazed at how wonderful they sound. With careful upgrading and modernizing, they rival any tube amplifier manufactured today. You can perform this rebuild for less than $65 per amp, and with some new tubes it can save you thousands over new equipment—plus you get the enjoyment of doing it yourself.

PHOTO 1: The rebuilt Heath W-5M amp.

CIRCUIT

The basic circuit is easy to follow (*Fig. 1*). The signal is coupled to one half of the 12AU7 grid and amplified. It is then directly coupled to the grid of the second half of the first 12AU7, which acts

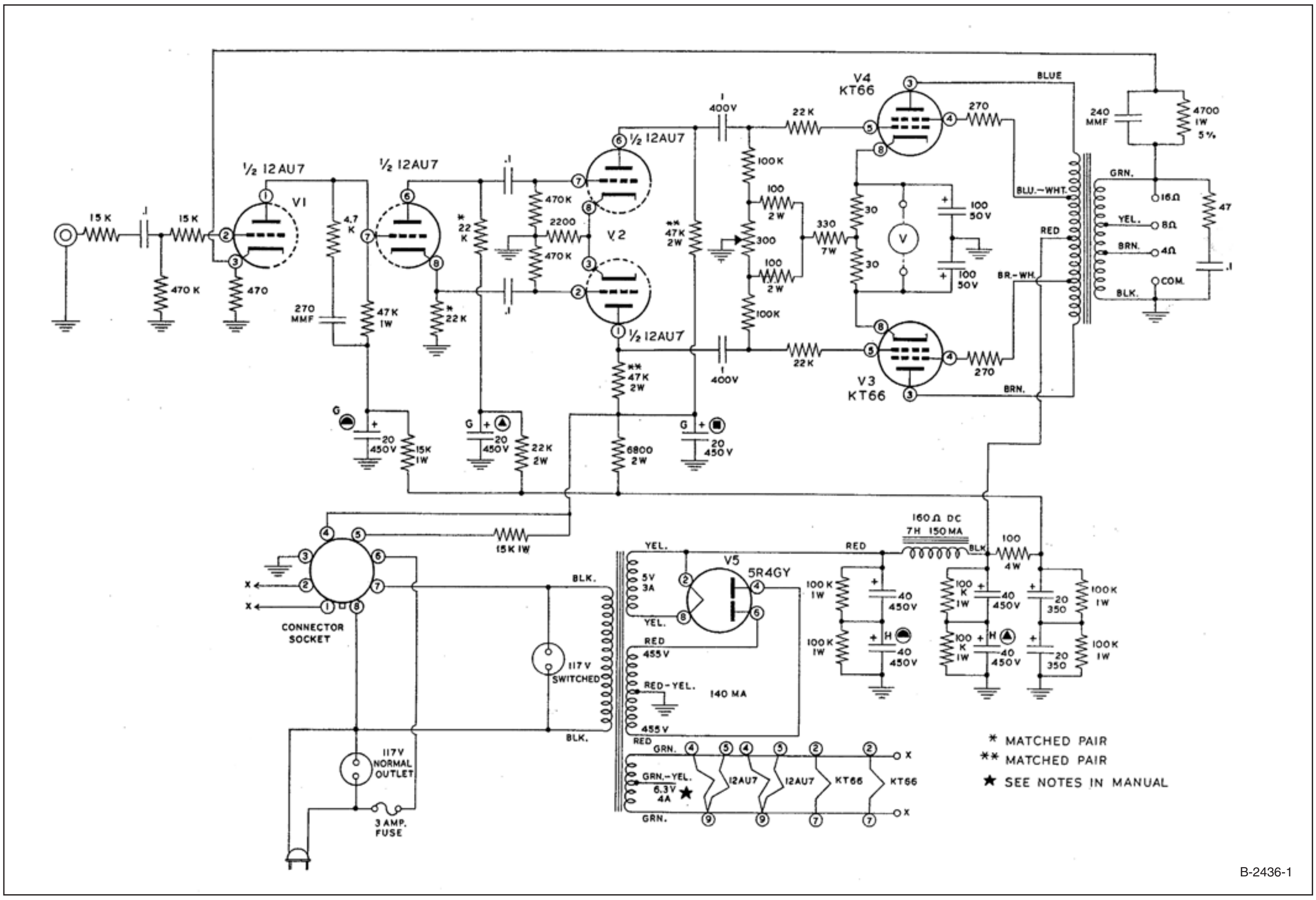

FIGURE 1: Heathkit amp schematic.

ABOUT THE AUTHOR

Bruce Brown is a registered pharmacist who works in the medical research area for a major pharmaceutical company. He has been experimenting with electronics for over 30 years, remaining actively interested in electronics, building kits, and "home brew" audio. Online auctions have stimulated new interests. He has just recently become interested in restoring vintage audio equipment and writing articles to assist other hobbyists. His wife, Kristin, has been invaluable help with editing his ramblings. He currently uses many restored and homebuilt audio amplifiers and welcomes communication about vintage equipment and restorations.

as a split-load phase inverter. The signal at the cathode follows the grid while the plate voltage swings in the opposite direction. The cathode and plates are coupled to the grids of the second 12AU7, and act as a push-pull driver stage through a pair of 0.1µF capacitors. The amplified signals are coupled to the grids of a pair of KT-66/6L6 tubes through a pair of 1µF capacitors.

Feedback is applied from the secondary of the output transformers back to the cathode of the first 12AU7 amplifier stage to reduce distortion and improve frequency response. The output tubes employ a circuit to balance the plate current, reducing harmonic distortion. This "balancing" circuit utilizes a simple voltmeter to perfectly balance the output tubes, which allows the use of unmatched output tubes. The resistor-capacitor string across the output transformer secondary provides high-frequency stability and is referred to as a "tweeter saver."

FEATURES

Everything about this amplifier (*Photo 1*) is conservatively designed. The 5R4 rectifier tube is operated well below its current and voltage ratings, and the filter capacitors are "stacked" so that they never approach their maximum ratings. Heath emphasized this in their manuals as a significant advantage compared to equipment of other audio manufacturers.

Since the actual voltage seen by the filter capacitors is around 500V, the use of this stack makes available 900V capacity, which might explain why so many of these amps continue to function 40–50 years after being built. Heath could have used 525V can caps, as Dynaco did in the Mark IIs and IIIs, but instead chose to be very conservative. I have used a Variac® on a number of W5s not used in 20 years, and after a gentle wakening, they have functioned perfectly, with no hum or noise.

The W5s used a heavy, potted power transformer, which supplies 140mA of high-voltage current—3A per filament for a tube rectifier and 4A for the other filaments. It also has a 7H potted choke. Even though the total capacitance of the power supply is not huge, these amps are typically hum and noise free. The output transformer is one of two different

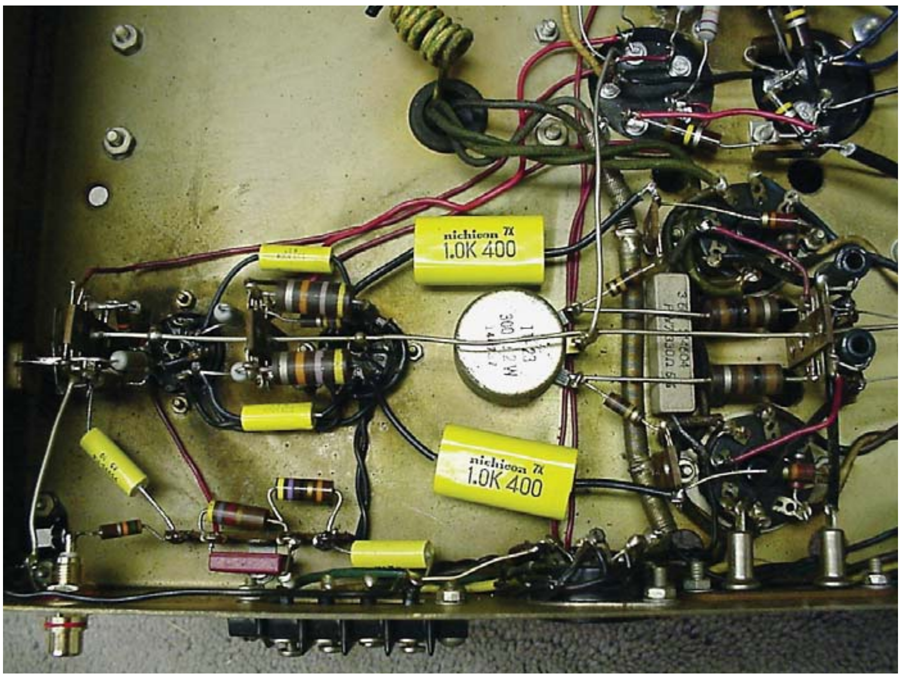

PHOTO 2: Rebuilt amp section.

Altec/Peerless ultralinear potted units. The earliest amps used the larger Peerless 16458, while those sold after 1957 used the Peerless 16309 transformer.

In comparison with the power transformer, either one of these outputs seems small. They are, however, very sweet and almost "bulletproof." There is debate about which is better, but I believe the difference is inconsequential. Either way these amps are very well designed and conservatively constructed. They are typically rated at 25W output, with an average maximum of 32.4W and a peak of 47.2W. The earlier amps also had a "surge protector" in the AC line, a bimetallic strip heated by an exposed wirewound resistor.

Other significant specs:

- Frequency response of 10Hz–100kHz ±0dB at 0.25W (normal listening level)
- Harmonic distortion typically 0.05%–0.1%
- Intermodulation distortion at <0.1% at 20W
- Hum and noise of 80.2dB below 250mW (normal listening level), 84.2dB below 5W (loud listening level), and 99dB below 25W
- Damping factor 40

The construction manual that came with the kit was well written and illustrated, and allowed even the most inexperienced builder to construct a superb amplifier. In addition, the manual contains many test graphs, charts, and oscilloscope pictures. I highly recommend that before you start to rebuild one of these amps, you obtain a copy of the original construction manual. I have found them available on various online auction services for under $15. I have had good results with WF6G Vintage Manuals (www.w7fg.com) as a supplier of high-quality manuals.

Since many of these amps are 50 years old, it is time to update and rebuild them to last for another 50 years. I will detail a typical rebuild and update. My discussion will be broken down into two parts: the amplifier section and then the power supply.

Although it is very important to have a copy of the manual, I am including a schematic (*Fig. 1*) and small pictorial (*Fig. 2*) for your use. All of the parts you will need are available from Antique Electronic Supply (I have included part numbers). If you use another supplier you must determine the equivalent parts.

AMP UPDATE

I don't like to drill new holes in the stock chassis, so I designed this rebuild to use existing holes. I will not describe the cosmetic restoration of the amp except to say that to refinish the chassis requires removal of all the components

TABLE 1
AMP PARTS LIST

4 — 0.1µF 630V film caps (Illinois C-TD1-630 or Solen CFSD1-630)
2 — 1µF 630V film caps (Illinois C-T1-630 or Solen C-FS1-630)
2 — 220µF 160V electrolytics (C-ET220-160)
1 — RCA input jack (S-H67R or S-H67B)

