

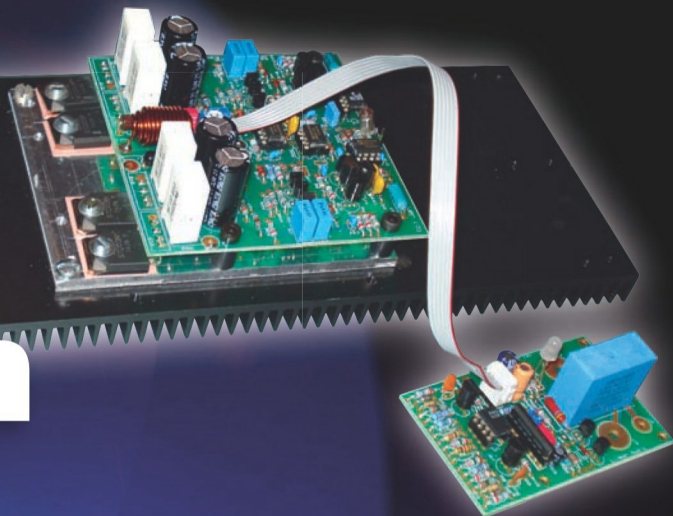
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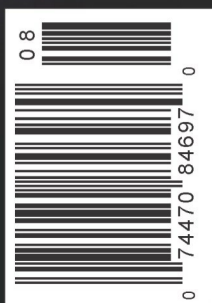
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paX, Part 1

Introducing an experimental error correction power amplifier. The first of this two-part series presents an output stage with error correction.

Most solid-state power amplifiers employ some form of global negative feedback to reduce nonlinearities and output impedance. In some cases, designers exploit alternatives such as feedforward to circumvent perceived disadvantages of global negative feedback. The present design uses error correction as (re) defined by Malcolm Hawksford in 1981¹. In part 1 of this article I introduce error correction for audio amplifiers and present an audio output stage based on error correction. Part 2 will extend the principle to the voltage amplifier stage and present a complete error correction power amplifier.

THE POSITIVE SIDE OF NEGATIVE FEEDBACK

Negative feedback (nfb) very often gets a bad rap from audio pundits. Comments such as “it cannot correct after the fact” or “it slows down the

amp” pop up again and again. When pressed for arguments, these are either not forthcoming or fly in the face of well-established engineering principles.

Negative feedback, as a general principle, is one of the most powerful tools available to the designer to build amplifiers that are transparent to the signal they amplify. With that I mean that they do not add or subtract anything to the input signal. In reality, circuits are never ideal, but the changes to the signal can be made so small that inaudibility of the changes is pretty much guaranteed. Negative feedback is also fully understood, and although bad-sounding feedback amplifiers are still being sold, it is totally unnecessary. So, you may ask, why bother with error correction (EC)?

For one thing, it is a different road, and the trip is often more enjoyable than the destination. Second, although you will see that EC is in many rec-

spects another face of nfb, there are some interesting differences and different challenges, leading to better understanding of the processes inside any feedback amplifier.

WHAT THE H.EC?

Let me start by narrowing down what I mean by EC in the context of this article. I refer specifically to Malcolm Hawksford’s paper¹. (I will use the term “H.ec” to refer to Mr. Hawksford’s topology, and “EC” to refer in general to error correction.) **Figure 1** is the basic topology from that paper. You see an amplifier block N, to which error correction will be applied. The difference between the input and output of that amplifier is either added to the input via block A (error feedback), or to the output via block B (error feedforward). All the summing and differencing blocks have a gain of 1.

Of course, it is important that the error correction signal is added in just the right proportion to cancel the original error. In this design I use the correction via block A, added to the input. I will further assume that the amplifier block N is an output stage from an audio power amplifier with a gain of almost, but not quite, 1. The object here is to make this gain, as seen from V_{in} , exactly 1, independent of output load, frequency and signal level. If I succeed, I will have an ideal output stage.

Of course, I won’t get that far, but as you will see, I can get pretty close.

Figure 2 shows the conceptual circuit. The gain of the output stage you want to linearize is “about 1.” In prac-

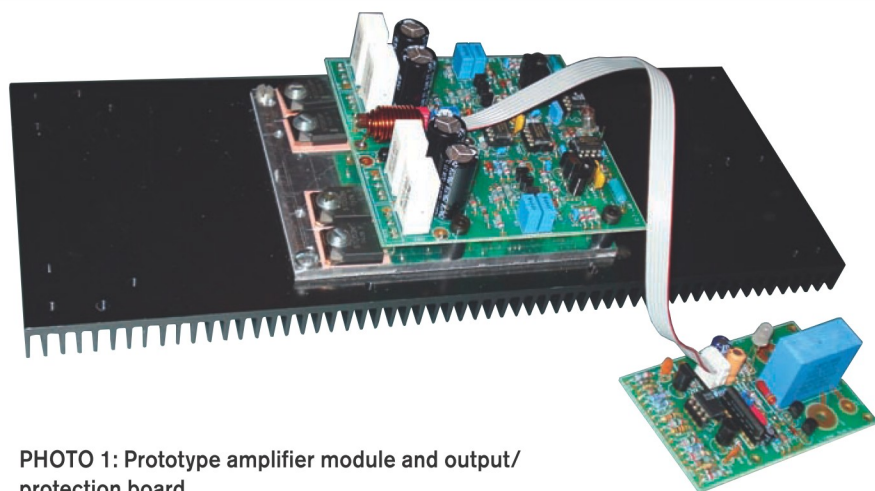


PHOTO 1: Prototype amplifier module and output/protection board.

tice, most output stage gains are anywhere between 0.92 and 0.98, depending on the topology, loading, signal frequency, and so on. You see immediately that V_c , the signal that is fed back to the input, is $V_{out} - V_e$, where V_e is the effective input signal to the output stage. You can now calculate V_e as $V_e = V_{in} - (V_{out} - V_e)$. You also know that $V_e = V_{out}/A$, and if you plug that into the former equation, and rearrange terms, you get: $V_{out} = V_{in}$.

What is significant here is that the actual amp gain A is no longer part of the equation. Whatever the shortcomings, nonlinearities or errors of A , I got rid of all of those. This is the ideal output stage!

If you are familiar with practical audio amplifiers, you probably are feeling a bit uneasy now. An ideal power stage? That would be the first one ever! And you are right—in reality, you can't reach that ideal. One reason is that you assumed that the summers $S1$ and $S2$ in Fig. 2 are ideal summers, without flaws. That cannot be. Those summers consist of passive and (most

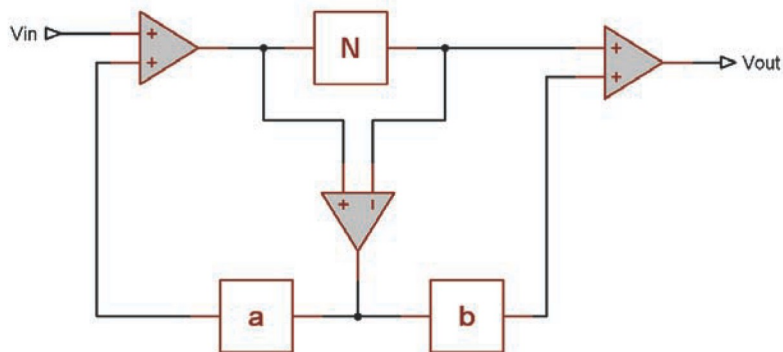


FIGURE 1: Hawksford's general feedback/feedforward/error correction structure.

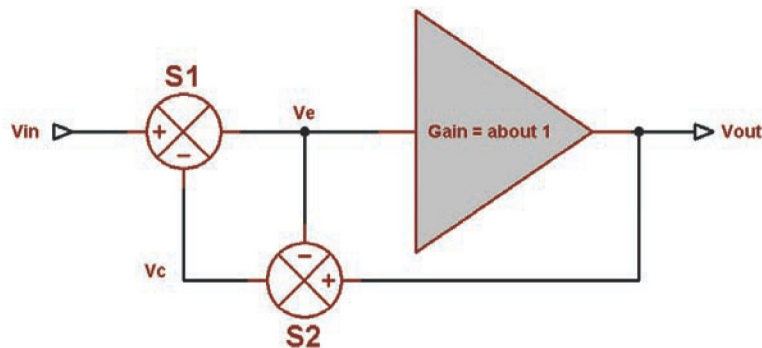


FIGURE 2: Error correction concept.

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probably) active devices that have their own nonlinearities, so the basic accuracy will not be ideal. Their characteristics will also vary with frequency, so the correction accuracy will vary with frequency.

Because the output stage gain will also vary with frequency and load, the amount of correction required will vary with frequency and load, meaning that signal levels in the summers vary with frequency and load as well. This further leads to accuracy limitations. Nevertheless, it is possible to greatly improve the performance of the output stage with relative simple means, as you will see.

THERMAL MEMORIES

There was one other goal I had in mind for this amp. In most power amps, the thermal bias compensation is obtained by mounting a transistor in the bias circuit on the same heatsink as the output devices. In that way, if the output devices heat up and start to draw more current, the bias transistor also heats up. That causes the

bias voltage to decrease, and the object is to dimension this thermal feedback loop such that the bias current in the output devices remains stable with temperature. But because it takes time for the heatsink to heat and cool and transfer this changing temperature to the bias transistor, this loop reacts relatively slowly, in terms of several seconds or more.

The first time I read about this was in an article by French designer (audio, not clothes) Hephaïstos². He realized that the dissipation levels in the output and driver devices vary with signal level variations, and that they are much faster than the reaction time of the thermal bias compensation circuit. When a high level signal burst appears, the devices' operating point would shift, and would return to the earlier state only seconds after the high level burst had disappeared. Such a burst is too short for the thermal feedback to adjust the bias. A smaller signal after the burst would thus be amplified with different operating conditions than before. He

called this thermal or memory distortion.

In my design I wanted to get rid of this problem as well. Therefore, I selected Sanken STD03N and STD03P devices for the output stage. These are Darlington's, with a bias diode integrated on the transistor chip. By using this diode in the bias circuit, it can track thermal cycles in the output devices almost instantaneously, thus hopefully eliminating thermal distortion. Because it is a Darlington, any (pre) driver dissipation would also be low enough to avoid memory distortion.

There's one catch: the on-chip sense diodes need to be run with a specific current to make their thermal bias changes (in millivolts per degree) equal to the V_{be} changes of the driver and output devices in millivolts per degree. This requires a different bias circuit from the usual V_{be} multiplier.

ERROR CORRECTION BASICS

Figure 3 shows the basic topology of the output stage without error cor-



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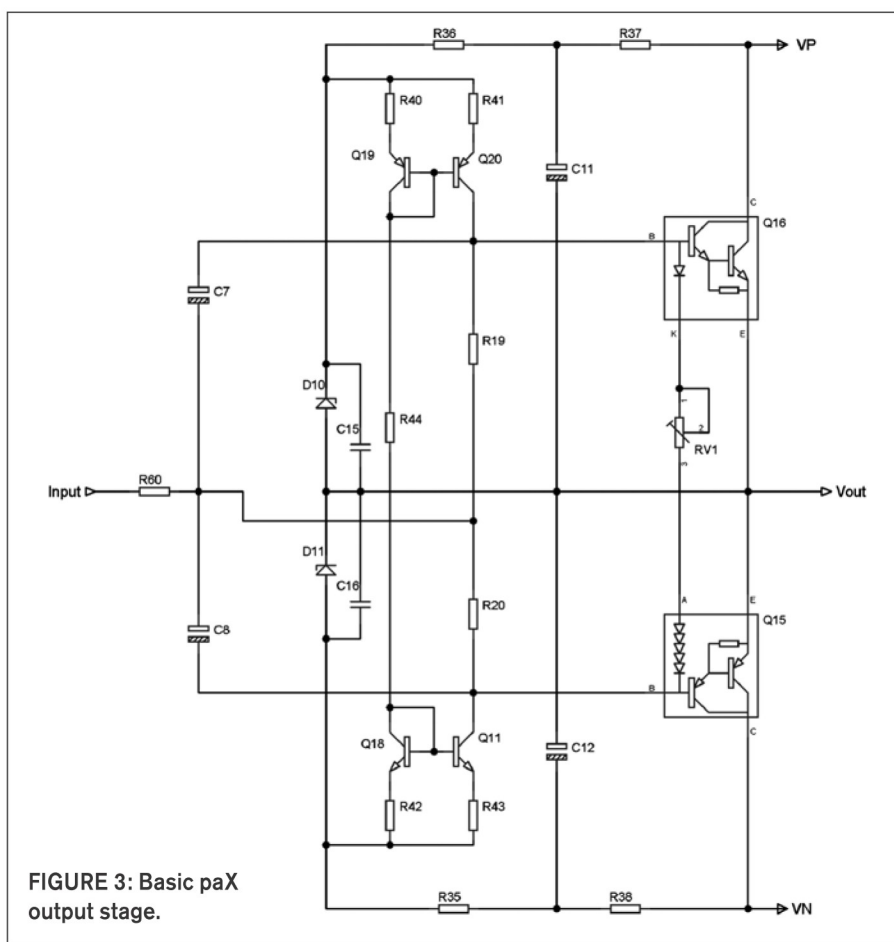


FIGURE 3: Basic paX output stage.

rection. You will notice the integrated bias diodes in the output devices. To get a precise tempco (temperature coefficient) matching between these diodes and the output Darlington's b-e junctions, Sanken's engineers used a string of five Schottky diodes in the P-device and a single silicon diode in the N-device. The datasheet specifies a bias current of 2.5mA through the diodes, and this current is set by the current mirror formed by Q19, Q20 and Q18, Q11. The current is set with R44 and mirrored into the diodes.

To keep this current stable the supply voltage for the current mirrors is regulated by zener diodes D10, D11 to 15V. The bias for the zeners is coming from the main supply via R36-R37 and R35-R38. To keep *that* supply stable, the junction between those resistors is bootstrapped from the output via C11-C12. The result is that with varying output signal levels, the zeners can always provide a constant supply voltage for the current mirrors.