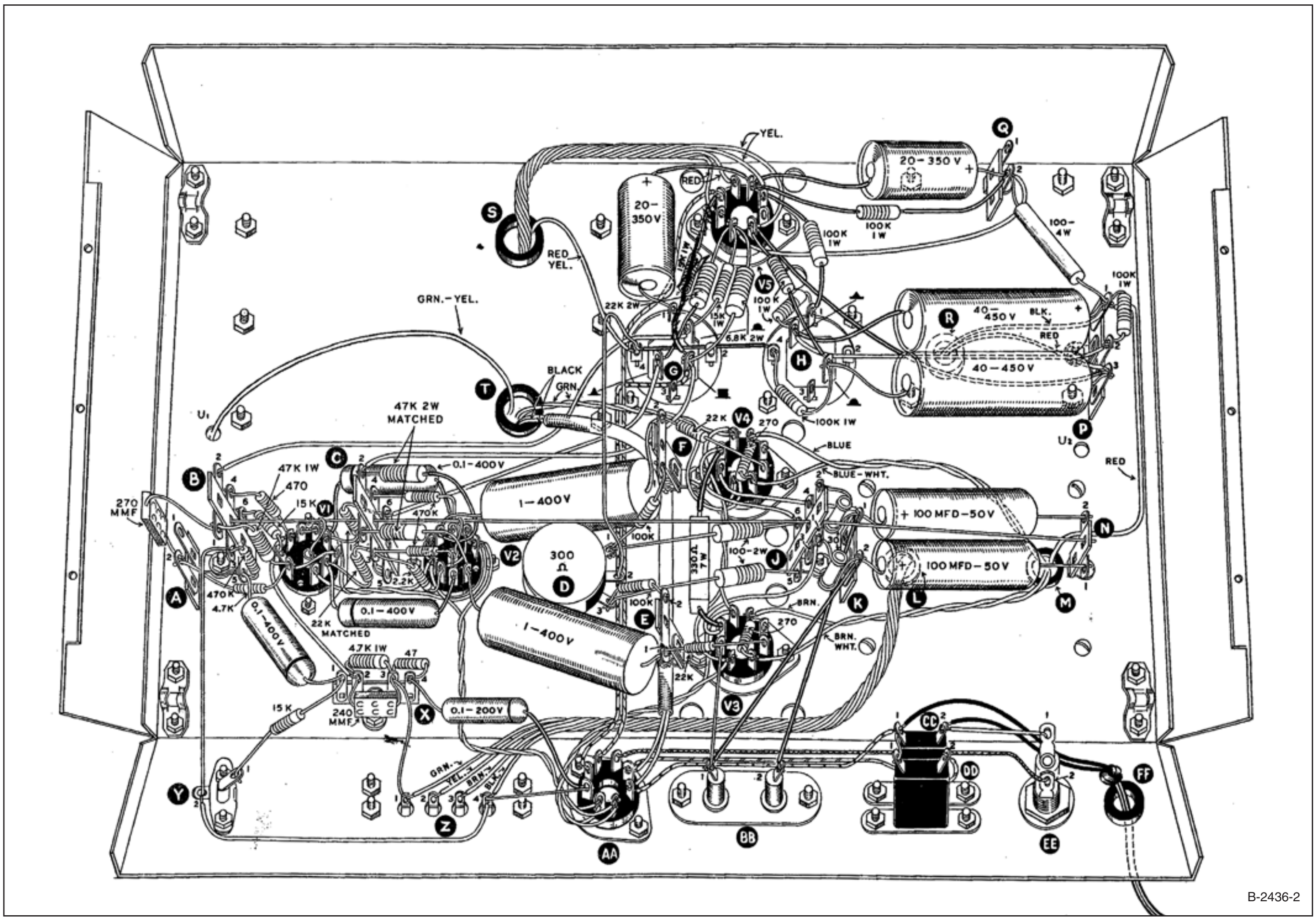

FIGURE 2: Heathkit's diagram of amp underside.

and a complete refinishing and reassembly. (I have not yet found the appropriate color of gold paint to match the chassis' original finish; if anyone knows, please contact me!)

Table 1 lists the rebuild parts for the amp section. Heath used 400V paper caps, but, again, I decided to be conservative, and use 630V film caps. Whichever you use, you will find that the new ones are less than one half the physical size of the original "pyramids."

If you are replacing the input jack, carefully remove the ground and bus wire from the input jack, as well as the 15k resistor to the center connector of the jack. Install the new jack and securely attach the ground and bus wires, as well as the 15k resistor.

Next, replace the 0.1µF input capacitor, then the 0.1µF cap going from terminal X to the front-panel octal socket (AA). If you are not going to use the octal power-supply connection, you can move the ground end of the 0.1µF cap from the socket directly to terminal 4 (Z—output strip). I usually deal with one capacitor at a time to avoid potential mistakes. Next, replace the 0.1µF coupling capacitors between tubes V1 and V2.

It is best to clip the leads of the old caps and use a set of hemostats and a pencil iron to remove the excess old leads. Use insulated sleeving on all exposed leads and dress your work according to *Fig. 2*. (The input sections of the amp are pretty crowded, so take your time and check your work carefully. Extra care here can save nasty sparks later!)

Next, replace the 1µF coupling capacitors that go to the output tube sockets. The last step in this section is to replace the two bias supply electrolytics. Heath used 100µF 50V units; I use 220µF 160V units, which provide more regulation for the bias supply, and a little more headroom.

Use your digital ohmmeter to check all the resistor values in the amp section. I have found resistors that were over 50% off marked value, even though the amp sounded fine. The most critical are the 47k 2W units attached to the plates of V2 and the 22k units on the cathode and plates of V2. They ideally should be matched to less than 5% difference. I usually replace anything outside 10% of marked value with 5% 1 or 2W metal film resistors.

At this point you have finished the amp section; check everything carefully again before proceeding to the power supply. Your amplifier should look similar to *Photo 2*.

Table 2 lists the new power-supply parts, which are considerably more

TABLE 2
POWER SUPPLY PARTS LIST

1 — 50–50–500V can electrolytic (C-EC50-50-500)
1 — 40–20–20–20–500V can electrolytic (C-EC40-20X3-500)
3 — 47µF 450V electrolytics (C-ET47-450)
2 — 22µF 450V electrolytics (C-ET22-450)
1 — 750Ω 10W resistor (see text)
1 — 3- to 5-lug terminal strip (P-05001H)
1 — 6-lug terminal strip (P-0601H)
2 — 100k 1W resistors (R-E100K)
2 — can clamps (5H122)


PHOTO 3: Replacements for original components.

compact than the original parts. See the difference in *Photo 3*.

POWER-SUPPLY UPDATE

The second part of the rebuild can be confusing, because Heath utilized electronically unused terminals of the rectifier tube as tie points for equalization resistors. Be sure to look closely at the figures and photos. Start by carefully unsoldering all the connections to the original electrolytic cans G and H (or do can G first.) With the exception of two 100k resistors, all the other components will still be attached to something else at the opposite end. If you will not be using the front-panel octal socket front power connections (AA), you can remove the 15k 1W resistor attached between can G and pin 5 of the rectifier tube (this is an unused tie point on the 5R4).

You can also remove the wire attached to pin 5 (which was used to power a WAP-2 preamp, which you do not want). Remove the existing mounting screws and the old can capacitors. The new can caps should be loosely mounted in their chassis clamps, which should be secured to the chassis. It may take a little moving, bending, or filing to get the right fit, depending on the clamps you use.

Once the clamps are in place, turn the chassis over and rotate the caps for orientation similar to the originals. Note the larger can has an extra 40µF segment (discussed later), so orient the three 20µF sections accordingly. On my cans the 40µF terminal is marked with an X.

Referring to the photos and figures, re-attach all the leads and resistors, soldering very carefully, since many of the terminals have multiple connections. It is critical that you connect the ground bus wire to the negative lead of both can caps. Replace the outboard individual 20µF 350V cap with a new 22µF cap, and then replace the two large 40µF 450V caps with the new 47µF 450 units (*Photo 4*).

I usually replace terminal strip Q with a 3–5-lug strip to allow fine-tuning of the power resistor (series or parallel) to obtain the correct voltages. You can actually leave the 100Ω resistor in place until later, or just leave nothing connected between the Q and P terminal strips. Replace terminal strip P with a 6-lug unit and move the 2-lug unit that was at P to the lower terminal of cap can H. Wire the top portion of the 6-lug strip exactly the same way in which the 3-terminal strip was wired. Now wire the 40µF section of the large can cap to add it to the power supply after the choke (*Fig. 3*).

Attach a wire from the 40µF section of can G to terminal 3. At the same time attach one of the new 100k resistors to the 40µF terminal and attach the other end of this resistor to the grounded terminal of the small strip (2 lug). Connect the other new 100k resistor between terminals 1 and 3. Run a wire from terminal 1 of this strip to terminal 4 of the new strip P. Connect the remaining 47µF 450V cap negative terminal to lug 1 of the small strip and the positive end to lug 4 of new strip P.

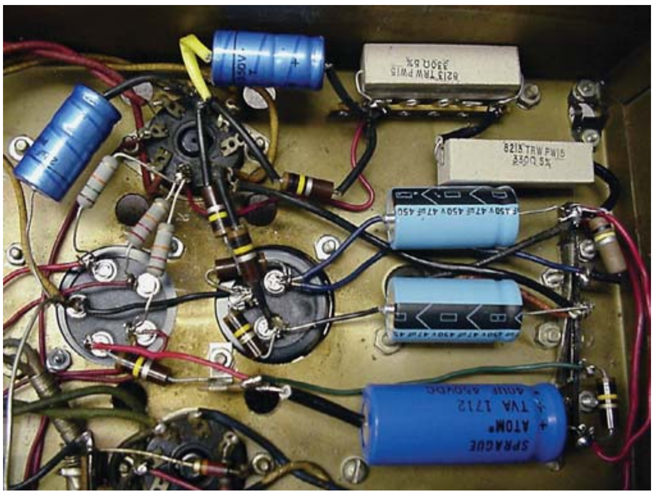

PHOTO 4: Power-supply mods.

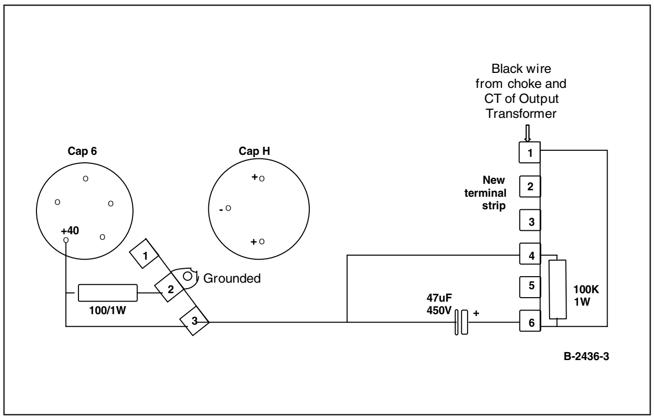

FIGURE 3: Wiring guide for power supply.

Connect a wire between terminal 1 and terminal 4 of the new 6-lug strip P. This effectively doubles the filter capacity of the power supply after the choke. As mentioned before, you will need to increase the 100Ω power resistor (between strips P and Q) to between 300 and 1000Ω during the final check-out. Your completed amplifier should look similar to *Photo 5*.

Now, double-check all your wiring of the power supply. You can insert the 5R4 tube and slowly power up the supply to check the voltages before and after the choke connections on terminal strip Q. You should have around 510V DC before the choke and slightly less after (the exact voltage is not terribly important at this point; you are just checking out completed power-supply wiring.)

Hook up a dummy load and input source to the amp, insert all the tubes, and slowly bring up the line voltage. Referring to the schematic, measure the voltages at the following points:

Plate terminal pin 1 of V1 (12AU7), 88V
Plate terminal pin 6 of V1 (12AU7), 280V
Plate terminal pin 1 of V2 (12AU7), 255V
Plate terminal pin 6 of V2 (12AU7), 255V

If these voltages are more than 20% outside listed values, you will need to increase or decrease the value of the 100Ω power resistor connected between terminal strips P and Q—anywhere between 300 to 1000Ω. Use a 10W series or parallel string to get a value that provides the needed voltages. In order to get exact voltages at all these points, you can also change the values of the 15k, 22k, and 6.8k resistors connected to can G and the tie point on the rectifier tube. (If you are obsessive, remember that Heath allowed a 20% variance.)

To balance the output tube voltages, attach a DC meter to the terminal BB of the front panel test point. Adjust the balance potentiometer (D) for a zero voltage reading. You can also use these test points to read the actual cathode bias on the 6L6s by measuring the voltage from each of these points individually to ground. They should read between –40 and –50V.

Now, measure the plate voltages on the 6L6s (pin 3 of each tube). Heath lists 480V DC as the correct voltage, but yours will probably be in the 500 to 540V range typical of today's line voltages. (Mine vary from 120–127V AC.)

When these amps were designed, the typical was around 110–117V AC. You can lower this by using an additional 10W power resistor in series with the center tap of the output transformer (moving the red center tap wire from terminal 1 of strip P and attaching it to the unused sixth lug of this strip. Place a 100Ω 10W power resistor between terminals 1 and 6 of strip P). This may affect your voltages going to the 12AU7s, and you may need to change the value of the power resistor between terminal strips P and Q to fine-tune everything.

TUBE SELECTION

A word of caution regarding tubes is important at this point: You should not replace the 5R4 rectifier tube with anything else, unless your actual voltage is low. This tube has the highest voltage drop of all commonly used rectifiers, and to use a 5AR4 or 5U4 will raise your high voltage significantly: 20 to 40V. Luckily, 5R4s are readily available from AES for a reasonable price. If you can get by the appearance of the Chatham "potato masher" style, they are less expensive than the regular versions.

The 12AU7s are common, and even most used ones are very quiet and work well. Many times the original tubes—obtained with the amp—will function very well. The original Heath W-5M used KT-66 Genelex output tubes, which have become very collectible and quite pricey. Many of the amps I have bought still have these tubes in place and they usually test "very good."

I typically use Sovtek 6L6/5881s, which seem to hold up well. You can also obtain some very nice well-matched, used tubes. I have not found an appreciable difference in sound between these and newer replacements. The suggestion may be heresy, but whatever you like will work. You should not use "coke bottle" 6L6s, since they usually will not take the plate voltages you are likely to experience with this rebuild.

Photo 1 shows a slide switch in place of one of the AC outlets on the front panel. If an outlet is broken, replace it with a DPDT slide switch (AES P-H35-242), parallel the DP sections, and wire it as a power switch. This switch fits the holes in the chassis perfectly, so it requires no chassis modification. You can either wire the switch and the outlet to use the outlet as a preamp on/off, or leave it on as a constant, whichever works for you.

Once everything is in order, you can hook up your amp to some vintage speakers and enjoy the sweet, mellow sound these amps produce. Occasionally check the balance voltage at the front panel connection. If this requires frequent adjustment, you'll need to replace the malfunctioning output tube or tubes. If you did not replace the 1μF units in the amp section during the rebuild, this can also indicate that the caps are leaky and need to be replaced. Since many of these amps have functioned for 40 plus years, with this rebuild you can expect many years of trouble-free service.

If you have any questions or need some additional assistance, please feel free to e-mail me at Tuninfork@aol.com. ❖

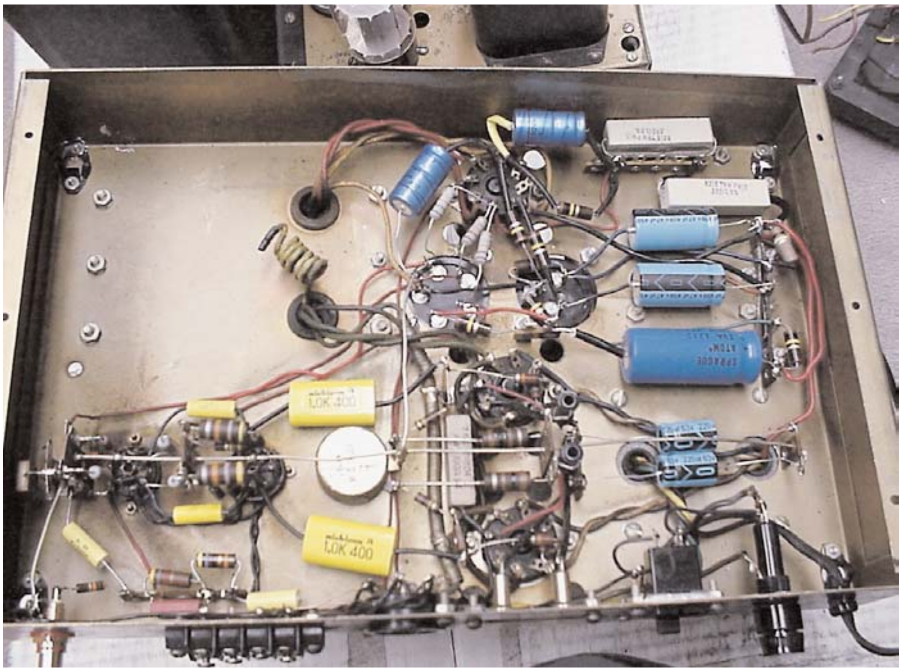

PHOTO 5: Completed mods.

SOURCES

Antique Electronic Supply
www.tubesandmore.com
480-820-5411

WF6G Vintage Manuals
www.w7fg.com

Sonic Comparison of Power Amplifier Output vs. Input

This test methodology uses the direct approach to evaluate power amps. **By Dennis Colin**

The output of an auxiliary power amp drives either a speaker or (through an attenuator) the input of the tested amp (*Fig. 1*), whose output then drives the speaker. An A/B switch and precise level matching allow instant comparison of the tested amp's input and output signals. Ideally, the auxiliary amp's load in switch position B (feeding the test amp) should be a second speaker whose impedance closely matches that of the auditioned speaker.

To avoid the error potential due to changing loads, and the power limitation (to that of the auxiliary amp), you can use the setup of *Fig. 3*: here the auxiliary amp always drives the speaker, and "listens" to either the input or output of the test amp. Both amps see a constant load versus switching. But I chose the setup of *Fig. 1* because I wanted the tested amp to directly drive the speaker, as in normal use. Also, *Fig. 3* requires a second speaker to correctly load the test amp, while in *Fig. 1* that's taken care of; and you can replace the "dummy load" speaker for the auxiliary amp with a resistor if that amp has a high damping factor and produces small changes versus load, compared to those produced by the insertion of the test amp in the path. The auxiliary amp's "sound," if reasonably transparent, doesn't matter; I considered its output the audio source throughout the testing.

In conventional A/B comparisons of two amps, you may prefer one over another, but that doesn't necessarily

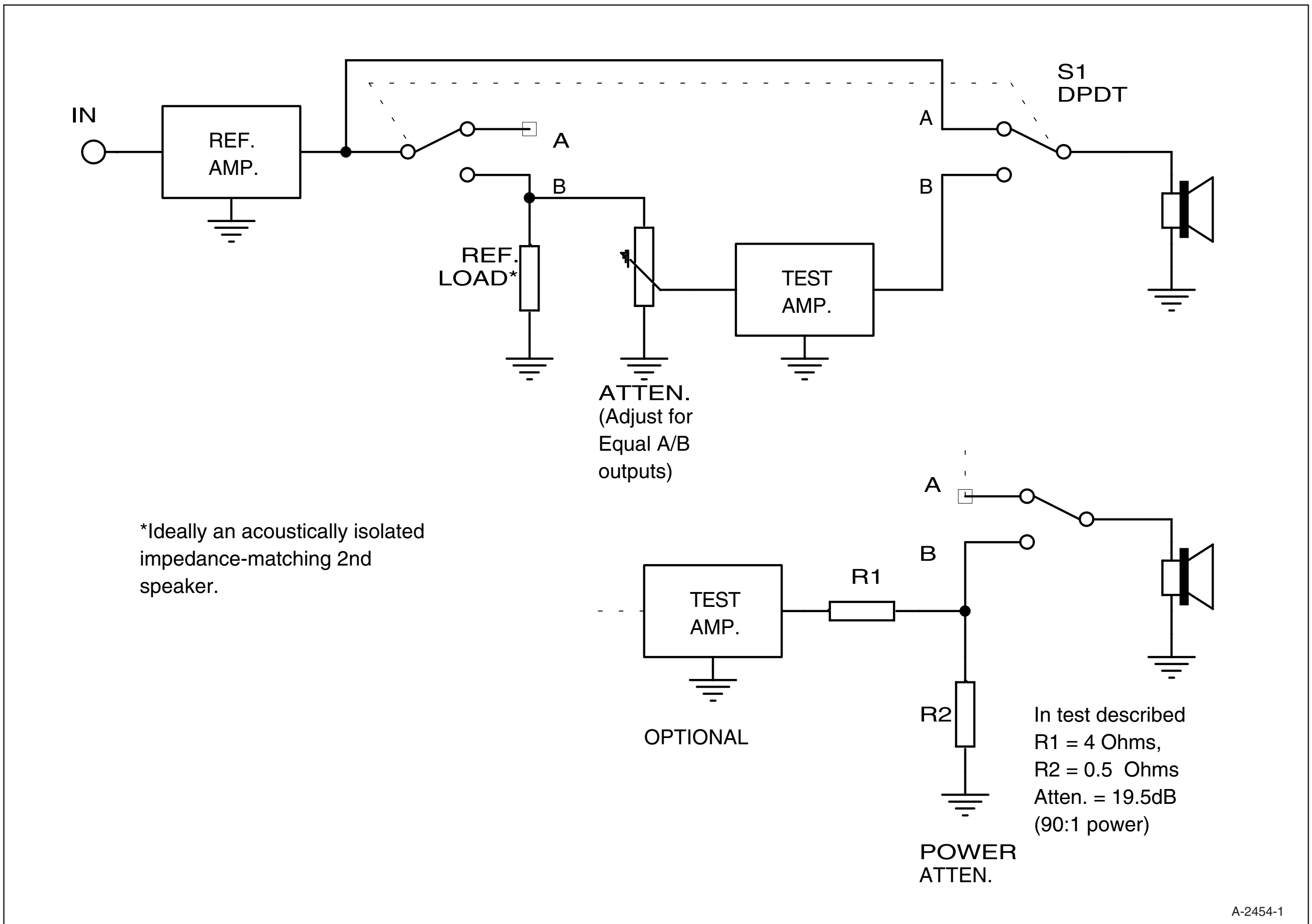

FIGURE 1: Sonic comparison of power amp output vs. input.

mean it's better in absolute terms (more faithful to its input source). It could have deviations from neutrality that perceptually compensate deficiencies elsewhere, or simply sound preferable to the other amp.

In contrast, the test described here allows you to hear only the colorations of the test amp. If it were the ideal "piece of wire with gain," the switching would produce no difference (other than the unproven claims of the microscopic audibility of short pieces of wire, and so on).

In *Fig. 2*, the speaker monitors the test amp's output-input difference signal (assuming a non-inverting amp). With sufficiently flat frequency response and low phase shift, you can adjust the attenuator for a deep enough null so that distortion residues can be directly heard, if sufficiently present.

A SPECIFIC TEST

The tested amp was "Mad Katy," a 125W per channel stereo unit, each channel comprised of four KT88s in push-pull Class AB with 67% screen tapping (nearly triode but 2× power) and non-feedback output stage linearizing. To (more or less) double any deviations from neutrality, the two channels were cascaded, each attenuated to unity voltage gain. The first channel was resistively loaded; the second drove the speaker.

The auxiliary amp was a 3W single-ended EL84 unit with a high enough damping factor (15W/8Ω load) so that the changing load (8Ω versus the speaker) was apparently of no sonic significance. The speaker was the Swans M1 (reviewed in *SB* 3/99), a very high transparency system with a ribbon tweeter. Its excellent coherence, even in the near field, allowed close monitoring (1M) for optimum in-room clarity. I performed the test in mono.

RESULTS

With 12 music selections and white noise, I performed switching with long segments, and also with rapid repetition of the first two seconds or several notes of the piece. With the latter, the exact and frequent repetition with input/output switching was a most sensitive test.

I didn't hear any obvious difference, so I continued switching for several hours. Any difference I occasionally thought I heard was on the edge of my

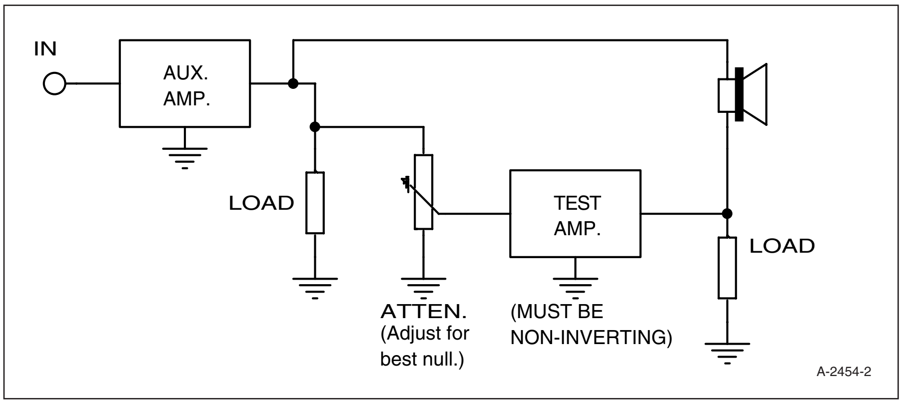

FIGURE 2: Audition of power amp output-input difference.

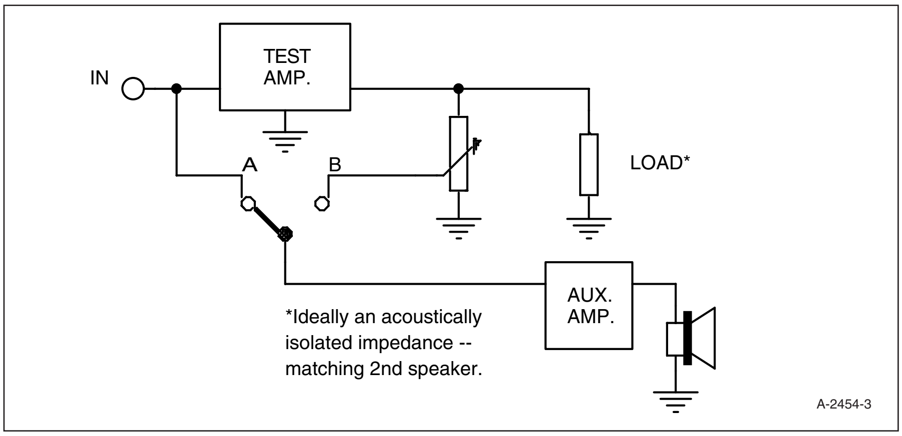

FIGURE 3: Alternate test methodology.

perceptibility threshold. But this first test was limited to 3W with a 125W per channel amp (basically a low-level transparency test), and the tested amp has a high damping factor (>20, 40Hz–10kHz) which delivers a very flat frequency response to the speaker. The high DF is obtained from a local current FB loop (no global FB is used).

FULL POWER TEST

In the previously described test, the test amp's channels were each attenuated to unity overall gain. But because the auxiliary amp's maximum power is about 3W, the test amp (125W per channel) output was also limited to 3W.

In a second test, a power resistor network attenuated the test amp's output by 19.5dB (90:1 power ratio). A/B levels were re-matched. Now, with 1.4W from the auxiliary amp, the test amp was driven to 125W, with 1.4W reaching the speaker. A drawback of this method is that the power attenuator presents a nearly resistive load to the test amp output, isolating it from the varying and reactive speaker impedance. To eliminate this drawback, the auxiliary amp would need a maximum power capacity comfortably greater than that of the test amp; this wasn't available.