The datasheet for the output devices also specifies a quiescent current of

40mA through the output Darlington for best thermal tracking. Although the diodes track the changes in the Darlington Vbe quite well, the absolute value of the diode threshold values varies from unit to unit. Thus, a way to adjust it is required, and that is the purpose of RV1, which you should set for 40mA through the output stage.

OK, so now you have a nice and stable output stage, but you need to drive it with a signal. That is done through R60 to the junction of R19-R20.

Suppose that the input signal goes positive: the junction of R19-R20

goes positive so there will be less current through R19. Because the current coming out of the collector of Q20 is fixed, there will be more current into the base of the Darlington and the output signal will also go positive, following the input signal. In this case there will also be more current through R20 so less from the base of the P-device. This will work quite well for DC and low frequencies and nominal, resistive (8Ω) loads, where the Darlington's have a very high gain and the current mirrors are almost perfect.

However, with increasing frequency and increasing load currents, the Darlington's need more input current, so capacitors C7 and C8 are added to bypass R19-R20 so the input source can directly drive the output devices. And in case you wonder why R19 and R20 are there in the first place, you need them to set the DC level at the output to zero, to avoid DC offset going to your speakers.

This output stage is quite simple, reasonably linear, and satisfies the requirement for accurate temperature tracking of the instantaneous output device dissipation. The next step is to wrap error correction around it. (Because the stage has no excess gain, you cannot use a global nfb loop around it. The nfb loop that normally includes the output stage of a feedback amp, of course, relies on the excess gain in the voltage amplifying stage to work its distortion-reduction trick.)

THE CURRENT CONVEYOR

For EC you need to derive the difference between the input and output signal, and add that difference to the input signal (Fig. 2). Subtracting or

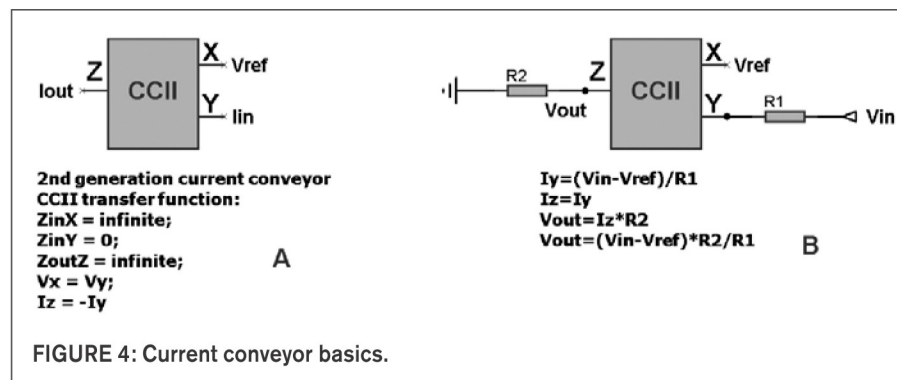


FIGURE 4: Current conveyor basics.

adding two signals that are of similar level is trivial with op amp circuits. You can use one op amp (S2) to subtract Vout from Vin, and another (S1) to add the resulting Vc to Vin. But in the spirit of EC, I didn't want to use an obvious high global feedback element in this circuit.

After a lot of head scratching and many Internet searches, I came up with the idea to use a current conveyor³. The basic circuit is shown in **Fig. 4A**.

This is a very interesting circuit. Basically, whatever you send into the Y terminal comes out in the opposite direction from the Z terminal, which works as a current source. Hence the name current conveyor: the current at the input is conveyed to the output. This particular type is called a 2nd generation conveyor, generally shown as a "CCII." Terminal X is a reference terminal for the current input, and the voltage at Y will be kept the same as that at X. What is this good for?

As an example (**Fig. 4B**), if you connect a signal to Y via a resistor, you know that the voltage across that resistor will be the signal minus Vx (remember, $V_y = V_x$). So the current into Y is this voltage across the resistor divided by the resistance. That same current comes out of Z, so you have now converted a signal level to a current that can be converted back to a signal via another resistor. If the second resistor on Z is twice the input resistor on Y, you have a gain of two. You can set the gain of this circuit with just one resistor ratio (**Fig. 4B**).

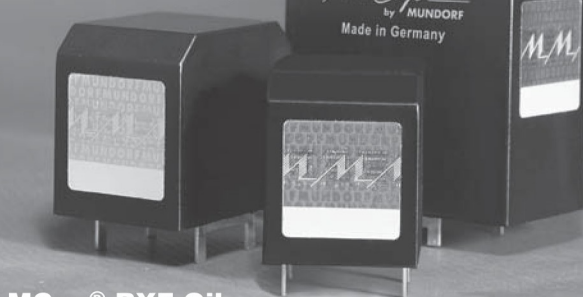
The nice thing here is that you can accurately set a gain (or attenuation, if that would be required) without needing to resort to nfb. The current conveyor does not use global feedback; it is an open loop circuit. (Current conveyors consist internally mainly of current mirrors; and it can be argued that current mirrors use 100% local feedback. But I won't get into that discussion here!)

There is our summer: I will use equal resistors for a gain of one. The error correction loop around my output stage is shown in **Fig. 5**. The error between the input and output appears across R34 and develops a current that is "conveyed" to R25. This adds the correction signal to Vin to make the H.ec work as in **Fig. 2**. The positive feedback loop in **Fig. 2** from Vc through Vc back to Ve is realized with the connection of Z to X in **Fig. 5**.

So, at this point, you need to become practical: where do you find this CCII thing? There are CCII's around, often designed in low-voltage CMOS IC technology, and mostly used in laboratories in university settings. They might as well be made from unobtainium. But, interestingly, CCII's are part and parcel of current feedback op amps and so-called "diamond transistors."

Figure 6 shows the simplified diagram of the internals of an AD844 current feedback op amp. Recognize the CCII terminals: pin 3 is the high-impedance X terminal, pin 2 the low impedance current input Y, and pin 5 labeled Tz is the current source output Z. There is an open loop buffer stage between Tz and the output at pin 6. You can readily see that whatever current is injected into pin 2 (Y)

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FIGURE 5: Output stage with error correction.

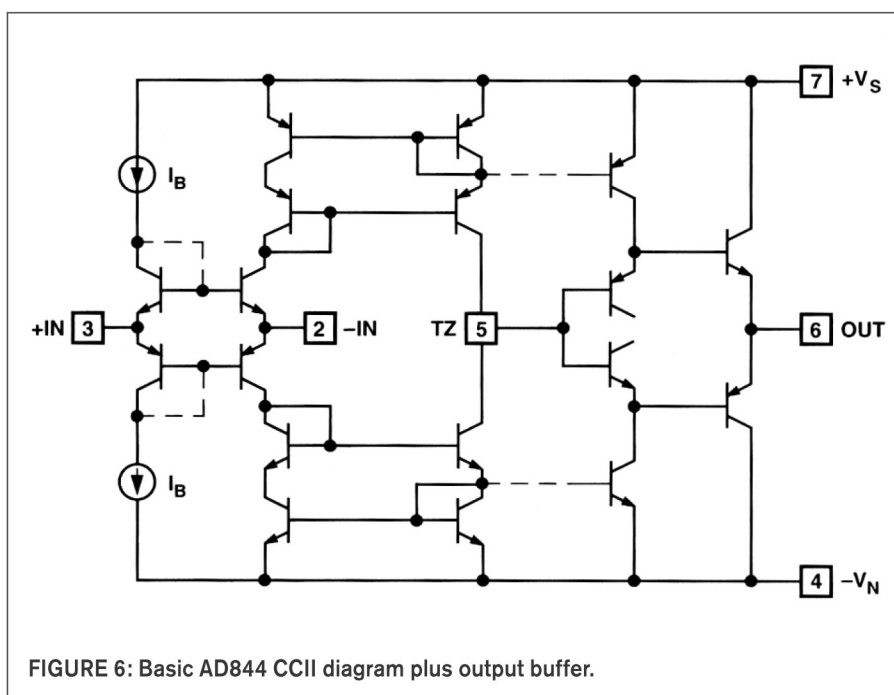
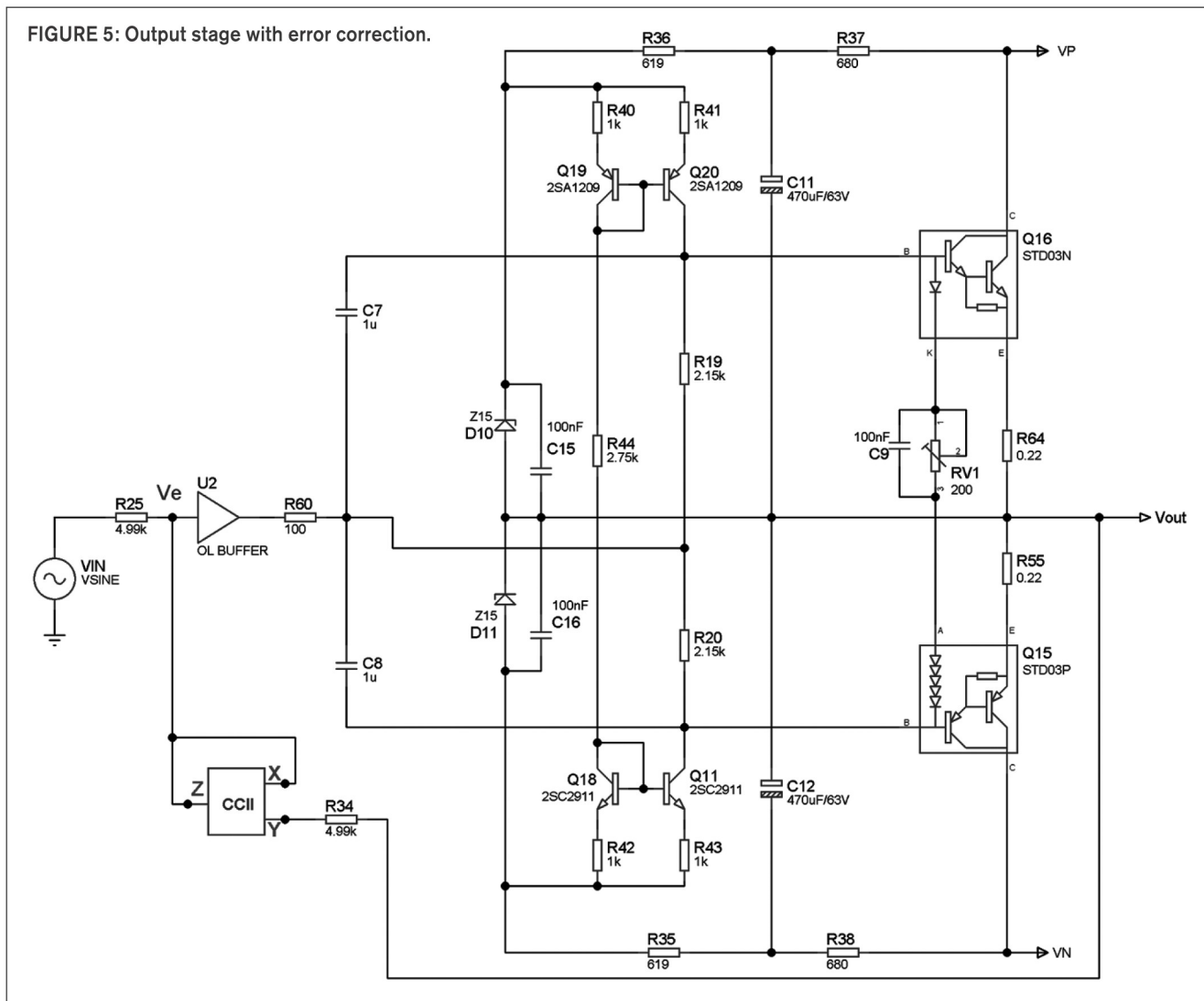


FIGURE 6: Basic AD844 CCII diagram plus output buffer.

comes out at pin 5 (Z), but in the opposite direction.

It seems this part has been custom designed for this! (Other similar parts are Maxim's MAX435 and MAX436 (now obsolete) and TI's OPA860, which superseded the OPA660 and OPA2660.) You can use the input stage for the CCII function, and the buffer to drive the output stage to isolate the input summer from the output stage's nonlinear input impedance. You see that the current output terminal is already internally connected to the buffer input—again, just what you need!