In this test, admittedly with the unfairly "nice" test amp load, little if any audible differences could be established. But levels approaching clipping (estimated 2% THD on peaks) effortlessly revealed the distortion.

OUTPUT-INPUT DIFFERENCE SIGNAL

With the test setup of *Fig. 2*, I achieved a null of over 30dB. With the resulting low SPL, my sensitivity to anomalies was certainly impaired. But for what it's worth, the sound was not noticeably distorted until overdrive became (rapidly and most detectably) audible as the drive level was increased.

A NOTE ON THE TEST AMP LOADING

In a previous test with a different speaker, whose impedance drops to 3Ω from 5–20kHz, the amp produced a slight "softening" when I used its 4Ω tap. This also occurred with the Swans M1 (7Ω at 300Hz and 20kHz) when driven by the amp's 8Ω tap. I heard no "softening" with the 4Ω tap, which I used in the tests. With this amp, you should use a tap (2, 4, 8Ω) that's lower than the lowest speaker |Z| within the audio band.

I apologize for not having tested other amplifiers with this direct input/output methodology—my motivation for this testing was to evaluate the "Mad Katy" design (after normal listening tests and A/B comparisons). But my reason for presenting this article is to describe this "intrinsic fidelity" test, not to report on specific amplifier evaluations—that's "your mission, should you choose to accept it."

However, I had previously compared several amplifiers in conventional A/B switching tests, and had no difficulty hearing small but obvious and consistent differences. Compared with the "Mad Katy" tube amp, two solid-state amps (conventional Class AB, good but not audiophile quality) sounded more "crisp" (artificially so, I think). Also compared, my 1960 vintage H.H. Scott Stereomaster 222D amp (20W per channel push-pull 7189 tubes, pentode, lots of global FB) sounded slightly "soft" and less detail-resolving—slightly less overall clarity. By "slight," I mean about three times my audibility threshold.

If anyone would like to try the "intrinsic fidelity" test on any amplifier(s), I (and hopefully other *aX* readers) would welcome a report on the results.

A NOTE ON NON-BLIND SWITCHING COMPARISONS

Hours spent with an A/B switch, precise level-matching, and a highly transparent and neutral component, can be very revealing not only of the component, but also of the influence of "expectational bias" (and other thought-process activity) on your perception. Blind testing, ideally with a pre-noted but random-appearing A/B sequence, can eliminate this influence. But with the auditioner also doing the switching, you must persist with extended switching repetition, until you can suspend all thought by forgetting about the test purpose and becoming "absorbed" into the music. At any given moment, you can think or observe, but not both together with full clarity.

A NOTE ON MY HEARING

My age is 62, and from 25–35 I played in rock bands. Nevertheless, I can hear to 15.2kHz at 103dB SPL, and to 14.0kHz at 83dB SPL. I can easily hear the improved clarity, resolution, HF smoothness, and freedom from grain of SACDs versus 16/44 CDs. I can hear differences between most of the amplifier pairs I've compared. On a very few occasions I might have heard differences in speaker cables (with long cables such as 12′). I have *not* heard the "sound" of different (good quality) connectors, well-constructed clip leads (I know, heresy!) or a few trillion copper oxide atoms.

CONCLUSION

These tests are simple in concept and most likely not new. I present them here to remind you that—amid all the controversy regarding "euphonic colorations" versus "musical truth," and so on—you can evaluate an audio power amplifier without comparison to another unit, but directly.

With this test, if the sound is "better" with the amp switched in, a higher "musical truth" may be *perceived*, but it is *not* "absolute signal truth"—the amp is obviously modifying the signal, perhaps functioning as an expensive "tone control." ❖

STATEMENT OF OWNERSHIP, MANAGEMENT AND CIRCULATION

(Required by U.S.C. 3685.) Date of filing: Sept. 30, 2004. Title of Publication: AUDIOXPRESS. Publication Number: 1548-6028. Frequency of Issue: Monthly.
Annual Subscription Price: $34.95. Location of the headquarters or general business offices of the publisher: Audio Amateur Inc., PO Box 876, Peterborough, NH 03458-0876.

Publisher: Edward T. Dell, Jr., PO Box 876, Peterborough, NH 03458-0876. Assistant Publisher: Dennis Brisson, PO Box 876, Peterborough, NH 03458-0876. Owner: Audio Amateur Inc., PO Box 876, Peterborough, NH 03458-0876.

Stockholders owning or holding 1 percent or more of the total amount of stock: Edward T. Dell, Jr., PO Box 876, Peterborough, NH 03458-0876. Known bondholders, mortgages or other securities: None.

	Average # copies each issue during preceding 12 months	Single nearest to filing date
Total # copies printed	12,625	12,000
Mailed Subscriptions	6,748	6,343
Sales Through Dealers Counter Sales and other Non-USPS distribution	4,257	4,188
Free Distribution (complimentary)	361	500
Total distribution	11,366	11,031
Copies not distributed	1,259	969
Total	12,625	12,000

I certify that the statements made by me above are correct and complete. Publication number 787-840. Edward T. Dell, Jr., Publisher.

New Chips On The Block

D2Audio Class D Audio Modules

By Charles Hansen

PHOTO 1: D2Audio module.

D2Audio's multi-channel digital amplifier module brings 90% high-efficiency sound to receivers, home theaters, and multi-room distributed audio systems. The compact amplifier modules are thinner, lighter, and run cooler than conventional Class-AB audio amplifiers. While Class-D amplifiers have been around for a while, especially in IC form, D2Audio has developed a switching amplifier that uses feedback to correct noise and distortion.

Analog inputs are converted to 384kHz PWM, and the module can directly accept serial digital PCM audio at 44.1, 48, 96, and 192kHz. The PWM output from the power MOSFETs as

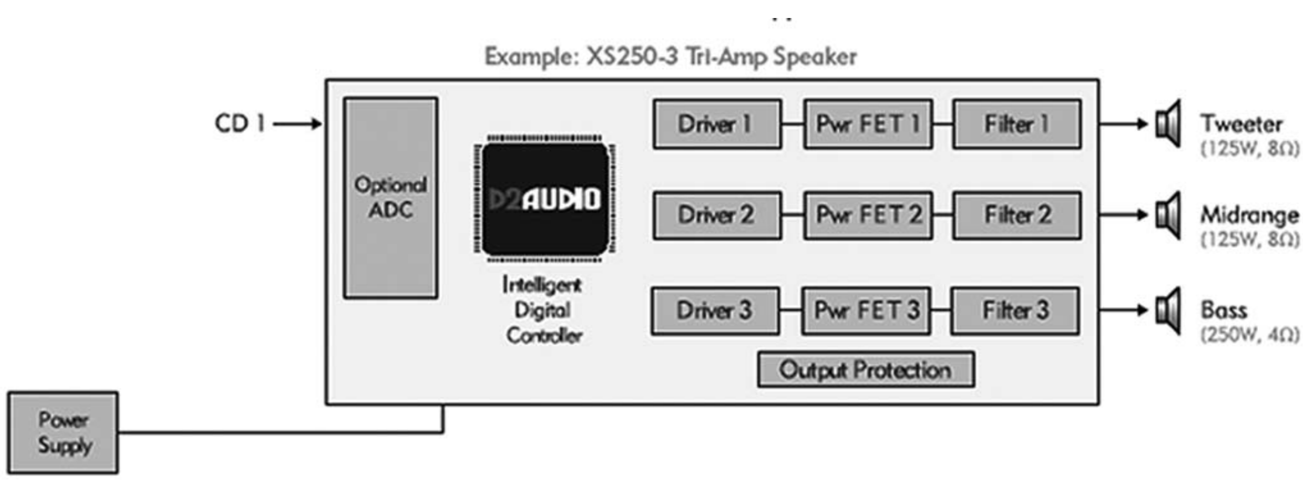

PHOTO 2: D2Audio block diagram.

well as the current delivered to the speaker is monitored and converted to digital signals that are applied to a patented DSP chip, which adjusts the PWM signal to correct for noise and distortion. This adaptive digital correction feedback results in THD+N as low as 0.05%, and power outputs from 100W to 2kW, comparable to conventional linear amplifiers.

One multi-channel, digital module allows up to a 7.1 receiver, amplifier, and DVD player to be combined into one slim box. The module reduces high-performance components to a single, simple, clean design. Moreover, the amplifier module has a unique intelligent digital control capability which can deliver innovative features and proprietary functions that are only limited by the imagination of the OEM manufacturer. The D2Audio modules are said to be the only switching amplifiers that are THX certified.

Rack mount, distributed audio amplifiers are traditionally large and heavy, with high heat dissipation. D2Audio digital amplifier modules diminish the heat and dramatically reduce the size and weight of bulky 16-channel receivers. Beyond the aesthetics, multiroom distributed audio designers can implement any number of the added features provided by the D2Audio intelligent module. The company provides a graphical development environment called Audio Canvas that OEM designers can use to customize the DSP control chip functions.

The latest trend in speakers is to install the amplifier inside the cabinet for networked audio. D2Audio modules increase the power capabilities of these new speakers by generating tremendous power in a small package. Wired and wireless networked speakers can benefit from lower distortion with D2Audio pure digital inputs. ❖

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2004 *audioXpress* Index

SUBJECT INDEX

ACOUSTICS

AMPLIFIERS

AUDIO AIDS

BOOK REVIEWS

DISTORTION

EDITORIALS

ELECTRONICS TIPS AND TECHNIQUES

ENCLOSURES

HORNS

LISTENING TESTS

LOUDSPEAKERS

NEW CHIPS

OBITUARIES

POWER SUPPLIES

PREAMPLIFIERS

PRODUCT REVIEWS

TEST EQUIPMENT

TONE CONTROLS

TUBES

AUTHOR INDEX

ALDEN, RAY

ALEKSANDERSEN, RUNE

AUGSPURGER, G.L.

BROWN, BRUCE

CAMPBELL, DICK

CLOW, SHERM

COLIN, DENNIS

CRAWFORD, DICK

CYR, DANIEL L.

DANAVARAS, GEORGE

DAVENPORT, DAVID

DELL, EDWARD T. JR.

DICKER, GRAHAM

DIDDEN, JAN
"Musicality and Distortion: A Conversation," (with Dennis Colin and Jean Hiraga), Oct., p. 14.

FERGUSON, DANIEL L.
"12″ Dual Voice Coil Servo Subwoofer System," Sept., p. 6.

FITZMAURICE, BILL
"The DR250 Horn," Jan., p. 20.
Producing in the Home Studio with Pro Tools by David Franz (Book Review), March, p. 57.
"The Tuba 24 Horn," April, p. 34.
"The DR200 Horn," Aug., p. 24.

FRANKLAND, SCOTT
Inside the Vacuum Tube by John F. Rider (Book Review), Jan., p. 68.

FRUITEMA, BERT
"A Dutch Tube Amp" (Showcase), April, p. 69.

GALO, GARY
"Adcom's GFP-565 Preamplifier, Part 3," Jan., p. 42.
"Adcom's GFP-565 Preamplifier, Part 4," Feb., p. 24.
"GFP-565 Preamp Follow-Up Mod," Dec., p. 26.

HAGERMAN, JIM
"Frykleaner: An Audio Burn-In Generator," March, p. 36.

HANSEN, CHARLES
"Thorens TD 295 MK III," Jan., p. 56.
"Bicron Trans and Burr-Brown DSD1700 DAC," March, p. 60.
"AD5241/2; WM8816 and WM8722," April, p. 45.
"Alesis ML-9600 Mastering Recorder," April, p. 46.
Op Amp Applications edited by Walt Jung (Book Review), April, p. 58.
"Speaker Cables Product Review," (with Muse Kastanovich), May, p. 52.
"BNC Jack Positioner," May, p. 72.
"DACT CT 100 RIAA Preamp," June, p. 58.
"Audio Interconnects," July, p. 52.
"Selectronic Triphon II Crossover and Quad Amplifier," Aug., p. 54.
"Selectronic Grand MOS Silver Design Amplifier," Sept., p. 42.
"Silonex Audiohm," Sept., p. 49.
"Pioneer DV-563A Universal A/V Player," Nov., p. 36.
"H.V. Differential Amplifier," Dec., p. 20.
"D2Audio Class 2 Audio Modules," Dec., p. 44.

HEBEL, MARTIN
"The MCUTracer, Part 1," (with Jack Walton), Aug., p. 40.
"The MCUTracer, Part 2," (with Jack Walton), Sept., p. 63.

HIRAGA, JEAN
"Amplifier Musicality," March, p. 32.
"Musicality and Distortion: A Conversation," (with Dennis Colin and Jan Didden), Oct., p. 14.