Figure 7 shows the full output stage circuit, with component values. Note that **the amplifier input is at pin 5 of the AD844, not at pin 2 or pin 3** as you would expect in a classical op amp circuit. This is *not* an op amp circuit,

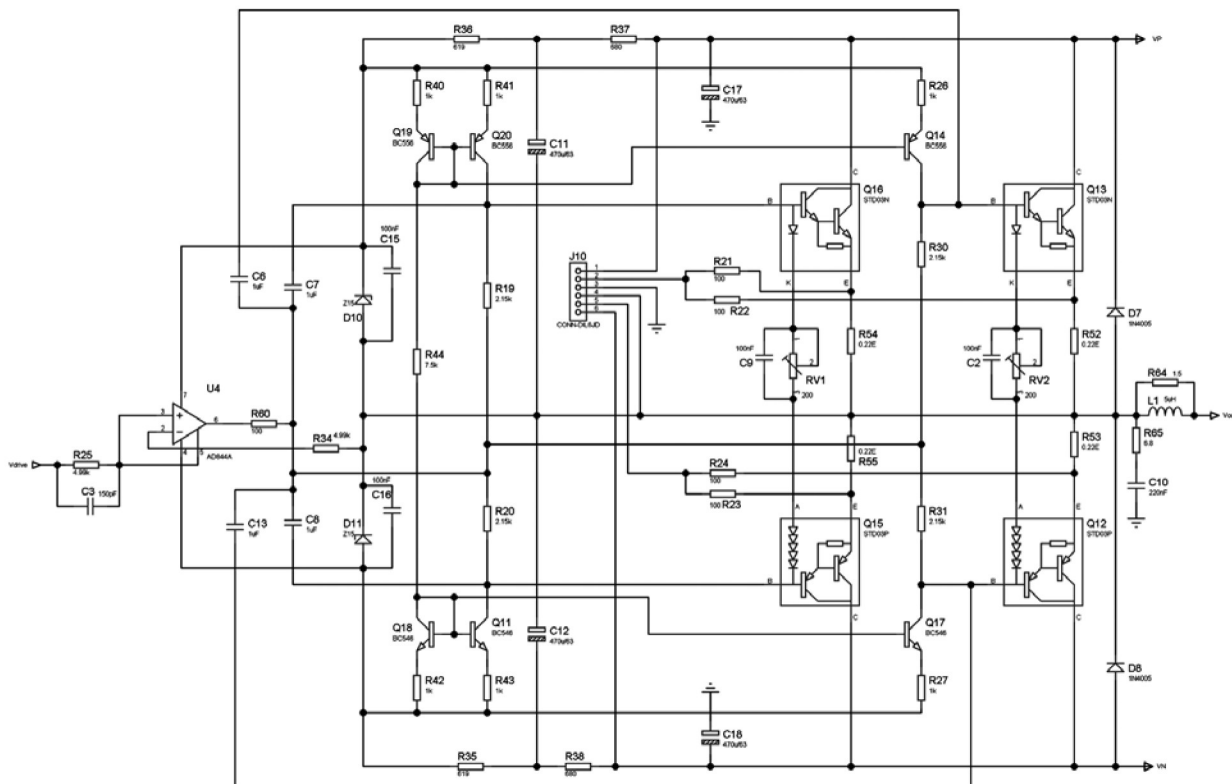


FIGURE 7: Full gain-of-one output stage with Hawksford error correction.

but I'm sure many people will be confused by this.

PUTTING IT ALL TOGETHER

This output stage has two pairs of output devices. The design goal was 100W in 8Ω and 200W in 4Ω. For a pure resistive load, one pair of devices would have been sufficient. However, speakers are not purely resistive; depending on the signal frequency they can act capacitively or inductively, especially if there is a complex cross-over filter. This leads to phase shifts between the output voltage and output current, so you can encounter the situation where the output voltage is negative, but with the current coming from the positive side (the N-device). The N-device will then have a quite large V_{ce} , and the current it is allowed to source is much smaller with a large V_{ce} than what you would think from the allowable dissipation.

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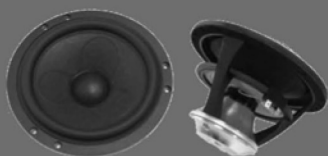
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There are a few extra components in the circuit that I haven't mentioned yet. R60 is a small resistor in series with the AD844 buffer output stage. A large part of the output drive goes via the two capacitors C2 and C4 to bypass the current mirrors. R60 isolates the buffer output from capacitive loads ensuring stability. Another capacitor—C3—is placed across R25.

As the frequency increases, the loop through the output stage as well as the current conveyor will exhibit phase shift. In a "classical" feedback amplifier, if the phase shift becomes too large, it will turn the nfb into pfb, which, as you all know, may lead to instability and even oscillations. In H.ec this can also happen, so you need to roll off the loop gain for higher frequencies. C3 does just that by decreasing the effective correction impedance (R25//C3) with increasing frequency. Also, remember that this stage needs to be driven from a low impedance source, because the source output impedance forms part of the EC scaling resistor R25.

Finally, there is the 6-pin connector J10 and some associated resistors.

This is the connection to the protection board. It provides the Vce and Ic related information of the output devices to the protection circuitry. This output stage is pretty linear as attested to by the curves in Fig. 8. Note that this is without classical negative feedback around it, and it can perfectly stand on its own when driven by a suitable voltage amplifier (Vas) stage.

So, let me take a break here; I'll attack that Vas, and the power supply, in the next installment and develop a full-fledged, high-quality audio power amp. Stay tuned. *aX*

REFERENCES

1. Hawksford, M.J., "Distortion correction in audio power amplifiers," *JAES*, Vol. 29, No. 1/2, pp. 27-30, Jan/Feb 1981.
2. "Thermal Distortion—it exists, I've seen it"—Hephaistos, *L'Audiophile*, nr 32, May 1984.
3. In the early 90s, an IC designer, Doug Wadsworth, designed a current conveyor for audio on his own money (PA630). The chip, the "Swift Current" chip, eventually found its way into Wadia DACs. It is no longer available for other parties, but I had bought some from him and knew they were a well-kept secret for hi-end audio.

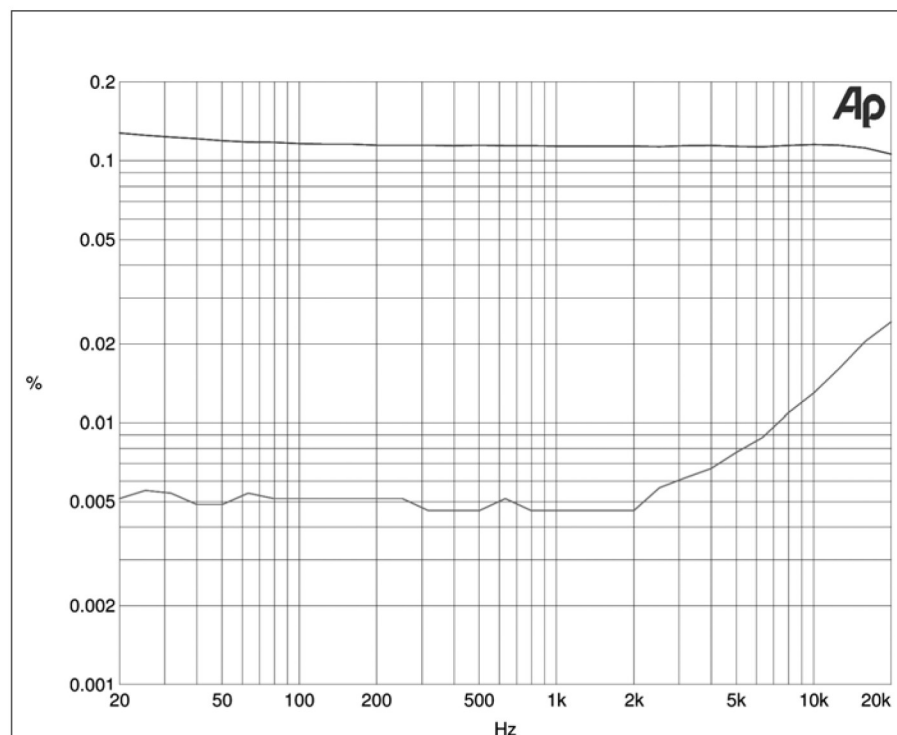


FIGURE 8: Output stage THD versus frequency with and without H.ec. Vo = 10V RMS at 1kHz.

How H.ec Differs from Global Negative Feedback—or Not.

Error correction works by feeding back (part of) the output signal to the input of the amplifier. It has many similarities to a classical negative feedback loop. But how is it different?

TWO SIDES OF THE SAME MEDAL

Figure A is the basic H.ec topology. You can identify two feedback loops: one positive feedback loop via V_e to V_c and back to V_e ; and a negative feedback loop from the amplifier output to V_c . That particular arrangement makes H.ec so interesting. Assume that the output stage A is ideal and has a perfect gain of 1. In that case, the correction signal at V_c is 0, because there is no difference between V_e and V_{out} .

You could just throw away the whole error correction part. But if the gain A is just a bit less than 1, V_c is a negative signal which is subtracted from V_{in} , and the effect is that V_e increases. This works as a positive feedback loop. In the case where the gain A is larger than 1, the signal at V_c is a positive signal, which will be subtracted from V_{in} to decrease V_e . In this case, the system works as a negative feedback loop.

So, what you see is a combined feedback loop that can act either as a positive or negative feedback loop, or not at all, depending on the error of the main amplifier block. It looks as though the phase and level of the loop adjusts itself as needed.

The validity of treating a combined feedback loop by “summing” the individual loop contributions has been established by several researchers in the past. Gerald Graeme, at the time analog IC development manager of Burr-Brown, has written several articles about it^A. If you combine the two feedback loops in a single loop, as is often done to make analysis easier, you get **Fig. B**. Again, you can see from the formulas that if the output stage gain $K = 1$, there is no feedback at all. For comparison, **Fig. C** shows the global nfb topology for a unity-gain amplifier.

STABILITY ISSUES

Mention positive feedback (pfb), and most designers will immediately worry about stability. Indeed, positive feedback is what makes an oscillator sing. But pfb is not enough to make a circuit oscillate or unstable. There must also be a loop gain that is greater than 1 to sustain the pfb loop.

In **Fig. A**, it looks as though the pfb loop gain is 1: from V_e through B and the unity gain summer S1, and I defined B as a unity gain summer as well. So, the knee-jerk reaction is: if B rises only a bit above 1, the whole thing oscillates! But, counterintuitive as it may be, this is not the case at all, because it only looks at one part of the total feedback loop. You must

Now you all know that in a real amplifier both K and B will be frequency dependent and show low-pass behavior. So, look into that as well. Assume, as a first approximation, that both K and B have two poles, at $\omega_1 = 5 \cdot 10^6$ radians and $\omega_2 = 5 \cdot 10^7$ radians. In this formula for G_{cl} , you then need to replace B and K with the following expressions:

$$B_{\omega} = B \left(\frac{1}{(1+j\frac{\omega}{\omega_1})} \right) \left(\frac{1}{(1+j\frac{\omega}{\omega_2})} \right)$$

$$K_{\omega} = K \left(\frac{1}{(1+j\frac{\omega}{\omega_1})} \right) \left(\frac{1}{(1+j\frac{\omega}{\omega_2})} \right)$$

Now you call MathCAD to the rescue and ask it to plot the frequency response of the closed loop gain with these substitutions (**Fig. D**).

There is a trifle of peaking, a dB or so, but otherwise it looks very well behaved. Of course, there is much more to this, which is beyond the scope of this article, but at least you should have gotten a flavor for this very interesting topology.

A. “Generalized op amp model simplifies analysis of complex feedback schemes”—Gerald G. Graeme, *EDN*, 15 April 1993, pp. 175.

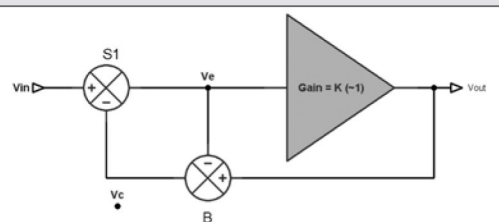


FIGURE A: Basic error correction topology.

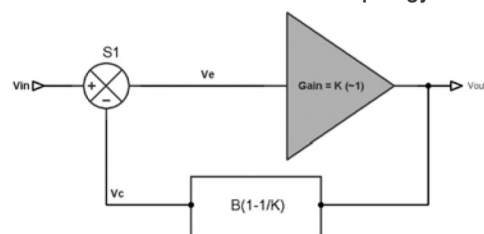


FIGURE B: Two feedback loops are combined. The two figures are equivalent.

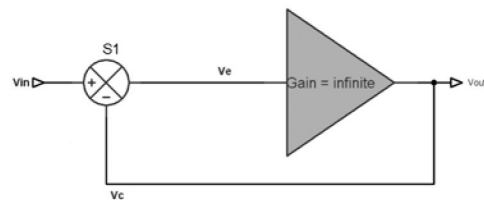


FIGURE C: Classical global nfb amplifier for a gain-of-one amplifier.

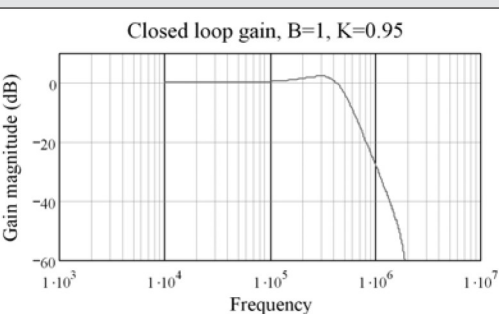


FIGURE D: Closed loop gain frequency response.

look at what part of the output signal is fed back, and that part is $B^*(1-1/K)$. If $B = 1$ and K is, say, 0.95 for an EF output stage, the pfb gain is only 1/20! So, B can even be 2 or 5 or 10 and it would still not oscillate! Isn't that interesting?

If you calculate the closed loop gain G_{cl} of **Fig. B**, you get:

$$G_{cl} = K \left(\frac{1}{(1+B(K-1))} \right)$$

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Here are construction plans for a power supply that you can use to test preamplifier and small power amplifier circuits with vacuum tubes.

A regulated power supply is a very useful device for the home constructor. Using a benchtop supply, you can fire up novel tube projects and test the circuit before you invest in a dedicated (in-built) power supply. You can also use the regulated supply to provide existing amplifiers or receivers with a clean plate voltage. A regulator can improve the reproduction of sonic details because it suppresses hum, noise, and other forms of interference on the supply rail.

My wish list at the start of the PSU project looked like this:

1. Output voltage should be in the range of 240 to 300V (suitable for most DIY tube circuits).
2. The output current limit should be at least 100mA, to allow experiments with (small) power amplifiers.
3. The 300V supply should be a tube circuit (for nostalgia's sake and for the sake of sturdiness).
4. In addition to the powerful 240 to 300V supply, a small regulator should provide a variable (negative) bias voltage.
5. A heater voltage supply should also be included. I decided to incorporate a 6.3V DC supply of maximally 5A.
6. Most electronic parts should come from the junkbox.

HIGH-VOLTAGE REGULATOR

After examining some literature on voltage regulators¹ and inspecting the inventory of my attic, I decided to build a circuit proposed by RCA as late as 1975². The schematic is shown in Fig. 1. This regulator, which can supply

maximally 225mA at 250V, met goals 1 and 2.

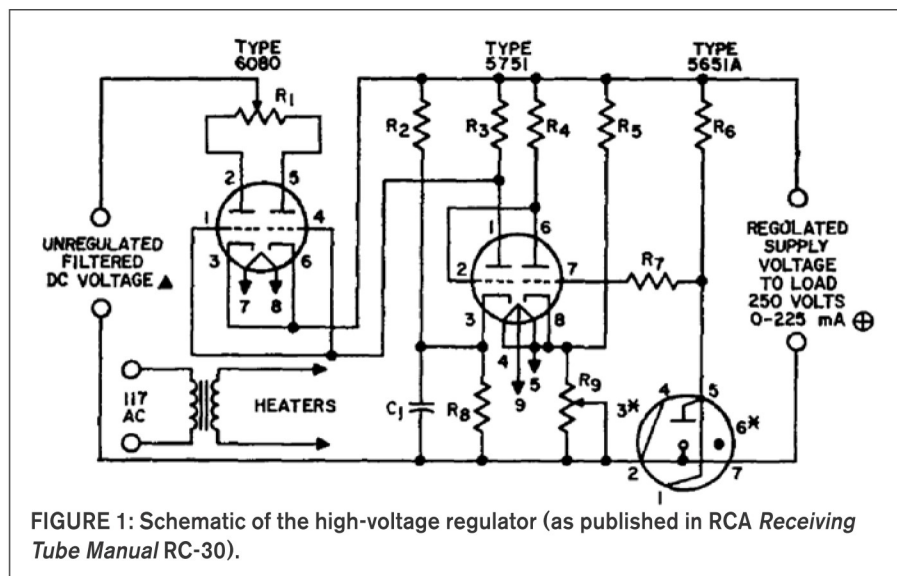
RCA employed a 5651A tube (voltage reference), a 5751 dual triode (error amplifier), and a 6080 dual triode (series pass element). I did not have these tubes in my junkbox, but I found useful equivalents: a 85A2 from Mullard (new old stock), a 12AX7 from Sovtek (brand new), and a 6AS7G from RCA (new old stock). Appropriate ceramic sockets (one 7-pin miniature, one noval, and one octal-type) were also present (leftovers from the Royal Dutch Navy).

Because I did not have an appropriate plate balancing potentiometer of 160Ω (R1 in Fig. 1), I used a rheostat of 100Ω and 10W and soldered two resistors of 33Ω 5W at both ends of the wiper track, so the total resistance between the two plates of the 6080/6AS7G became 166Ω.

VOLTAGE DOUBLER CIRCUIT

RCA specifies that the unregulated DC at the input of the regulator should be 375V at zero load current and should not drop more than 50V at maximum load (225mA). I didn't have a suitable transformer from the tube era, but I discovered a power transformer with many secondary voltages in my attic which I had salvaged from a damaged solid-state amplifier. The size of the transformer core suggested that it could supply at least 250W, and it included a secondary winding of 140V with a center tap (i.e., 70-0-70V). I combined this junkbox transformer with a voltage doubler circuit to generate the appropriate unregulated DC for the voltage regulator.

The schematic of my voltage doubler is shown in Fig. 2. The output voltage of a doubler drops more under load than the output of a bridge rectifier. How-



ever, with the hefty transformer and the big reservoir caps which I employed, the input voltage to the regulator is maintained within the margins specified by RCA. Other disadvantages of a doubler are: (1) there is more ripple on the raw DC than with a bridge rectifier, and (2) the ripple frequency is 50Hz (in the US, 60) rather than 100Hz (US, 120). This is the price you pay for using junkbox parts.

I included resistors R1 and R2 (Fig. 2) to limit the current surge through the rectifier diodes when the power supply is switched on. Right after switch-on, capacitors C1 and C2 haven't been charged yet and they constitute virtually a short-circuit for diodes D1 and D2.

Resistors R3 and R4 are "bleeders" which ensure that the electrolytic capacitors C1 and C2 are rapidly discharged after power has been switched off. Without R3 and R4, C1 and C2 maintain an electric charge for several hours, threatening the lifespan of the unwary experimenter (a.k.a., yours truly).

OUTPUT VOLTAGE CONTROL POTENTIOMETER

After I had built the circuits shown in Figs. 1 and 2, my regulator worked nicely, but the output voltage tended to drift. I thought the 85A2 tube was to blame, but this turned out to be not the case. Potentiometer R9 (Fig. 1) was causing the problem. Power dissipation in this device depends on the setting of the wiper and can be as high as 1.3W. Most modern potentiometers can handle only 0.25W!

I had used an above-average-quality 10k pot made by Bourns, but even this could not handle more than 0.5W. The pot became blazingly hot, resulting in a rapid increase of its resistance. It was clear that it did not have the proper power rating. During the tube era, I had used wirewound potentiometers with a power handling capacity of several watts, but I did not have such items in my junkbox.

After some thinking, I came up with the following solution (Fig. 3): I replaced R9 with a ten-turn 1k potentiometer from the East German company RFT (military stock, power handling 2W) with 4k7 and 820Ω wirewound 5W resistors in series. A switch (S1) allows the selection of two voltage ranges: 235-260 and 260-290V. Within each range, P1

allows continuous adjustment of the output voltage. After replacement of R9 by the circuit of Fig. 3, the HV regulator is quite stable.

My on-off switch has two poles and three positions. The first position is "off," the second position is "standby." In this position, mains power goes only to a 220:6.3V transformer which powers the heaters of the 12AX7 and the 6AS7G. After about 15 seconds, when the heaters

have reached their operating temperature, the switch is set to the "on" position. Then, all circuits are receiving power (including the circuit of Fig. 2) and the supply becomes fully operational. The opposite procedure is followed during switch-off.

HEATER SUPPLY

My heater supply (not for the 12AX7 and 6AS7G tubes in the HV regulator,

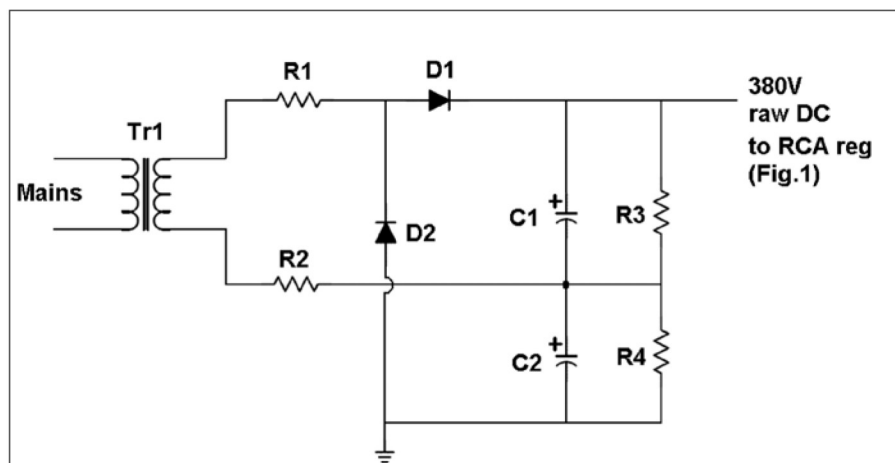


FIGURE 2: Voltage doubler generating raw DC for the RCA regulator (shown in Fig. 1).

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but for powering the heaters of external circuits) is shown in Fig. 4. Initially, I had made a 6.3V DC supply based on an LT1084CP regulator and a Linear Technology application note³. However,

this circuit failed after the initial tests. Because the LT1084 I used was salvaged from a damaged piece of laboratory equipment, the IC probably was already broken before I (re)used it.

Unfortunately, I did not have another LT1084 in my junkbox, and our local electronics stores do not sell this component. Thus, I tried a 7806 with external pass transistor and current limiting circuit. Although that circuit (taken from the Linfinity application note⁴) worked well, it turned out that a rather high input voltage was required if the circuit was to deliver several amps at 6.3V. The raw DC supplied by my 9V 50VA transformer with bridge rectifier and smoothing capacitor was too low (by 1 or 1.5V) for proper voltage regulation by the circuit based on the LM7806.

In a *very* old issue of *Elektuur* (known in the US as *Elektor Electronics*), I discovered a flexible circuit built from a small number of discrete components (Fig. 4)⁵. An input-output voltage difference of only 2V is required for proper operation of this regulator, and the circuit can supply currents up to 7A. Output voltage is identical to the zener voltage of D3 minus 0.5V. Zener voltages ranging from 4.7V to 48V are possible. The zener is fed from a current source (Q1, D1, D2, R1, R2). Q2 and Q3 are the error ampli-

fier, and Q4 is the pass device.

Because I had all parts in stock and could build the circuit in a single evening, I decided to give this vintage regulator a try. It performs nicely. The output voltage drops only 90mV (from 6.29 to 6.20V) when the output current is increased from 0 to 3A, indicating an internal resistance of approximately 30mΩ.

NEGATIVE BIAS SUPPLY

The regulator providing a variable negative bias voltage is even more conventional than the 6.3V heater supply. It is a standard application of the LM337T IC (see Fig. 5, taken from a SGS-Thomson application note⁶).

My junkbox transformer powering the RCA voltage regulator has both a 70-0-70V secondary winding and a separate winding of 28V AC. With a bridge rectifier (B250 C5000) and 4700μF 80V smoothing capacitor, the latter winding generates a raw DC of -35V for the circuit of Fig. 5.

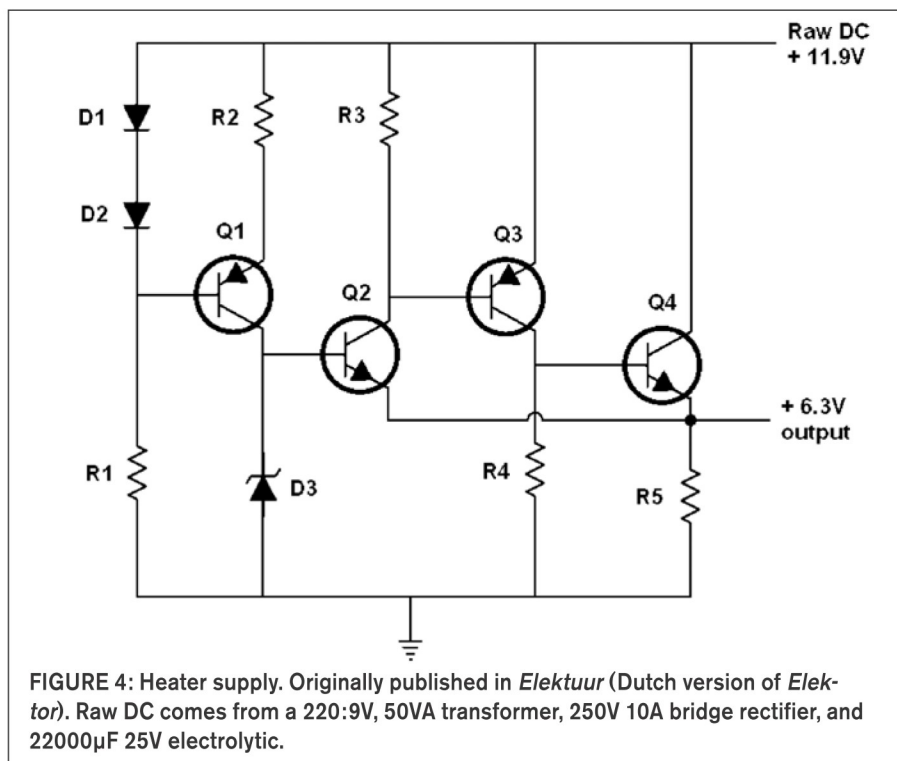
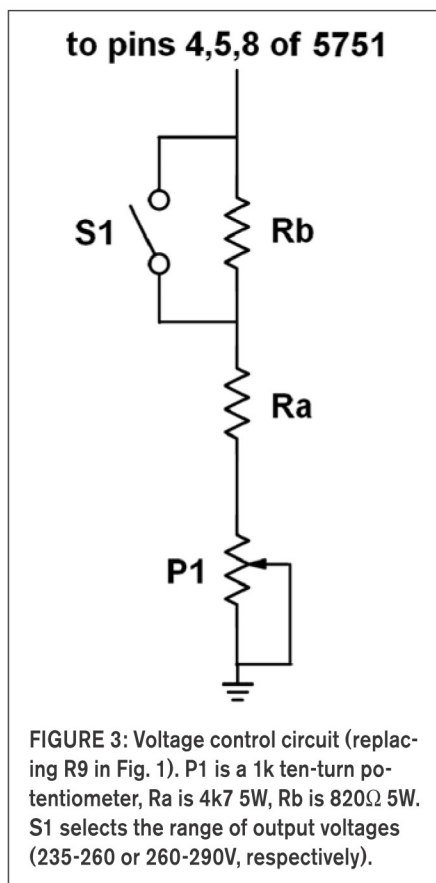
I added protection diodes D1 and D2 to ensure survival of IC1 when the regulator is feeding a capacitive load and mains power is switched on (or off). Capacitor C2 improves the suppression of mains-induced hum.

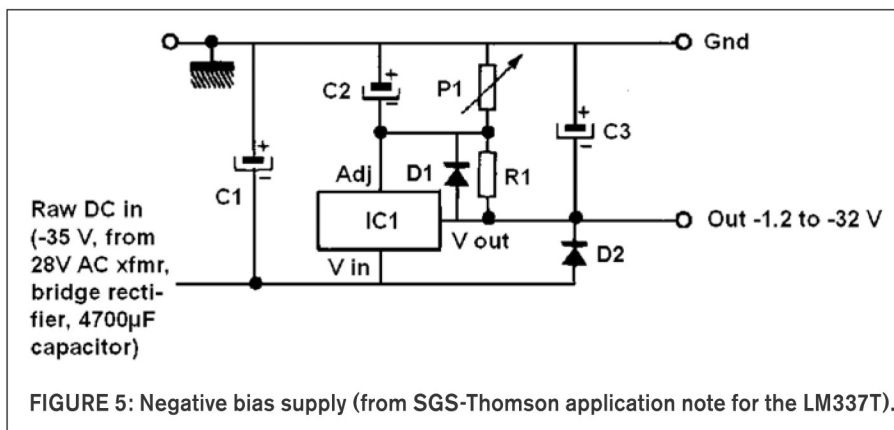
CABINET AND FINAL TESTING

Because of the size and mass of the components used in this power supply (three big transformers, several large smoothing caps, tall 6AS7G triode), I could not find an industrial cabinet large enough to hold all parts. Thus, I made a cabinet from plywood, MDF, and perforated aluminum (18mm thick MDF for the bottom, 6mm MDF for front and back, 16mm plywood for the cabinet sides, 1mm perforated aluminum for the cover). All parts are held together with screws and wood glue.