HONEYCUTT, RICHARD (AND BETTY JANE)
"Virtos Noise Wizard," March, p. 58.
"Triphon Electronic Crossover, Listening Tests," (with Betty Jane Honeycutt), Aug., p. 62.
Sound Recording Advice by John J. Volanski (Book Review), Sept., p. 53.

KANG, DON
"An 8W 2E24 Amp," March, p. 20.

KASTANOVICH, MUSE
"Speaker Cables Product Review," (with Charles Hansen), May, p. 52.
"Kimber and Supra Interconnects Listening Tests," July, p. 60.

KECELI, JENOË
"Build a Universal PC Sound Amplifier," May, p. 36.
"Audio-Optical Isolation Amp," Oct., p. 56.

KOBAYASHI, SATORU
"KT88 Hybrid PP Stereo Power Amp," June, p. 34.

KOONCE, G.R.
"Why Speakers Have Slanted Fronts, Part 2," Jan., p. 50.
"Why Speakers Have Slanted Fronts, Part 3," Feb., p. 48.
"Why Speakers Have Slanted Fronts, Part 4," March, p. 52.
"Vented 8″ Driver Subwoofer, Part 1," (with R.O. Wright), May, p. 18.
"Vented 8″ Driver Subwoofer, Part 2," (with R.O. Wright), June, p. 44.
"Vented 8″ Driver Subwoofer, Part 3," (with R.O. Wright), July, p. 44.

KORNACKER, MICHAEL
"High Current-Regulated Heater Supplies," June, p. 72.

KRUSE, DANA
"The J-Low Project" (with Nelson Pass), Feb., p. 14.

LIN, JAMES
"Canto Sirena, Part 1," Jan., p. 32.
"Canto Sirena, Part 2," Feb., p. 38.
"Modifying Dynaco's SCA-35," Aug., p. 46.

MACDONALD, RICK
"A Battery-Powered Class A Headphone Amp," July, p. 14.

MATTERN, JOHN
"Against the Wall: An Upgrade," March, p. 40.

MCKENZIE, MARK
"Remaking Tang Band's W3-8815," Dec., p. 54.

MILLETT, PETE
"A Low-Mu Triode Preamp," Feb., p. 4.

MORIYASU, JAMES
"Usher CP8871," Feb., p. 54.

MOULTON, DAVID
Mastering Audio by Bob Katz (Book Review), July, p. 68.

Classifieds

VENDORS

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AD INDEX

Yard Sale

FOR SALE

Sonic Frontiers SFL-1 preamp—PC board complete with Vishay, Caddock, MIT caps, power transformer, no chassis or switches, $150.00. SF Line 1 preamp as above, $375. D2D 24/96 upsampler, $250. Assemblage ST 40 (2x EL 34 ST 70 type) transformer set (power and outputs), new, $235 per set. dave.pitt@rogers.com or 905-819-8462.

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

CORRECTIONS

In the excerpt from Ray Alden's new book (*aX* 10/04), we inadvertently neglected to acknowledge the source for Figs. 2 and 3 (p. 21). The figures were used courtesy of Parts Express. We apologize to Parts Express for the omission.—Eds.

We reprint the following figures that were washed out in last month's review of the Pioneer A/V Player (pp. 45–49). We apologize to author Charles Hansen and to our readers for any inconvenience this may have caused.—Eds.

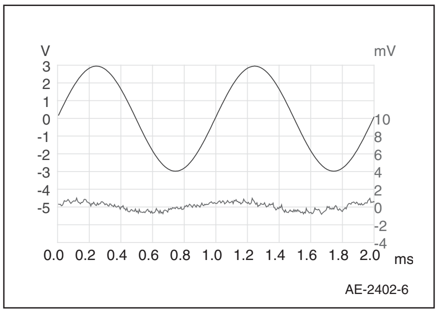

FIGURE 6: Distortion residual of 1kHz sine wave (CD).

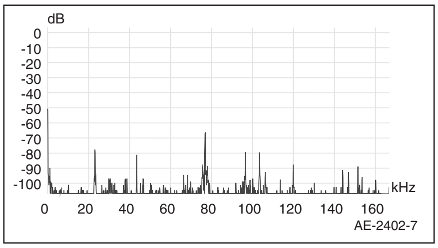

FIGURE 7: Spectrum of 1kHz at –90.31dB extended to 166kHz (CD).

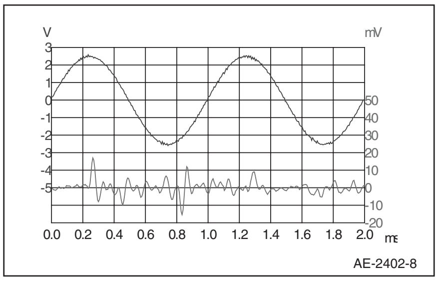

FIGURE 8: Distortion residual of 1kHz sine wave (SACD).

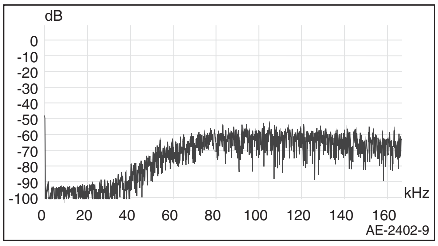

FIGURE 9: Spectrum analysis of 1kHz –90dB sine wave extended to 166kHz (SACD).

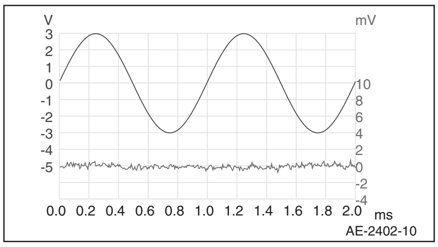

FIGURE 10: Distortion residual of 1kHz sine wave (DVD-A at 24/96).

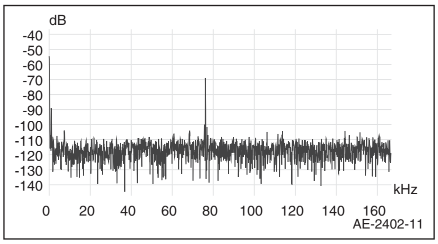

FIGURE 11: Spectrum analysis of 1kHz –90dB sine wave extended to 166kHz (DVD-A at 24/96).

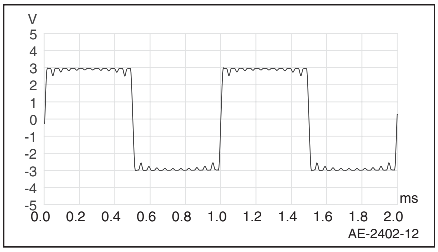

FIGURE 12: 997Hz square wave response (CD).

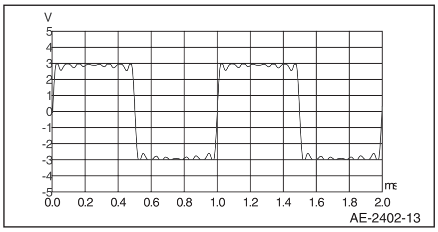

FIGURE 13: 997Hz square wave response (MP3).

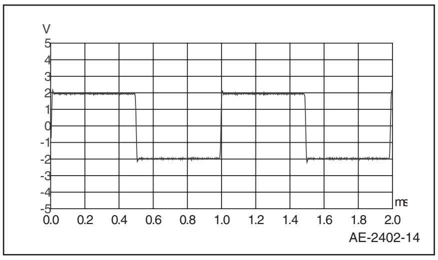

FIGURE 14: 1kHz square wave response (SACD).

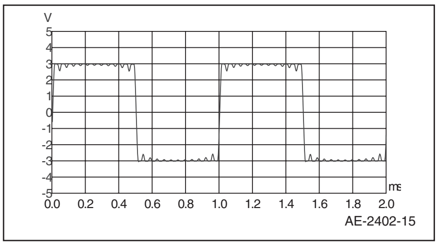

FIGURE 15: 1kHz square wave response (DVD-A 24/48).

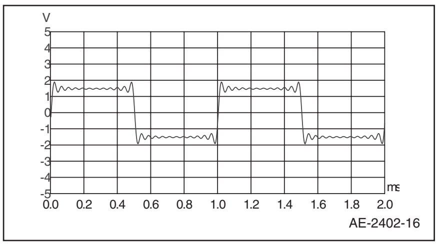

FIGURE 16: 1kHz square wave response (DVD-A 24/48 –6dBfs).

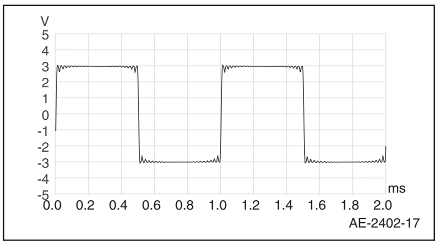

FIGURE 17: 1kHz square wave response (DVD-A 24/96).

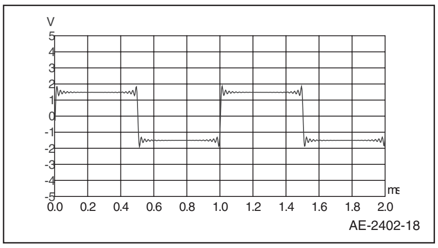

FIGURE 18: 1kHz square wave response (DVD-A 24/96 –6dBfs).

There is an error in my article published in the September issue of *audioXpress*, "Saga of a Tube OTL Amp." In Figure 7 on page 24, the next to the last line under "Notes" reads: "= 600Ω – 82Ω + 518Ω," it should read: "= 600Ω – 82Ω = 518Ω."

Glen Orr
glenorr@essex1.com

In looking at the schematic in Figure 2 of the "High-Quality Tube Control Unit" (*aX* 10/04, p. 26), I could see that C5 was way too big for the proper RIAA EQ curve. The parts list shows 0.012µF, so the schematic should say 12nF rather than 120nF.

Mr. Still didn't mention it, but both the phono stage and line stage outputs are the inverse of their input signal polarities. This will result in inverted output polarity for line inputs, but normal polarity for the phono input. For those who believe absolute signal polarity is important, they will need to reverse their speaker connections for line inputs (assuming no inversion in the power amplifier, which is the usual case) and switch to the normal connection for their turntable.

Chuck Hansen
Ocean, N.J.

Joseph Norwood Still responds:

Thanks for noting that the capacitor regulator tube schematic should be 12N instead of 120N. Depending on the number of stages in the power amplifier, the input and output signals may or may not be inverted in a given audio system.

As to the inversion of the signal polarity, I don't consider it a relevant problem. For any reader who considers it a problem, the method you described for correcting the input and output inversion is a satisfactory solution.

An incorrect schematic for Figure 6 appears in David Davenport's article, "Line Stage Odyssey Continues" (November, p. 58). The correct figure (*Fig. 6*) appears below.

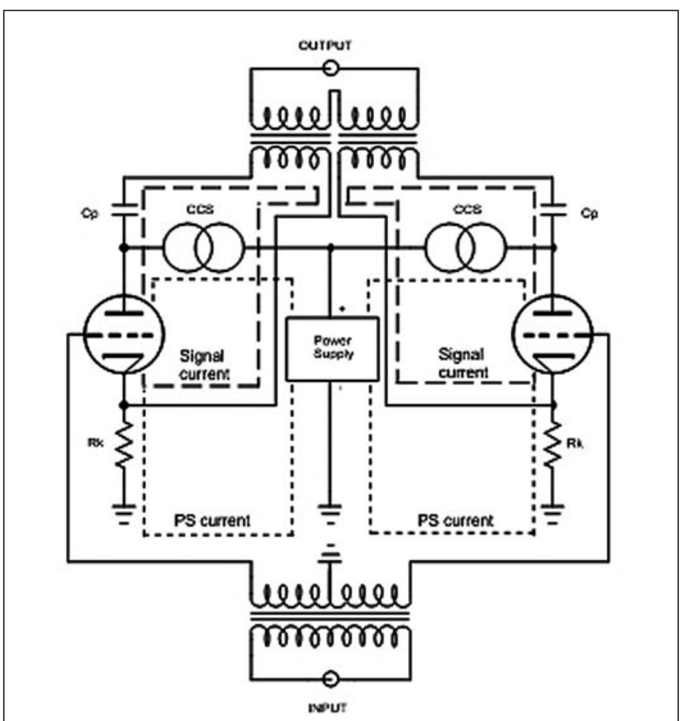

FIGURE 6: Push-pull parafeed line stage.

HELP WANTED

I'm having a heck of a time trying to find someone who makes horn adapters—preferably aluminum, but composite OK. I'm a wood horn maker with a few designs I'm working with—most of which are vintage recreations (i.e., Altec 811/511/311, JBL 2397, Westlake, and so on). For a professional and finishing touch, I need adapters in various sizes that are new and current and readily available. Would you know of anything?

Also, any insights on how I might be able to successfully market these wood horn products would greatly be appreciated.

Horn-Crazy (Dean L.)
deanlabbe@shaw.ca

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

Remaking Tang Band's W3-881S

Discover these three simple modifications to improve the performance of this 3″ cone driver.

By Mark McKenzie

The foundation of an accurate loudspeaker is its drivers. Start with a better driver, end up with a better loudspeaker. Following this adage, I have been focusing on making a better wide-range, small diameter cone driver. Any good research program includes examining what is already available, determining where it is weak, and figuring out how to make it better. The Tang Band W3-881S 3″ polypropylene cone driver showed promise in stock condition and could, with the application of three modifications, become a "better" driver of high-fidelity quality.

FULL-RANGE PROS & CONS

Full-range is the ideal in loudspeaker driver design. With a full-range driver, designing the rest of the loudspeaker is easy. In contrast, the list of disadvantages of loudspeakers with multiple drivers splitting the bandwidth is long. For example, the bandwidth of each driver is bracketed by the problems of motor resonance at the low-end and diaphragm break-up at the high-end. Typically, the frequency of one of these problems in one driver will correspond with the most linear reproduction band of the other driver. While the crossover network minimizes the magnitude of the problem, it does not eliminate it.

ABOUT THE AUTHOR

Dr. Mark McKenzie, along with Don Spangler, co-authored the *Speaker Builder* article "Modified Strathern Ribbon Speaker" in the early 1980s. Since then he served for several years as a reviewer and technical editor of *Sensible Sound*, while also playing at being a college professor and scholar. He is currently the editor of an academic journal and running his own acoustical design firm (madspeaker.com).

Then there is the blurring of transients caused by multiple drivers. When mounted on a flat baffle, the acoustic origins of tweeters and woofers do not match up. The acoustic origins of tweeters are always leading the acoustic origins of woofers. Transients are wide bandwidth signals. The bandwidth reproduced by the low-frequency driver will be separated in time from the transient bandwidth reproduced by the high-frequency driver.

PHOTO 1: Backlit view of front of diaphragm with dust cap removed. The photo shows dimple and glue line placement. With driver mounted in enclosure, the glue lines are not visible.

This phenomena exists because the point of acoustic origin is always behind the driver mounting plate and the deeper the driver, the farther back the point of acoustic origin. Then there are the design challenges of the crossover network itself. With full-range, there is no crossover, transients are not blurred, and there is only one motor resonance and one range of cone break-up resonance to contend with. Designing full-range is much simpler and more accurate in every way.

This full-range ideal, however, is more theoretical than practical. Even with a loudspeaker bandwidth goal of 40Hz to 20kHz and a reasonable variance range in output level, you are far from true full-range. Large-diameter drivers with system resonance in the 40Hz range do not approach accurate reproduction of higher-frequency overtones and transients. A very small number of small-diameter drivers come close to high-frequency accuracy but cannot reproduce low frequency sounds. At best, with an extended range driver, you can have as broad as possible coverage of the middle ranges of sound and push the crossover and all of its problems to the ends of the spectrum.

Since system resonance is less complex in structure and easier to compensate for in design, choosing a small-diameter driver to reproduce the bulk of the sound and crossing that over to a woofer at as low a frequency as possible seems the better way to go. Also, considering room interactions and psychoacoustics phenomena, a low-frequency crossover seems a better fit for the way you hear than trying for a high-frequency crossover.

DRIVER SELECTION

Still, even after deciding on a design using a small-diameter driver reproducing the bulk of the sound, you still must find a driver capable of doing this. This, too, is not easy. While most manufacturers make small-diameter electromagnetic drivers, few are rated beyond

10kHz, and most will top out between 5kHz and 8kHz.

Right now, small-diameter electromagnetic drivers are divided into two categories. There are the very expensive and the very inexpensive. The expensive category consists of just two brands, Jordan and Bandor. Both brands produce drivers with cones about 2″ in diameter and are rated beyond 20kHz. In North America, the 2″ wideband driver from either company will cost well in excess of $100.

The inexpensive category consists of three main brands: Fostex, Hi-Vi, and Tang Band. Prices in this category generally range from $32 down to $8–9. All of these drivers have audible problems. For example, none of the Hi-Vi drivers are rated beyond 15kHz, and none that I have tested make it to 15kHz with reasonably flat output. Their high-frequency rolloff has begun before 15kHz. This includes all of their small-diameter drivers, even their 2″ series with actual cone diameters closer to 1.25″.

The Tang Bands suffer from many of the same problems as the Hi-Vi drivers. Every paper cone driver tested has a narrow (high Q) breakup resonance that marks the high-frequency limit, and this limit is well below the highest rated frequency. In their 4″ paper cone driver (rated to 20kHz), this resonance is centered at 12kHz. Acoustic output falls steeply beyond the 12kHz cone resonance. In their 3″ paper cone driver (actually a 2″ diameter cone) this breakup resonance is pushed to about 17 or 18kHz, but it is still an audible problem that reduces the driver desirability.

Tang Band's polypropylene cone series is also down in output before 20kHz, but most of their models do not exhibit the high Q cone resonance. Over the last year I have tested many inexpensive small diameter drivers. Of these, the Tang Band 3″ polypropylene cone driver (2″ cone) sold by Parts Express, with the model number W3-881S, has the most promise. Tang Band makes many models, however, including a couple of polypropylene cone drivers with deeper baskets and longer voice-coil formers that have a high Q resonance at about 18kHz very similar to their paper coned models and should be avoided.

At about $14 this is the best of the inexpensive offerings, but it still has audible problems, which you can correct, however, through a modification or remaking of the driver. Once the driver is remade, it has performance adequate for covering the frequency range of 150Hz to 19kHz with a degree of accuracy suitable for a high-fidelity loudspeaker system.

STOCK PERFORMANCE

The W3-881S driver has many desirable features in a very inexpensive driver. It has a cast aluminum basket, Santoprene surround, and a self-shielding neodymium magnet structure. It claims to have a frequency range of 100Hz–20kHz, an f_S of 100Hz, Q_{TS} of 0.63, and power handling of 15W RMS. In contrast to the claimed specification, expect the stock driver to have an f_S closer to 120Hz, a Q_{TS} of over 0.7, and a frequency response of 100Hz–20kHz only with an acoustic output variance rating of ±6–9dB (varies from driver to driver).

Still, the performance shows promise. It has enough high-frequency output to put you close to 20kHz for the top end and an f_S low enough to be usable to 150–200Hz on the bottom end. It also has 50% more power handling capacity than the more expensive Fostex full-range 2″ cone diameter drivers and comparable low-frequency capabilities.

I was drawn to this driver because of its impulse response (*Fig. 1*). In impulse testing the drive signal is a short duration, positive-going electrical pulse lasting a much shorter time than the acoustic output from the driver. The driver begins accelerating during the applied signal, continues to deflect in the same direction after the applied signal stops, and then once the spring force of the spider and surround equals the inertial energy of cone structure, snaps back.

SPIKES

Electromechanical drivers always overshoot their rest position and produce a negative polarity acoustic output. For want of a better term, these large magnitude acoustic outputs are called spikes. A perfect loudspeaker in a sealed cabinet or infinite baffle will show three spikes: a first positive, first negative, and second positive. These three spikes are considered the onset response. Each successive spike in the onset response ought to be much less in magnitude than the preceding one.

After the first series of spikes, the driver continues to vibrate until finally coming to rest at the beginning position. This overshoot and later decay is caused by the combination of the motor system resonance and a complex variety of cone and surround resonant structures.

It is the magnitude ratio of first posi-

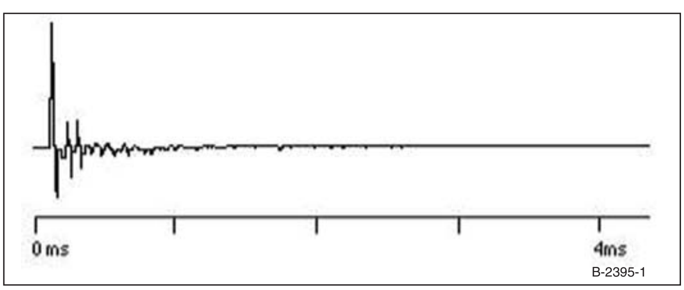

FIGURE 1: Stock Tang Band W3-881S impulse response. The driver shows promise. The ratio of the initial positive spike to first negative overshoot is good, and the overshoot is several times less in magnitude. The decay portion of the response is not as good, showing a double "echo" of the initial positive to negative spikes.

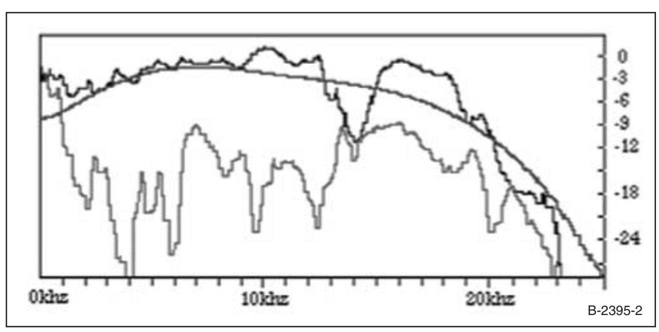

FIGURE 2: Stock Tang Band W3-881S, serial number 206, frequency response graph. The frequency response is drawn in black, the decay response is drawn in red, and the onset response is drawn in blue. This particular example would have to be rated 130Hz–19kHz ± 6dB. The onset response shows substantial rise in output from 500Hz to a maximum output at about 7kHz. There is also a great deal of output in the decay spectrum of the driver response.

tive spike to first negative overshoot that makes this driver so desirable. In the transient domain, this is a driver that is performing as well above 3kHz as the majority of tweeters. Those tweeters, often costing two to four times as much as the W3-881S, will exhibit a first negative-going spike that is larger in magnitude than the first positive-going spike. In contrast, the W3-881S has a first negative-going spike that is only one-third the magnitude of the first positive-going spike. That first positive to negative ratio is audible and is the reason many tweeters sound edgy and hard when trying to reproduce cymbals.

DECAY PROBLEMS

In stock condition, in contrast to the excellent onset, the decay portion of the impulse response shows several problems. Of particular interest are the two almost echoes of the first positive and first negative spikes. These should not be there and will reduce the clarity of the sound. There is also a small discontinuity between the onset response and the two echoes.

The stock driver frequency response (*Fig. 2*), reading from lowest frequency to highest, shows the response up at low-frequency resonance (consistent with Q_{TS} of over 0.7). The response then falls before rising again at about 3kHz, peaking just below 11kHz, and then generally falling above 11kHz. The contribution of the decay portion of the response is substantial. At several points the output level of the decay is down only 10–15dB from the combined output. Last, the onset frequency response shows a driver whose response is rising to 8–9kHz and then falling

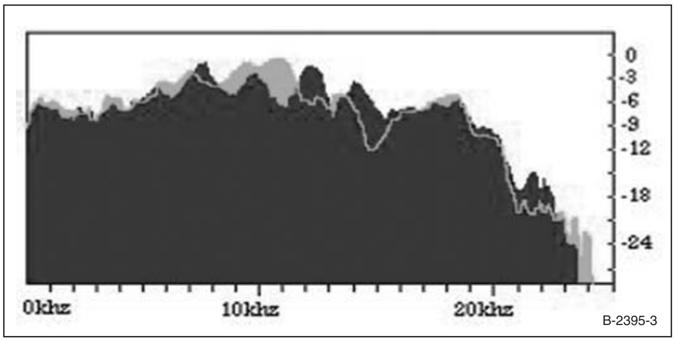

FIGURE 3: Stock Tang Band W3-881S, serial numbers 206 and 495, frequency response comparison graphs. The frequency response of driver serial number 206 is drawn in green; the frequency response of driver serial number 495 is drawn in blue. Key features in the response of any driver may not be present in the response of any other driver.

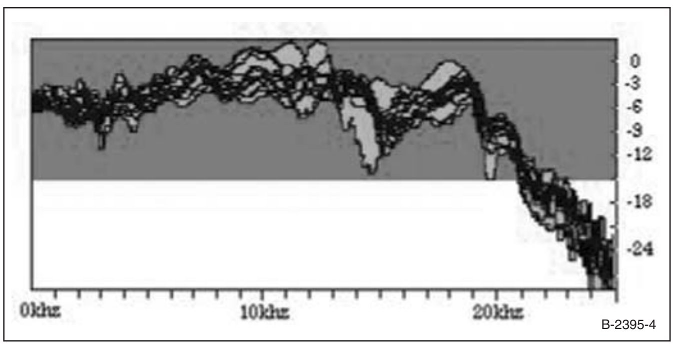

FIGURE 4: Frequency-response comparison of 14 stock Tang Band W3-881S drivers. To include all drivers, frequency-response rating should be ±9dB. Driver to driver variance can be as much as 18dB.