The circuit grounds of the three power supplies (high-voltage regulator, 6.3V heater regulator, negative bias voltage) are kept separate (not joined within the cabinet) to avoid ground loops when the supplies are used to feed an external tube circuit. Now you can connect the negative (or positive) ends of the heaters of the external circuit either directly to the B+ ground or to the B+ ground via a small capacitor, or keep them floating.

Because of the simplicity of the three





regulator circuits, I did not use etched printed circuit boards. I glued narrow strips of single-sided, copper-clad PC board material on pieces of unetched PC board. I soldered the components between these insulated “islands,” and used the unetched PC board as a ground plane.

I mounted meters for reading the B+ voltage and the negative bias voltage (300V and 35V full scale, respectively) at the front of the cabinet, to indicate proper operation of the power supply and disappearance of the output voltage after switch-off.

Tests showed that the high-voltage regulator can easily supply more than 100mA (I did not go further because I didn’t have a proper dummyload), the

negative bias supply more than 500mA, and the heater supply more than 3A before any appreciable drop of the output voltage occurs.

Thus, I managed to build a flexible power supply for vacuum tube projects, completely from junkbox parts, which will, I hope, serve me well for a long time.

aX

REFERENCES

1. The classical treatment of this subject is F.V. Hunt and R.W. Hickman, “On Electronic Voltage Stabilizers,” *Review of Scientific Instruments*, Vol. 10, 6-21, 1939. A recent article providing schematics with “modern” tubes is: Terry Bicknell, “Valve (Tube) Regulated Power Supplies,” May ‘08 *audioXpress*, p. 26. Another good introduction is “Vacuum-Tube Regulator Design” by Denzil Dan-

ner, *Glass Audio* 3/93, p. 16. Nice introductions can also be downloaded from the Internet, e.g., a five-part series on “Tube Voltage Regulators” by John Atwood, published in *Tube CAD Journal* (volumes 1 and 2, 1999-2000), and another five-part series by Steve Bench, available from his website, <http://greygum.net/sbench/sbench101/>. Steve’s website is a highly recommended goldmine of tube lore.

2. *RC-30 Receiving Tube Manual*, Including Picture Tubes and Industrial Receiving Tubes, RCA Corporation, Camden N.J., 1975, p.710.

3. *LT1083/LT1084/LT1085 7.5A/5A/3A Low Dropout Positive Voltage Regulators*, Linear Technology Corporation, Milpitas, Calif. 1994 (see figure TA04 in this document). NB: If you need datasheets or equivalents for any electronic component, try www.datasheets.org.uk or www.datasheetarchive.com or www.alldatasheet.com.

4. *SG7800A/SG7800 Positive Fixed Voltage Regulator*, Linfinity Microelectronics, Garden Grove, Calif. 1997 (see Fig. 7 in this document).

5. “Spanningsstabilisator met hoge konstantheid en lage Ri,” *Elektuur*, July/August issue 1972, p.763.

6. *LM137/LM237/LM337 Three-Terminal Adjustable Negative Voltage Regulators*, SGS-Thomson Microelectronics, 1993 (see bottom figure on page 8 in this document).

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TABLE 1: Parts list for RCA regulator (Fig. 1)

Part	Value
R1.....	Plate current balancing potentiometer, 160Ω
	10W (see text)
R2, R5.....	12k 2W
R3, R4.....	470k ½W
R6.....	68k 1W
R7.....	1M ½W
R8.....	15k 2W
R9.....	10k output voltage control potentiometer (see text)
C1.....	0.1µF 400V
V1 (left).....	6080 or 6AS7G
V2 (middle).....	5751 or 12AX7
V3 (right).....	5651A or 85A2

TABLE 2: Parts list for Voltage Doubler Circuit (Fig. 2)

Part	Value
Tr1.....	Power transformer, 140V secondary (see text)
R1, R2.....	Surge limiting resistors 15Ω 5W
R3, R4.....	Bleeder resistors 560k 2W
D1, D2.....	1N5408
C1, C2.....	1500µF 450V

TABLE 3: Parts list for the heater supply (Fig. 4)

Part	Value
D1,D2.....	1N4148
D3.....	Zener 6V8 400mW
Q1.....	BC557
Q2.....	2N1613
Q3.....	BD140 with cooling fin (I used a SD350)
Q4.....	2N3055 on big heatsink
R1, R5.....	22k ½W
R2.....	180Ω ½W
R3.....	47k ½W
R4.....	100k ½W

TABLE 4: Parts list for the heater supply (Fig. 5)

Part	Value
D1, D2.....	1N4002 (protection diodes)
C1.....	0.1µF 50V electrolytic*
C2.....	10µF 50V electrolytic
C3.....	1µF 50V electrolytic
IC1.....	LM337T
R1.....	120Ω 1% ½W
P1.....	Potentiometer 3k#

* Mounted close to IC1
In my prototype, a potentiometer of 10k with a 4k7 resistor in parallel

Maintaining Your Balance, Part 2

Follow this author's instructions and you're on your way to better LP-to-CD conversions.

LOCATING COMPONENTS ON THE BOARD

Using the program ExpressPCB, first determine the dimensions of the board based on components' sizes and shapes. Once that is set into the layout program, placement of the components can begin, the easiest way being to use the component manager to select a compo-

nent outline and place it into the layout. **Photo 1** shows an electrolytic capacitor with 0.2" lead spacing.

Note that if the component is polarized, the positive lead has a square pad, a convention that makes correctly orienting the component easier. You can also rotate the component before insertion into the layout. Once it is in the layout,

you can move the component on the grid and rotate it to different orientations. You will need to make sure that the placement allows for the maximum component dimensions so that it does not interfere with adjacent components. The grid markings make that simpler. Be especially careful of items such as heatsinks, which can extend well beyond

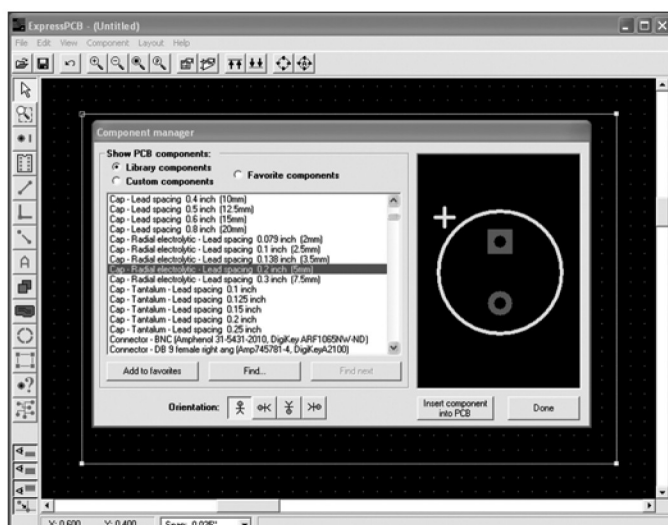


PHOTO 1: Device in Component Manager.

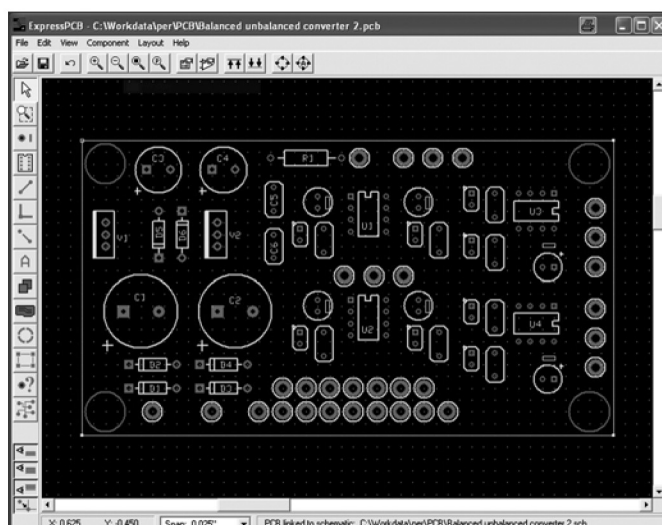


PHOTO 3: Parts layout.

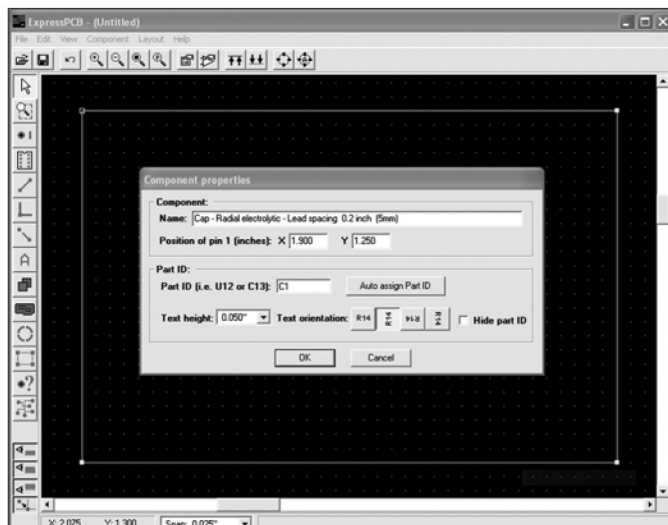


PHOTO 2: Setting component properties.

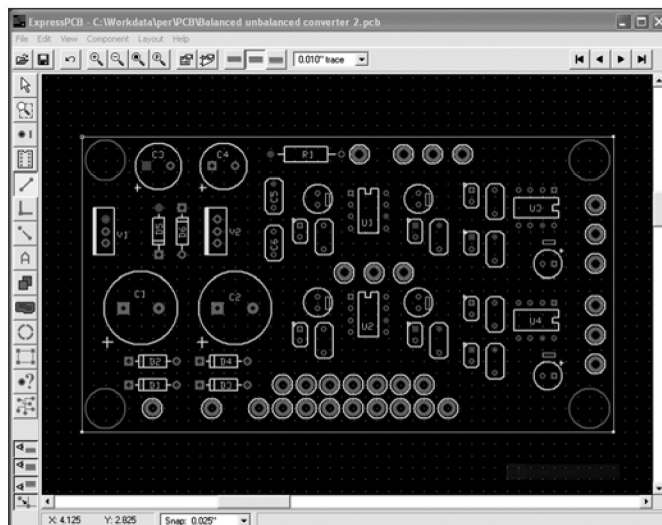


PHOTO 4: Highlighted pins.

the device outline.

If you have previously produced a schematic for this circuit using ExpressSCH, you can achieve other results at this time. By linking the board to the schematic, you can set device properties as shown in **Photo 2**.

Once linked, the board program knows all the connections. If you set the position of pin 1 in the horizontal and vertical positions, you will not need to drag it into position once in the layout. You can also choose the designation from the schematic, which will assist in the trace layout. The component will automatically assign the text used in the schematic to the component in the layout. Sizing and rotation of the text is also possible.

LAYING DOWN TRACES

After the components are in place, you need to lay down the traces to connect the components. This is where generating a schematic in ExpressSCH can be a major time-saver, because ExpressPCB will automatically indicate the pins that you need to connect. **Photo 3** shows the parts layout of the project board before the traces were defined.

To begin the trace layout, you can click on any pad and all the pads that are connected together on the schematic will be indicated by a color change to blue. In **Photo 4** the output of V1 was highlighted and the following pins have been marked in blue:

The output of V1
The anode of D5
The positive side of C3
One side of R1
One side of C5
Pin 6 of U1
One side of each B+ power-supply bypass capacitor
Pin 7 of U3
Pin 7 of U4
Pin 6 of U2.

You can now use the trace layout tool to connect the highlighted pins with appropriately sized traces. You can also

pick which side of the board (top or bottom) a particular section of trace should use. For example, you may choose to have a connection on the top of the board go to one pin and have the connection from that pin go to the next component on the bottom of the board. The plated-throughhole will connect both sides of the board. This helps reduce the need for jumpers.

Once you have made all the connections to a particular set of interconnected pins, you can highlight another pin from a different group and repeat the process until all the traces are laid. The completed trace map is shown in **Photo 5**.

Both programs include many more tools which are not discussed here. The best way to learn about them is to download the programs and experiment.

You are now ready for final checking before sending the file out for board production. To assist in checking, the program provides several options to print various board configurations as shown in **Photo 6**.

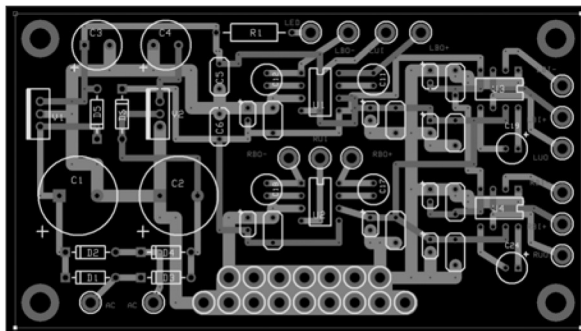


PHOTO 5: Completed trace map.

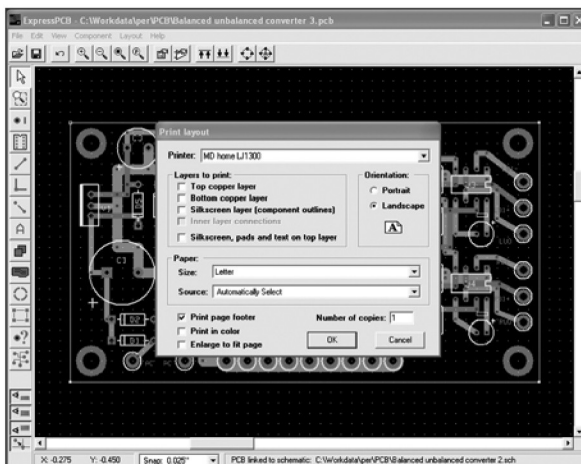
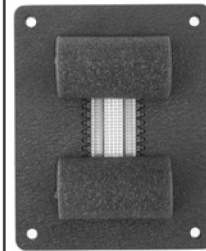


PHOTO 6: Board print choices.

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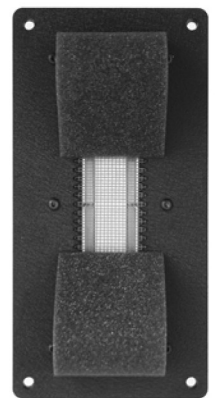
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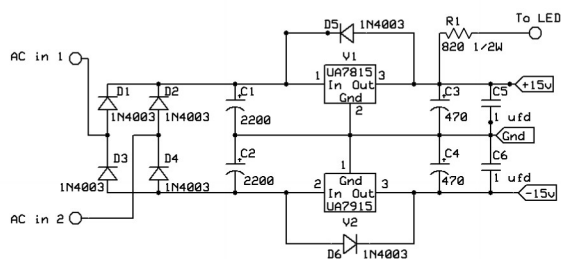
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 RBQ- ○ Right Bal. Out -
 LBI+ ○ Left Bal. In +
 LBI- ○ Left Bal. In -
 LUO ○ Left Unbal. Out
 RBI+ ○ Right Bal. In +
 RBI- ○ Right Bal. In -
 RUO ○ Right Unbal. Out

Gnd in 1 ○ Gnd
 Gnd in 2 ○ Gnd
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 Gnd in 4 ○ Gnd
 Gnd in 5 ○ Gnd
 Gnd in 6 ○ Gnd
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Unbalanced/Balanced Converter		
Thomas Perazella	Rev 1.1	
	9/11/2007	

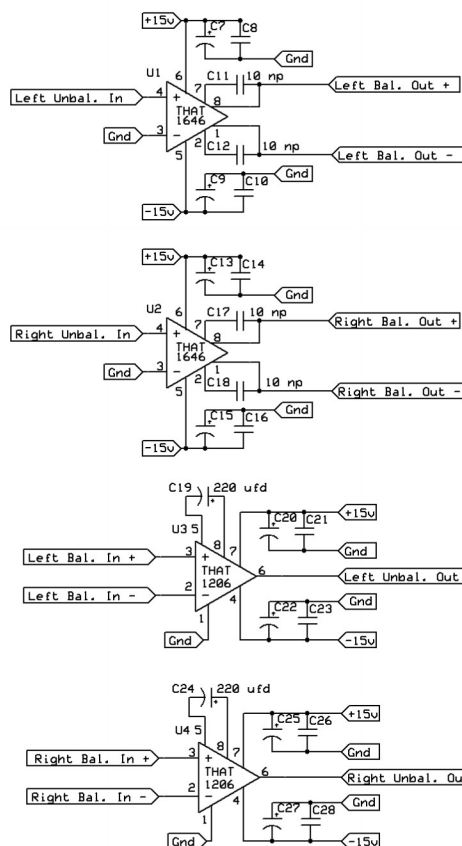


FIGURE 1: Unbalanced/balanced converter circuit.

You can print out several aids including:

- A schematic as previously shown in Fig. 1.
- A component layout diagram without pins as shown in Fig. 2.
- A component layout diagram with pins as shown in Fig. 3.
- A trace map of the top side of the board as shown in Fig. 4.
- A trace map of the bottom side of the board as shown in Fig. 5.

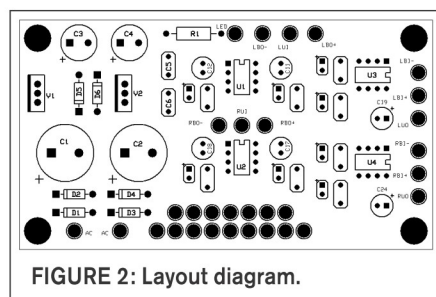


FIGURE 2: Layout diagram.

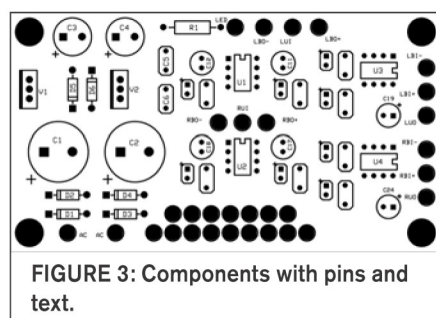


FIGURE 3: Components with pins and text.

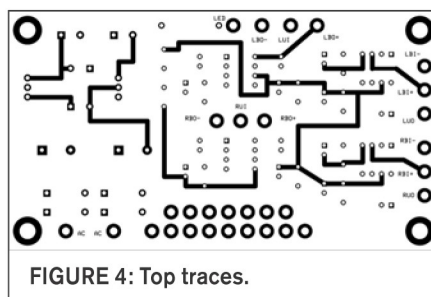


FIGURE 4: Top traces.

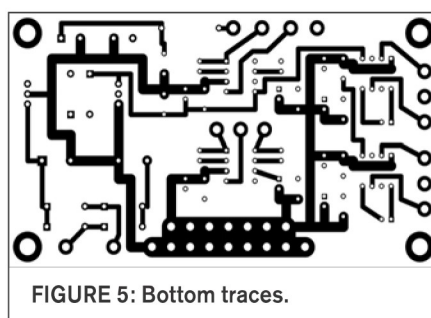


FIGURE 5: Bottom traces.

Using these, you can trace all connections to ensure the file is correct before sending it out for processing. One thing is certain—complacency leads to errors. I checked the schematics and maps several times before sending the file out, and everything checked OK. However, I was complacent and checked the pin connections for the positive regulator from memory because I had used that device so many times. I remembered the pinouts incorrectly. As a result, I reversed the output and ground connections, which went unnoticed until the boards were built and I began testing. Shame on me!

I sent out the file via e-mail, and the completed boards arrived a few days later. The quality looked very high. Views of the top and bottom of the boards are shown in Photos 7 and 8.

As is my regular process for building circuits, I build the power supply sections first and test them before adding other components. This helps prevent catastrophic failure in case the power supply has problems.

After the power supply components were added to the board, I connected the transformer and slowly increased the input voltage with a Variac while measuring output voltage. The mistake I made with the regulator pin connections was immediately evident in the output voltage measurements. After checking the board and finding that it was correct to my specifications, I looked up the pin diagram for the regulator and realized my mistake. I made a simple manual fix to the board to re-route the traces. Pictures of the fixes for both sides of the board are shown in **Photos 9 and 10**.

I removed two of the traces with a sharp knife blade and used replacement connections made out of solid hook-up wire. The re-tested supply worked properly. I re-checked my other connections and corrected the schematic and board layouts—all the diagrams in this article are correct. I added the rest of the components to the board and tested the completed board (**Photo 11**).

Preparation of the housing started with selection of the appropriately sized die-cast aluminum box. I prefer metal boxes to plastic for their ease of machining and shielding. I mounted the transformer and circuit board on the base of the box with all the signal connectors attached to the main body of the box, and with all the necessary holes drilled and de-burred. I then cleaned, primed, and painted the box parts.

In the past, when doing a one-off project, I would use press-type letters with a protective coating of clear urethane. Unfortunately, those type sheets are no longer to be found. For this project, I formed the letters and symbols using a Brother P-Touch label printer with a tape cartridge having white type on a clear background. It worked, but was not as professional-looking as the press-type. The finished box is shown in **Photo 12**.

To mount the board and transformer, I made two sets of brackets from flat steel barstock. The brackets allowed vertical mounting of the two pieces for better internal space utilization. I wired the transformer to the board as shown in **Photo 13**. Note the central chassis grounding lug mounted under the nut on one of the circuit board brackets.

The next step was to mount the signal connectors to the box. One of the

biggest problems with noise in balanced circuits is the so-called “pin-1 problem” as detailed by Neil Muncy in his June 1995 paper. The preferred method of

grounding XLR connectors is to connect pin-1 and any connector-shell grounds to the chassis as close to the connector as possible. **Photo 14** shows the pin-1 con-

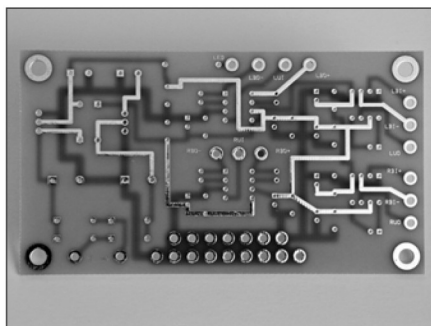


PHOTO 7: Top of board.

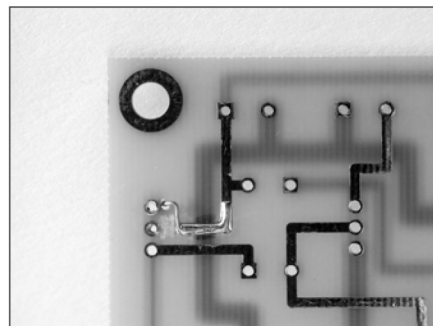


PHOTO 9: Fix for error on top side of board.

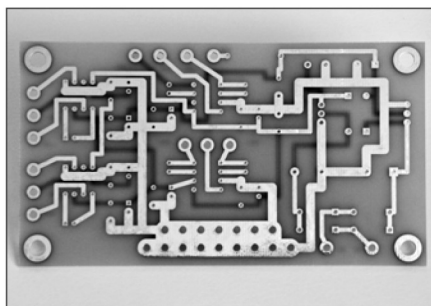


PHOTO 8: Bottom of board.

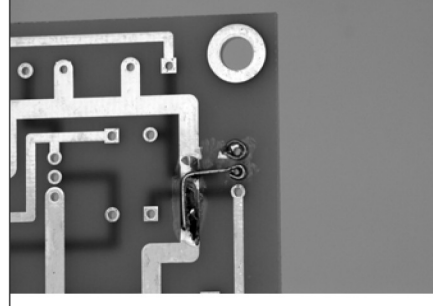


PHOTO 10: Fix for error on bottom side of board.

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nections from the XLR connectors going to a chassis ground lug right next to the connectors.

Photo 15 shows the RCA connectors using insulators to float them from the chassis at the input point. Unlike the balanced XLR connectors, the shells of the RCA jacks are signal-carrying and

are connected through individual wires to the central ground point on the circuit board rather than at the chassis. **Photo 16** shows the ground connections on the board as well as ground leads from the box and the power cord going to the chassis ground on the lid.

The wire connection from the ground

lug on the box and the ground lug on the lid provides a ground connection when the lid is off for testing. Once closed, the mating sections of the metal box provide the definitive overall chassis ground. After making the power input and indicator connections, I was ready to test and close up the box. **Photo 17** shows the

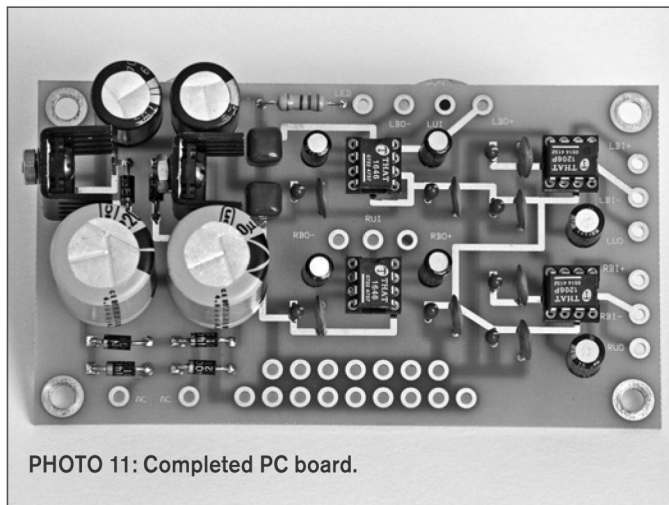


PHOTO 11: Completed PC board.



PHOTO 14: Pin 1 grounding.



PHOTO 12:
Completed box.



PHOTO 15: RCA jacks insulated from chassis.

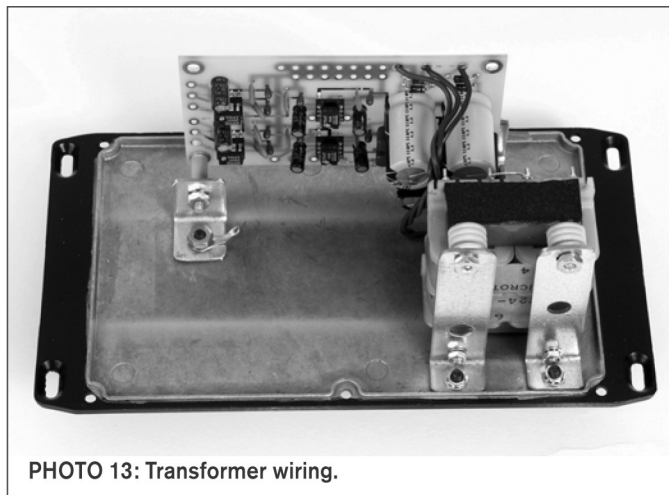


PHOTO 13: Transformer wiring.