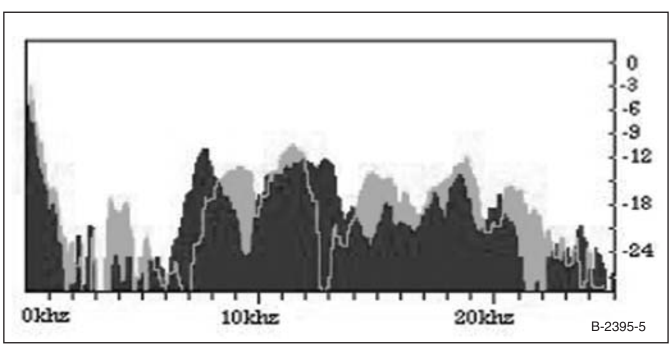

FIGURE 5: Stock decay response comparison of W3-881S drivers serial number 206 and 495. Serial number 206 decay response is drawn in green, and serial number 495 decay response is drawn in blue. Much of the difference in response between drivers is due to differences in damping of the resonant structures in the diaphragms. The dust cap accomplishes this damping in the stock model. Using the dust cap to damp the cone resonance works, just not well and just not consistently.

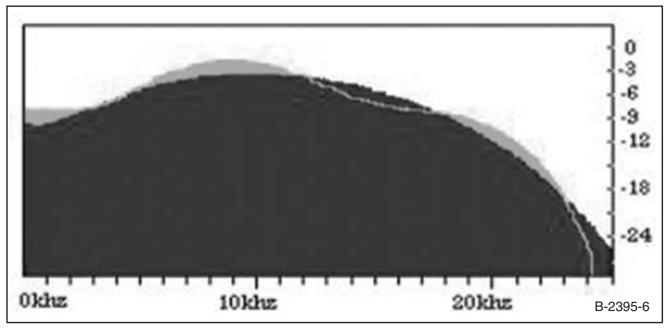

FIGURE 6: Stock Tang Band W3-881S onset response comparison graphs. The onset response of driver serial number 206 is drawn in green; the onset response of driver serial number 495 is drawn in blue. Driver variances are audible in all areas of the sound. Whether listening to percussion, strings, or voices, the drivers sound different.

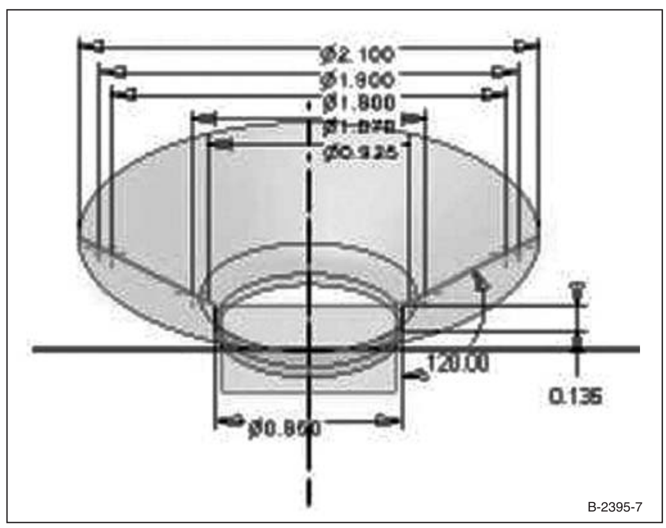

FIGURE 7: The basic cone shape of Tang Band's polypropylene 3″ drivers. Main resonance problems addressed in this modification are the cone edge termination and the inner diameter curvature of the cone. The series of diameters listed at the top of the drawing represent (in descending order) the diameter of the cone, the inside diameter of the surround, the diameter of the most outside grouping of eight dimples, the next series of four dimples, and finally, the two most inside dimples.

above that frequency. The stock driver favors the lower treble frequencies and despite its first positive to first negative spike ratio has tendencies toward sounding both harsh and grainy.

INCONSISTENCIES

In stock condition there are yet more problems with the driver. When you buy one of these drivers you never quite know what you are getting. There can be large differences in output between and among the drivers (*Fig. 3*). For example, in a very general way driver serial number 495 is the same driver as serial number 206, but in the details of its frequency response it is very different. Indeed, there is little consistency in the detail variations of output.

Figure 4 shows the response of fourteen W3-881S drivers with serial numbers ranging from 35 to 498. You could buy two drivers and have the response differ by no less than 2dB at any frequency and as much as 18dB. This is hardly what would constitute a stereo pair. This variation is not just a conventional frequency response phenomenon. It also shows up in the decay response and in the onset response (*Figs. 5* and *6*).

What causes the problems in the response and the driver-to-driver variability? It is a combination of the shape of the cone, a resonant structure problem in the cone, and how Tang Band chose to address the cone problems. The diaphragm is a slightly modified cone shape (*Fig. 7*). The cone terminates short of the center after being bent (change in slope) so that where the cone joins the voice coil, the cone surface is parallel to the voice coil former. In addition, this change in slope is constricted to just one small area of the cone surface. The change in slope is constructed by the addition of a curve or fillet to the slope of the cone.

CONE RESONANCE

To help in understanding what is happening with the cone, using any planar material, make a truncated cone (ending with a hole instead of a point at the center) and tap it. As all materials will, it vibrates and produces a sound. If you tap it from outside edge to inside edge its sound will change. This is simply because of the changing diameter. It is going to happen on any truncated cone shape.

Then add the change in the slope found in the Tang Band cones and you have produced a discontinuity structure that will cause the tap sound to change even more. This discontinuity will redirect or bounce vibrations traveling through the cone material. The basic material resonant structure problems of this cone are caused by a poorly terminated outside edge and the slope changing fillet structure near the inside edge. You can see the magnitude of this resonance and termination problem in *Fig. 8*. The frequency response drawn in black is the output of the cone without the dust cap.

DUST CAP FIX

Tang Band chose to address this resonant structure problem in the cone by attaching a hard plastic dust cap. The outside diameter of the dust cap rests right on top of the slope-changing fillet. The combination of the additional rigidity of the dust cap connecting all points of this inside resonant structure through the center of the cone and the damping qualities of the glue attaching the dust cap to the cone surface is able to smooth the driver's response.

Cone resonance is, however, chaotic. Small changes in initial conditions can lead to large differences in output. Tang Band is unable to place the dust cap or apply the glue with the precision needed to alter initial conditions in exactly the same way time after time. This is why there is so much driver-to-driver variation in response. Without the dust cap, there is still driver-to-driver variance, but the major response features are consistent in all drivers. What is needed is a better way to control the cone resonant features.

REMAKING THE W3-881S

There are three steps in remaking the W3-881S. The first is to dimple the polypropylene diaphragm. The second is to replace the dust cap with a custom-designed "phase" plug. And the third is to apply two short lines of glue to the underside of the diaphragm near the outside edge.

Polypropylene is different from most other plastic. While most plastics are amorphous polymers, polypropylene is a semicrystalline polymer. When amorphous polymers are heated, they soften and flow. When they cool, they harden. If you heat a small area of a larger polymer membrane and change its shape but do not change the thickness of the membrane, once cooled, the characteristics of the treated area will be identical to the untreated area.

In contrast, polypropylene's semicrystalline structure allows the formation of areas of greater and lesser rigidity, and greater and lesser flexibility simply by controlling the heating and cooling processes. This first modification takes advantage of this semicrystalline polymer phenomenon to change the origin state flexibility and rigidity of small sections of the diaphragm for the purpose of interfering with the formation and persistence of chaotic vibrations within the cone.

To begin, remove the stock dust cap. With a small-bladed knife, pry up the dust cap lip enough to grab with a pair of needle nose pliers. Using the pliers, pop off the dust cap. Because of the non-porous polypropylene, the dust cap will release without damage to the cone.

HOT DIMPLES

Using a heated die (*Fig. 9*), dimple the surface of the cone in 14 places. The die temperature is kept below the melting point of polypropylene with the intent of slightly softening the material but keeping it below its flow temperature. Temperatures between 260 and 280°F are ideal. Place the tip of the die against the front surface of the cone and apply pressure, perpendicular to the face of the cone. Even with the heating a fair amount of force is required to dimple the surface. This is necessary to the process because you want to alter the crystalline density within and around the dimple.

Do not be afraid to use sufficient pressure to deflect the voice coil in the gap, and even stretch and begin to wrinkle the surround. If you are concerned about keeping the voice coil centered

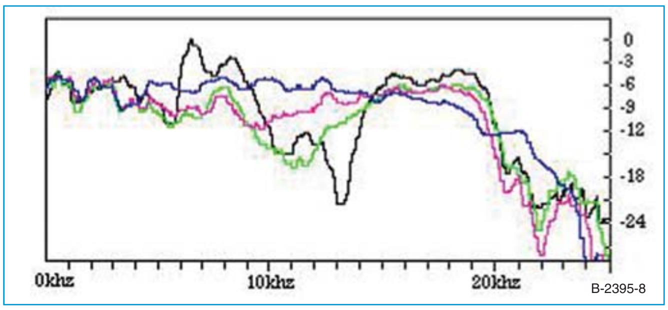

FIGURE 8: This compound graph shows the Tang Band W3-881S, serial number 206, modification sequence frequency response. The driver frequency response without dust cap is drawn in black. The frequency response with eight outer edge dimples is drawn in green. The frequency response with eight outer edge dimples and six inner dimples is drawn in pink. The frequency response with 14 dimples, two outer under diaphragm glue lines and diaphragm air mass loading regulator is drawn in blue.

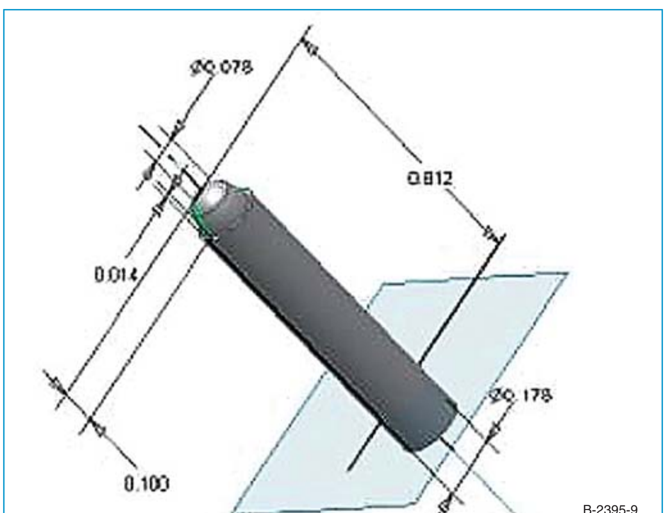

FIGURE 9: Dimensions and shape for the diaphragm dimpling die. The die drawn here was lathed from an aluminum rod. A soldering iron tip of the proper radius with the tip rounded will work as well. It is suggested to make a shallow groove to mark dimpling depth.

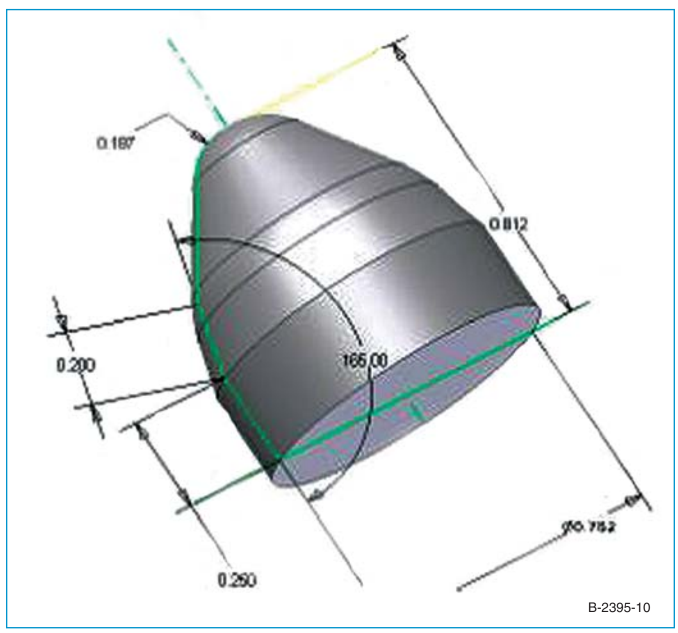

FIGURE 10: Dimensions and shape of diaphragm air mass loading regulator. The shape is specific for this driver with the specified dimpling pattern. This shape is unlikely to be optimum for other drivers.

during the dimpling process, then place four small strips of Plaid, Simply Stencil model number 28584 evenly around the pole piece in the air gap between the inside of the voice coil. The pole piece is perfect for keeping the voice coil centered.

The dimpling pattern is shown in *Fig. 7* and *Photo 1*. There are eight evenly spaced dimples around the outside edge of the diaphragm, with the outside of the dimple just touching the inside edge of the surround. Then four dimples are placed along a circle centered in the fillet, and finally two dimples are placed inside two of the four inner dimples on opposite sides of the cone. For a simple point of reference, I used the four mounting holes in the basket to reference the placement of the dimples.

RESONANCE CONTROL

The easiest way to place the dimples is to construct a ledger stick and then mark the diaphragm with a number two pencil. You can easily remove the graphite from the pencil with a little alcohol on a cotton swab once the dimpling is completed. With the inside edge of the surround as one reference point and the outside lip of the voice-coil former as the other reference point, the measurements along the ledger stick are as follows:

- 0.08″ in from the inner edge of the surround to the center of the dimple for the outside ring of eight dimples.
- 0.5″ in from the inner edge of the surround to the center of the dimple for the inside ring of four dimples.