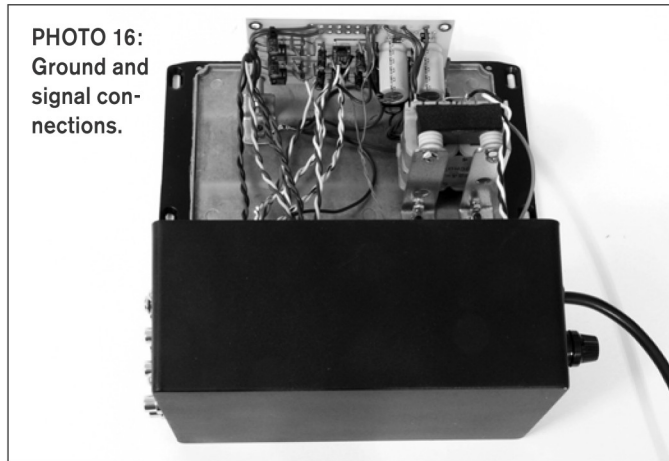


PHOTO 16:
Ground and
signal
connections.

completed project just before closure.

Photos 18 and 19 show the completed box and a close-up of the connector arrangement. Note that for the unbalanced connectors there is one pair for input and three pairs for output. The triple output allows for connections to the computer sound card, a headphone amplifier, and any other test equipment I might choose to put into the circuit.

TESTING

A good measure of how well an electronic device performs is to determine how closely it compares to a straight wire when a signal is passed through. To

test this project, I made two cables that would allow me to loop the output of the balanced circuit back into the balanced input. That would allow me to input a signal to the unbalanced circuit and measure the output signal of the unbalanced circuits after passing through two conversion steps. I used test sine waves of 1V RMS at 5Hz, 1kHz, and 500kHz with the input and output signals displayed on my oscilloscope.

Figures 6, 7, and 8 show the results at 5Hz, 1kHz, and 500kHz, respectively, with the input signal as the top trace and the output signal as the bottom trace. They are virtually identical. To make

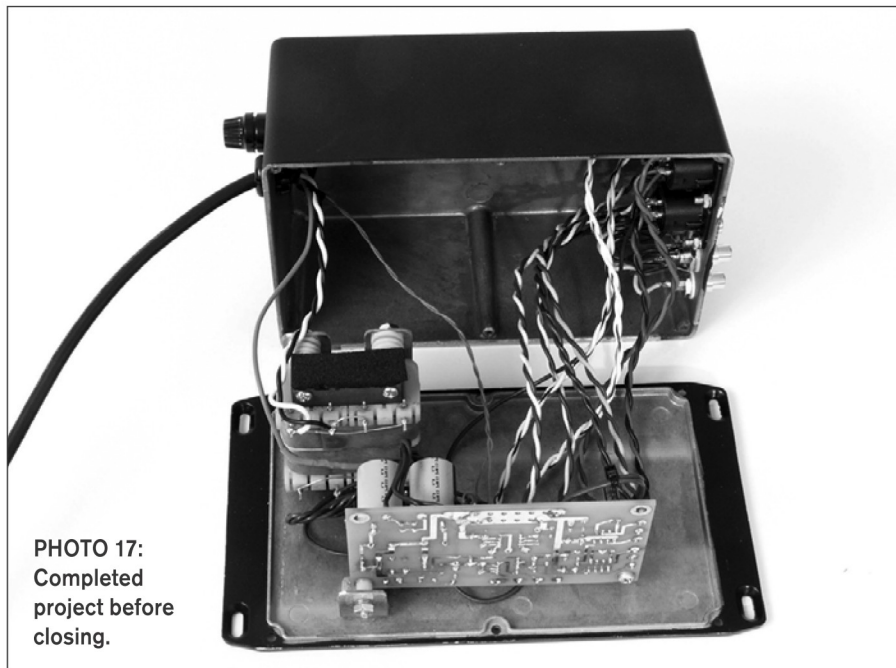


PHOTO 17:
Completed
project before
closing.



PHOTO 18: Completed project.

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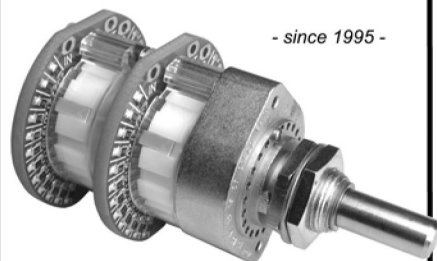
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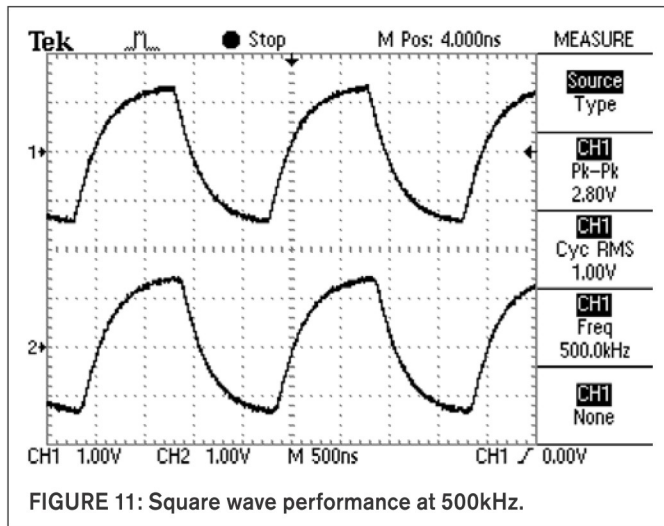
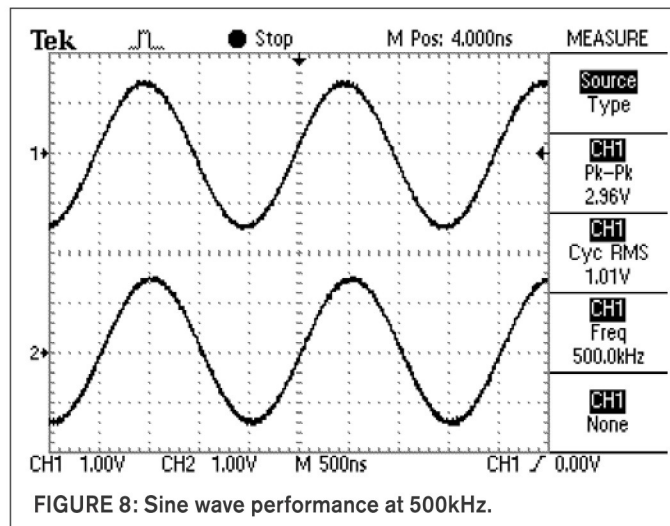
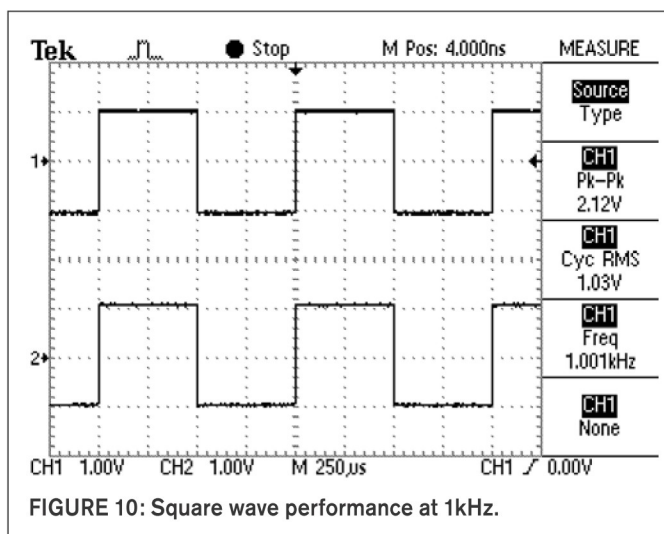
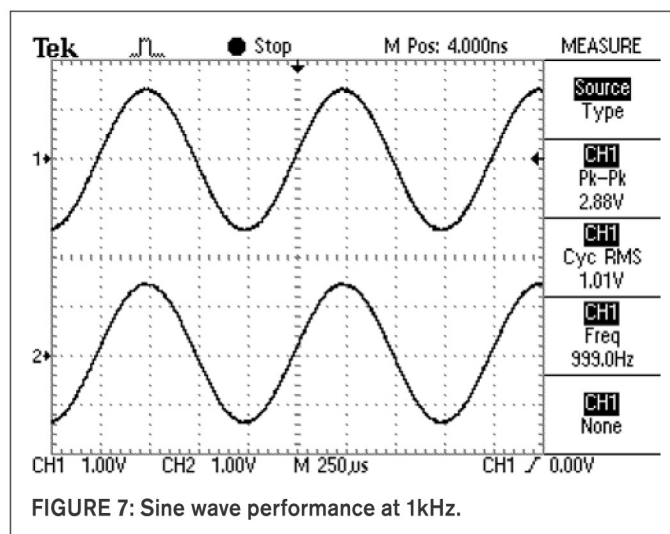
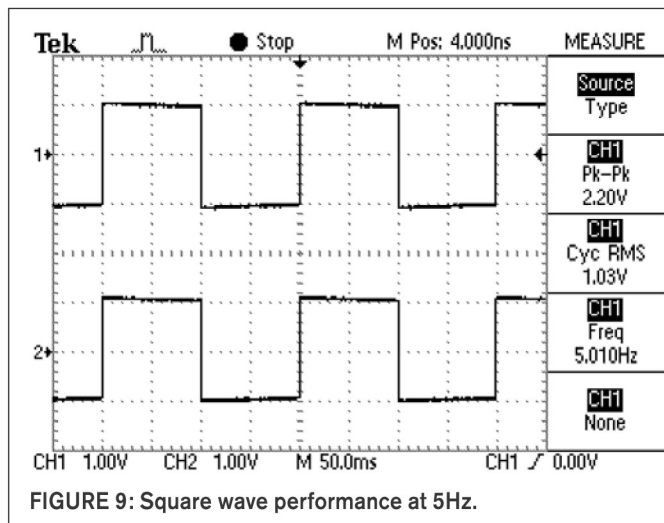
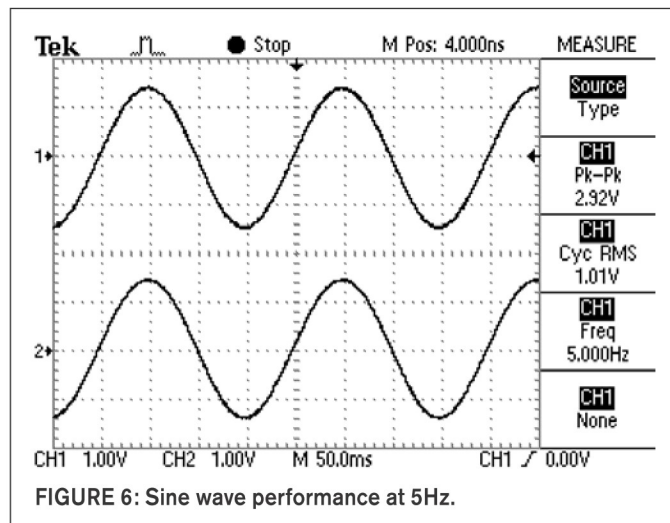
the test harder, I repeated the test using square waves at the same frequencies. **Figures 9, 10, and 11** show the results with no visible problems.

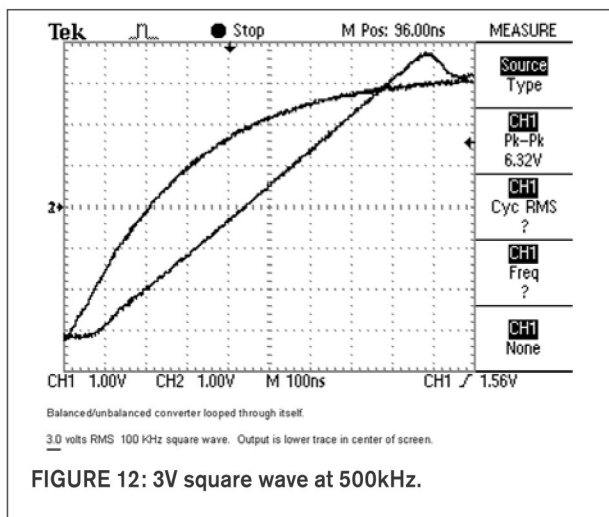
For an even more difficult test, I increased the signal level to 3V RMS and

expanded the trace of the 500kHz square wave to determine whether there were any differences, even though they would be far outside the audible band. **Figure 12** shows the leading edge of the square wave with a time-base setting of 100ns/

division.

You can see a rise-time lag of about 100ns, with a very slight overshoot lasting about 100ns. This is certainly not audible and indicates the excellent performance of both devices.





I wanted a live test in a recording venue, and contacted my friend Michael Morgan,⁸ who makes fine professional recordings for NPR and others. Although most of his equipment uses balanced interconnections, he set up a test using unbalanced and balanced connections. The project worked with no problems.

Finally, I connected it in my LP-to-CD conversion setup where it worked as hoped with no audible added noise or distortion. **Photo 20** shows that setup, along with my Dual turntable with Shure Ultra 500 cartridge, my home-built RIAA phono preamp based on OP27 op amps, the balanced-to-unbalanced converter, a Behringer DEQ2496 equal-

izer, a Headroom headphone amp, a pair of Etymotic ER4S earphones, and the output to my PC.

I have only had time to burn a few CDs, using Audacity™ as the recording software, but the results have been great.

CONCLUSIONS

The addition of the THAT Corporation balanced-to-unbalanced circuits has eliminated many of the problems inherent in those types of conversions without the expense and frequency response limitations of many available transformers. High-quality transformers still hold the top spot for ultimate CMRR at higher frequencies, and are mandatory where

significant and possibly dangerous voltage offsets exist. However, for most situations, these THAT ICs will provide excellent performance that is certainly far beyond using adapter cables. Kudos to Bill Whitlock and the folks at THAT Corporation. *ax*

REFERENCE

8. Michael Morgan, Isle of Skye Audio Productions, www.RecordClassical.com.



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THIEL SCS4 Loudspeakers

By James T. Frane



For the past few years, I've been the happy owner of a pair of THIEL CS2.4 floor-standing speakers. So, it was a pleasant surprise when THIEL offered to send me a pair of SCS4 speakers to review.