The two innermost dimples are easy to place since you just center the dimple in the space remaining between the inner dimple and the voice coil former lip.

When the die temperature is correct and the right amount of pressure is applied, the polypropylene will take on a slightly white color. If die temperature is over 300°F and approaching or exceeding the melting point of polypropylene, the diaphragm surface will deform easily, there will be no change in color, and the polypropylene will tend to stick to the die as you separate the die from the face of the diaphragm. Deforming the diaphragm with temperatures high enough to put the polypropylene into its flow region will not produce the optimum restructuring of the diaphragm surface and will not produce the desired change in performance.

This first modification produces about 80% of the improvement. Most of the resonance structures are significantly reduced. This is shown in *Fig. 8*. The spectrum drawn in black is the starting response. The spectrum drawn in green is the response after eight outer dimples, and the spectrum drawn in pink is the response with all 14 dimples in place.

REPLACEMENT

The second modification is to replace the dust cap with a custom "phase" plug. I place "phase" in quotation marks because the operation of the plug has nothing to do with phase. You are not bouncing sound off of it and phasing it with sound coming from the diaphragm. Instead, there is a mass of air sitting against the diaphragm. Some of this air actually molecularly adheres to the diaphragm. When the diaphragm moves, this air mass moves with it as well as being compressed or rarified.

The plug sitting in the center of the diaphragm changes the volume of this air mass and also changes the volume shape of the air adhered within the cone area. This plug is actually controlling the air mass loading on the diaphragm. Change the diaphragm loading and you will change its response. Although long and truly a mouthful, I suppose that "diaphragm air mass loading regulator" is a more accurate descriptor of what this device does than phase plug.

PRESSURE REGULATOR

The shape of the loading regulator is critical. I have included a drawing with dimensions for its reproduction (*Fig. 10*). This shape has been designed specifically for this modified cone structure. The improvement using this shape of load regulator is shown by the spectrum drawn in blue in *Fig. 8*. Just as you cannot achieve the level of performance shown in *Fig. 8* using the shape of the Tang Band phase plug used in their 3″ drivers, using this shape on some other driver is unlikely to be optimal. One size definitely does not fit all when it comes to diaphragm air mass loading regulators.

The loading regulator material is not important. If delicacy is not a problem, then a polymer clay (Sculpey clay, for example, bakable at 275°F) or plaster will suffice. You can make more robust loading regulators from metal, polyester epoxy, or a molding compound like Synair's "Por-A-Kast Mark 2."

For a loading regulator produced by casting, I made a model of the regulator to all specifications (except length) on a lathe, then made a mold and cast the parts in that mold. The longer length allows you to cut the regulators to length and to cut flat bottoms on the regulators for easier mounting. Molding the parts allows for producing nearly identical regulators without a lathe duplicator.

With the four narrow strips of the stencil material around the pole piece, it is easy to position and secure the new pressure regulator. The best way to attach it is with a very small dab of gel-type super adhesive. Apply the adhesive to the top center of the pole piece. Using a small droplet sprayer, dampen the bottom of the loading regulator and press together. If you use one of the softer loading regulators, then seal the surface of the regulator with polyurethane and attach it to the pole piece with contact cement.

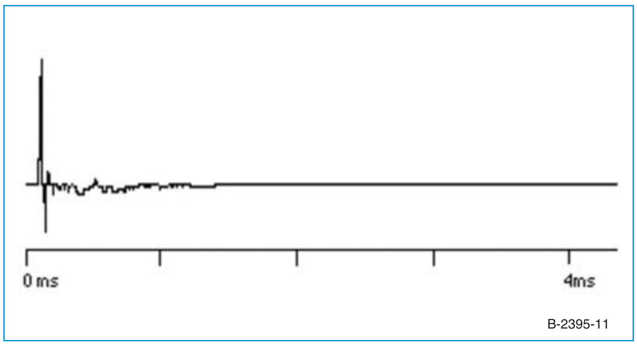

FIGURE 11: Modified Tang Band W3-881S impulse response. Peak magnitude of the decay portion is much reduced. The double "echo" of the initial positive-going spike and negative overshoot is gone. Most of the higher frequency components in the decay are gone.

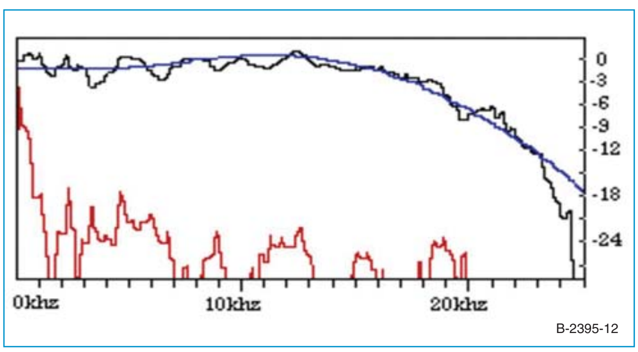

FIGURE 12: Modified Tang Band W3-881S frequency, decay, and onset response. Frequency response is drawn in black, decay response drawn in red, and onset response drawn in blue. Driver may be rated 130Hz to 19kHz ±3dB. Onset response may be rated at 130Hz to 19kHz +1.5dB and –4dB.

GLUE APPLICATION

The final modification is the simplest. It consists of applying two glue lines under the diaphragm connecting two pairs of the outer diameter dimples. With a small aperture applicator, apply two glue lines about the width and height of the underside dimple bumps. The glue line can be a straight line, and you can choose any of the nonadjacent dimples.

For the sake of symmetry I usually choose dimples on opposite sides of the diaphragm. Any of the water-soluble aliphatic glues will work. PIC "Flex-White Glue," Beacon "Gem-Tac," or Crafter's Pick "The Ultimate" will all serve the intended purpose.

PERFORMANCE

These three modifications make the W3-881S a much better driver (*Fig. 11*). The double echo in the decay response is gone. After the first three spikes of the onset response, decay magnitude is significantly reduced. Most of what remains is the low-frequency motor system resonance. And, although you have increased the low to high treble balance with relatively more high-frequency output in the remade driver, the first positive to first negative spike ratio is almost the same.

Figure 12 shows an overlay of the frequency response, decay response, and onset response of the modified driver. All spectra are much improved in comparison to the stock driver. There are now simply fewer anomalies to get between the listener and the music being reproduced by the driver. It is now valid to claim a frequency response of 130Hz to 19kHz, ±3dB.

Figure 13 shows a comparison of before and after the remaking of serial number 206. In the remade driver the octave-to-octave balance is better, and at frequencies over 7kHz, the reduction in decay contribution is as much as 20dB. While the reduction in resonant problems and smoothing of the frequency response is sufficient to make the driver remaking worthwhile, the greatest improvement is in driver-to-driver variance.

UNIFORMITY

This type of graph, overlaying the frequency response of multiple drivers, is almost never published. Why? Possibly because people find it easier to assume that every driver is much the same as any other driver than it is to select and design for drivers of the same make and manufacture that might not be that close to one another in performance. Yet, if it were the fact that there is little driver-to-driver variance, why would any manufacturer provide matched pairs, and why would they charge so much more for matched pairs?

Finally, *Fig. 15* shows a frequency-response comparison of driver serial

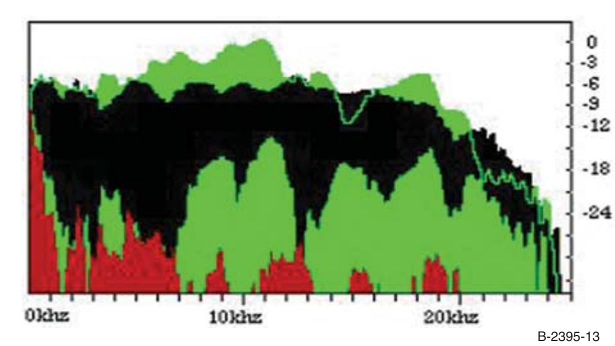

FIGURE 13: Stock and modified frequency response and decay response comparison of W3-881S driver serial number 206. Stock frequency and decay responses are drawn in green. Modified frequency response drawn in black, and modified decay response is drawn in red (for contrast). Not only is the modified frequency response much flatter, above 7kHz, the decay response has been improved by as much as 20dB.

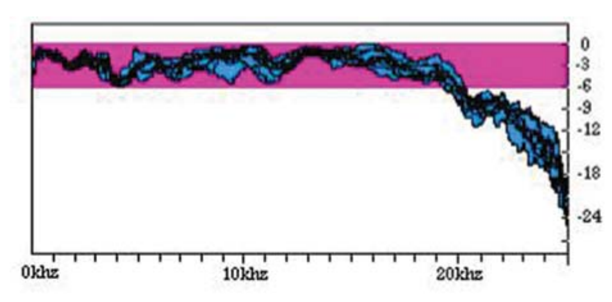

FIGURE 14: Frequency-response comparison of 14 modified Tang Band W3-881S drivers. All drivers can be rated ±3dB from 130Hz to 19kHz. Driver to driver variance is no more than 6dB, with most significantly better than 6dB.

number 208 and 495. Both drivers are within ±3dB from 130Hz to 19kHz, and the drivers are matched to one another within 1.5dB. In contrast to the stock condition, it is now possible—perhaps even practical—to select matched pairs of drivers.

In addition, with the modification, not only is it practical to produce matched pairs, it is possible (although still much more difficult) to match drivers in octads. Matched octads would allow for closely matched stereo loudspeakers utilizing four drivers each in a closely spaced linear array. And with a small manipulation of the baffle, it would be possible for each four unit linear array to transiently sum at the listening location.

THE SOUND OF THE REMADE W3-881S

The sound of the remade driver is very easy to describe: clear, clean, open, and detailed. Other than those descriptors, the sound of the system is mainly the sound of the music playing through the system. Soundstage, tonal balance, instrumental and vocal presence all change with every change in recording. And on music intensely altered and processed ("FX"ed) in the studio, the soundstage, tonal balance, and instrumental and vocal presence change even within a single song.

Peter Gabriel's CD, *Up*, first track, is a great example of how much detail can be revealed by a transiently coherent loudspeaker. Yet, *Up* sounds completely different from Enya's *The Celts* or *Celtic Moods*, also filled with intensely studio-constructed sounds, paired, overlaid, and alternated with very naturally recorded sounds. Even when Enya is repeatedly overdubbing her own vocal, through a pair of the modified W3-881S, it is not just a wall of voice; you can begin to hear separation between the overdubs.

Pop, classical, new age, or jazz music—it is possible, through listening to a wide range of material, to come to the conclusion that the loudspeaker is hiding very little of the recording. Some of the music you will rediscover and come to love for the totality of its sound. In other cases, however, while you will still love the music, you will not love its sound. But whatever music and whatever sound, you will come very close to hearing no more and no less than what was recorded.

THE BOTTOM END

Since cutoff for the W3-881S is 130Hz, a loudspeaker using this driver does require a woofer and a crossover. If you chose a frequency of 180Hz for the crossover, the low end of the loudspeaker system can be easy to design. With 180Hz as the crossover frequency, even a subwoofer can serve as the woofer.

With the response of the W3-881S rising slightly below 200Hz, all that is required for the W3-881S is a first-order crossover. Set the knee point between 150 and 180Hz. Then match it to a 120Hz third-order low-pass filter feeding the woofer.

With a first-order high-pass filter between 150 and 180Hz, the W3-881S motor resonance seems to influence the sound very little. I do not hear a need for a series notch filter wired in parallel with the driver and tuned to the resonant frequency. If you use a passive, speaker level crossover, however, the series notch filter will increase the stopband performance of the high-pass capacitor. Set the levels between the woofer and nearly full-range driver correctly and, with or without the notch filter, the sound will be seamless. ❖

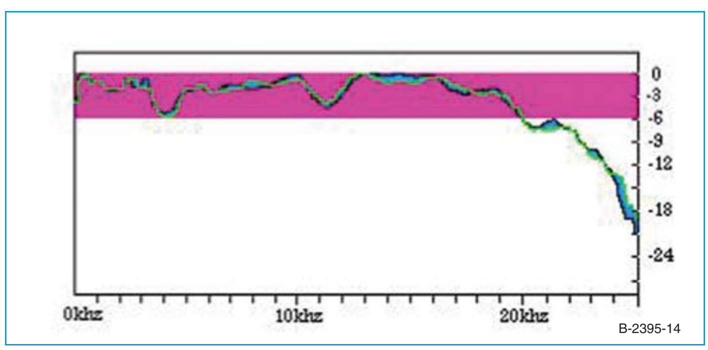

FIGURE 15: Frequency-response comparison of Tang Band W3-881S drivers serial numbers 208 and 495. Drivers not only meet a frequency-response rating of 130Hz to 19kHz ±3dB, but drivers can be matched to within 1.5dB.