The speakers arrived packed in sturdy cardboard boxes with lots of polyurethane foam padding. The SCS4 is a two-way loudspeaker designed to be used full-range for stereo. After lengthy listening and measurement sessions, I'm convinced they are also well-suited for use as surround-

sound, center-channel, side, or rear speakers. The review speakers had lacquer-finished black ash cabinets, but they are also available in natural or dark cherry.

THIEL Loudspeakers
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mail@thielaudio.com

FEATURES

These speakers have a die-cast aluminum baffle to provide a rigid support for the drivers, as well as double-laminated 1" thick MDF (medium density fiberboard) walls and internal bracing to eliminate unwanted resonances. No feet are provided with these speakers. In the center of the back panel are two gold-plated binding posts that accept bare wire, spade lugs, and banana plugs. The post's

mounted vertically or horizontally, on a shelf or a stand. The woofer has a short voice coil and long gap to ensure the coil remains within the fixed magnetic structure of the speaker for constant control and to prevent distortion. The woofer cone shape reduces diffraction of the tweeter output. The drivers are mounted in the vertical and horizontal center of the front baffle.

A crossover between 1st and 2nd-order divides the output between tweeter and woofer and results in a cohesive full-range sound. Diameter ports measuring 1 1/4" are located above and below the drivers. The center of port to center-of-driver distance is 5 1/2" on the speaker vertical centerline, and each port has a 5" long tube extending into the cabinet.

The aluminum chassis drivers are protected by an easily removable metal mesh grille held in place magnetically. The mesh is so fine as to be visually transparent to a large degree. The cabinet front edges are rounded to prevent diffraction (a redirection of sound waves that can be caused by sharp enclosure edges in the sound wave path). The enclosures are 8.4" wide x 11.7" deep x 17.6" tall (21.3 x 29.6 x 44.7cm). Each speaker weighs 25 lb (11.5 kg).

The rated frequency response of the SCS4s is 48Hz to 20kHz $\pm$ 2dB with a sensitivity of 87dB and nominal impedance of 4 Ω (3 Ω minimum). The recommended amplifier power to be used with the SCS4s is 30-200W. The supplied owner information pamphlet states: "At least 50 hours at moderately loud levels are required before the speaker is performing near optimum. You should notice even more improvements after 100 hours of playing."

TESTING

I placed the speakers 36 1/2" above the floor on a solid cabinet, 12" from the back wall, with the centers of the speakers 6' apart. I used the speakers daily for a few weeks to ensure they were properly broken in. There were neither enhanced sibilants nor any hollow or booming sounds at any listening level. Removing the grilles

did not make any identifiable difference in the sound. Whether with TV, FM, DVD, or CD, the sound was always clear and natural.

The SCS4 size accommodates many placement options, but they should be at about ear level to enjoy the best sound because high frequencies are directional. You can enhance the total frequency response by adding and carefully tuning in a good subwoofer to reach the

“they surpass many speakers their size as well as some larger speakers”

knurled knobs also have wrench flats for tightening, but I achieved a snug connection using my fingers.

The driver arrangement is different from that used on most speakers. A 1" metal dome tweeter (similar to THIEL's flagship model CS3.7 speaker) is coaxially mounted in the center of a 6.5" diameter metal diaphragm woofer. The coaxial mounting is designed to provide correct time response whether the speakers are

lowest octave. The need for a subwoofer will depend on the type of music you like and, to a degree, the room reinforcement of the sound. The SCS4's reproduction of voices was very realistic—something not all loudspeakers achieve as well. The integration of tweeter and woofer was seamless, as was the blending and bass enhancement from the two ports.

SELECTIONS

I used a wide variety of music types and several movies plus television shows to audition the speakers. Actors' and singers' voices as well as all other sounds were reproduced clearly and naturally to the extent of the recording quality.

The following are a few of the CDs I used: Julie London's "I've Got You Under My Skin" and "Get Out of Town" from the CD *All Through the Night* (Jasmine CD 308) had her voice centered and realistic with good rendition of the spaciousness of these two cuts. Bass was adequately reproduced. "At Last" from *The Best of Etta James* (MCAD-11953) had a smooth melding of vocal and backup orchestra with a solid bass line in the background. You could easily imagine her on the stage.

The superb "Georgia on My Mind" by Ray Charles (Dunhill DZS036) had the depth and feeling over the wide-spread stage of this recording. His voice was very natural, and the background singers were realistically set in the background. "Unchain My Heart" from the same CD has a little deeper bass than is fully reproduced with the SCS4s, but you would likely notice this by comparison with much

larger speakers, such as my Thiel CS2.4 speakers.

Allegro from Johannes Brahms' "Quartet for Piano and Strings, Op. 25," performed by Murray Perahia and Members of the Amadeus Quartet (CBS MK 42361) had the right separation and location of instruments. There was a depth, three-dimensionality, and musicality to put the listener at the performance. Gerry Mulligan's "The Lady is a Tramp" from his live performance *In Paris* album, Volume 2 (BMG/RCA Victor/Disques Vogue 09026-68212-2) was reproduced with all of the spaciousness and feeling of a live recording. Instrument locations, timbre, tone, and depth took me to the performance—a great recording done justice by these speakers.

These recordings were a small sampling of the tunes I enjoyed with the SCS4 speakers. The speakers were never fatiguing, harsh, off key, or lacking depth, and they surpass many speakers their size as well as some larger speakers. Fans of organ music will find the lower registers diminished or lacking, but most music was complete and enjoyable through these speakers. They compared quite well with my favorite small speakers for over the last decade—the Mach One Acoustics M-Two speakers, which, unfortunately, are no longer produced. I would not be displeased with the SCS4 speakers as part of my home theater system.

Home Theater System Used

Carver CT26V tuner-preamp,
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iGrado

With these "street style" headphones available in black or white, Grado enters the MP3 portable market. They are wraparound-type headphones, with the headband resting on the back of

your neck and the ear pads tightly pressed against the ears. They weren't as uncomfortable as I expected them to be, and even though they are not adjustable, they fit me just fine. But I have a more or less small frame.

The iGrados sounded surprisingly good for what they are. I did the majority of my listening to compressed files through the iPod, but also hooked them up to a Headroom Micro headphone amp and played CDs and records to see how they compared to my reference. The bass went unexpectedly deep, and the mids were very clean and relatively transparent. The highest treble was a bit rolled off, but considering their price, the highs were remarkably extended.

I am using words such as "unexpectedly" and "remarkably" because I had to keep reminding myself that these headphones cost

noisy environment, and especially with compressed files (of course), it didn't seem to much matter that they weren't faultless.

Out-of-doors, I'm used to the isolation of the now discontinued \$180 in-ear Shure Ec3s using the foamy earplug-type inserts. They block out most noise, but the iGrados, on the other hand, have little isolation, so occasionally the external noise interfered with the music. I tended to turn up the volume a little too much to compensate for this. On the plus side, the iGrados sounded much more powerful than the in-ears, especially in the bass, where the minuscule drivers of the Shures don't come close to the sense of real low-end that the bigger drivers of the iGrados deliver. I'm aware that the Ec3s—even though they are not a state-of-the-art product by any means—sound much more transparent and as a result more neutral, but the iGrados are larger, and they sound it. But as far as blocking out external noise and a sense of privacy, the in-ear phones are better, and, to me, more comfortable.

One more small complaint I have regarding the iGrados is its cord. Although it is short (40"), which makes it perfect for a portable player attached to one's belt or in a pocket, because it is thin as a thread it often became tangled up

"The bass went unexpectedly deep, and the mids were very clean and relatively transparent."

only \$49, and for \$49 it *is* remarkable that they practically approached what would be considered to be high-end sound, especially when paired with an external headphone amplifier playing uncompressed files (of course). Their overall sound was rather natural, and although they didn't sound like the real thing inside my skull, instruments had a relatively uncolored sound that made it easy to forget that these were of the budget variety. Their sound came dangerously close to Grado's SR-60s, which shouldn't be that surprising given that they use the same drivers. They definitely had the Grado family sound.

Judged against my Sennheiser HD-600s (which cost more than seven times the street price, so maybe it wasn't a fair fight), the bass was a little one-note-ish, the mids were pushed pretty far forward in the mix, and although the treble was more or less adequate, it was, for lack of any other term, sloppy. The overall sound of the phones didn't have nearly the resolving power. But because the iGrados are not sealed nor have any noise reduction capabilities, when used in a

when the headphones weren't in use. But I don't know why I'm complaining—the thicker cord of the Shure in-ears gets tangled up, too, although not as frequently. The iGrados also have a rather small plug on the end of its cord, and even though I liked its width because the small size made it fit perfectly into the iPod's headphone socket even when in its skin, I still have concerns about this diminutive connector's durability.

Even if it seems as though I have more than a few complaints, I'm really just picking nits when you consider their price to performance ratio, which makes these headphones a relative bargain. I'm not going to profess that I'm an expert on these types of hip headphones; but what Grado is obviously targeting is the segment of the youth market who might be even remotely concerned with sound quality. They will quickly realize that the iGrados are unquestionably better than the crappy earbuds that Apple supplies with their iPod, or any other giveaway earphones that come with a typical MP3 player. In Grado's words, the iGrado "outper-



forms anything currently available in the low-end portable market." I agree.

SR80

I'm well aware that the SR80s have been on the market for some time now, but I think it's important to note that perhaps they've lasted for so long because they're just a very good product. [John Grado notes that the SR-80 has been discontinued and replaced with the SR80i (for improved). See manufacturer's response.—Eds]. They have not been revised to Mk. IV-a status, so perhaps Grado must have done something right the first time. For a lousy \$26 more than their lowest priced model (apart from the iGrado)—the SR60s—they don't come anywhere near to breaking the law of diminishing returns. They are way more comfortable, and that alone is worth the extra pocket change.

Sound-quality-wise I would dare say that the venerable SR80s come within earshot (sorry) of the basic qualities that make the Sennheiser HD-600s so good. When I hooked them up to the Headroom headphone amplifier, the bass was deep and tuneful (Grado claims the bass goes down to 20Hz, but gives no accuracy) and the mids were quite transparent. The highs were excellent, although I think that's the area that highlights the difference between the Grados and the Sennheisers. The more expensive cans' treble was more detailed, with each sound differentiating itself more than the others and more natural sounding.

The more expensive headphones' dynamic capabilities also set them apart from just about any others that I've heard. But don't get me wrong, the SR80s' rolled-off highs and other slight limitations still left them in good company, and when you consider that they only cost \$95, it is astonishing that their overall sound quality is so good.

TESTING

I was amazed how good the doctored live version of *The Lamb Lies Down on Broadway* from the Genesis' box set *Archives, Vol. 1: 1967–1975* sounded when using the Headroom amp with an Arcam CD player. On this recording the bass guitar and synth bass pedals were extremely powerful, as they should be. Although the mids were a little pushed forward, there was still a good level of transparency and sufficient detail. Cymbals and the other high-frequency sounds were quite good, and it was easy to detect that there was some signal processing applied to the otherwise excellent recording. Even though they couldn't compete with the best, there were still plenty of passages that showed the phones had very good dynamic capabilities and punch. They rocked.

I also played John Adam's *Chamber Concerto* performed by the Ensemble Modern on RCA released in 1997. On this CD I again noticed some sins of omission such as the highest treble, but all in all the sound was excellent. The strings sounded like real instruments recorded in a real space, and

the winds and percussion were locked in place within the soundstage between my ears. Even though the SR-80s weren't perfect, it was easy to become lost in this great music.

The Grados are much more efficient than my Sennheisers. When listening to an iPod the more expensive headphones are nearly useless without an external headphone amplifier. Whether that is because of their challenging impedance characteristics or just simply their lower sensitivity, I'm not sure.

Yet when listening to the Grados through an iPod without the external amplifier they were so much easier to drive that their bass ended up being stronger, and the mids and treble seemed to pass from the source to my brain with greater ease. I don't know why I'm going on about using the SR80s with an iPod so much; the cord is much too long to use outside, and they have little or no isolation. But, I did spend some time listening to them this way indoors while desk-bound and they performed admirably.

Audiophiles tend to spend much more than \$100 on a pair of speakers, so it makes sense that you might spend more than that on a set of headphones. So, sure, you could spend more money on a pair of headphones, and I doubt that spending more on a pair from the Grado catalog won't bring you closer to the music. But considering that they only cost about as much as ABBA's *The Complete Studio Recordings* box set, the SR80s are a steal. You could use these headphones, like the iGrados, as a stepping stone to Grado's more ambitious efforts, or use them as part of a long-term daily regimen of sustained musical enjoyment. Highly recommended.

Manufacturer's response:

What does the *i* stand for in the new SR80*i* from Grado? Improved, that's what! Built on the same features as the SR60*i*, the SR80*i* utilizes a four-conductor connecting cable, and the diaphragms are put through a special "de-stressing" process in order to enhance inner detail, the result of which gives a more open stage. The new SR80*i* has an upgraded driver design, and they have enlarged and improved the mass distribution in the plastic housing. The way the SR80*i*'s new driver and plastic housing move air and react to sound vibrations are now less affected by transient distortions. The SR80 provides an improved bass resonance, which enhances the overall detail. With the SR80*i* you will notice improved control of the upper and lower range of the frequency spectrum with both better supporting Grado's world renowned midrange. The SR80*i* will produce a sound that is pure Grado, warm harmonic color, rich full bodied vocals, excellent dynamics and an ultra smooth top end. Listen and enjoy!

rr

John Grado
President, Grado Labs



Sony XDR-F1HD

By David A. Rich

Sony XDR-F1HD Radio Tuner
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1-877-865-7669 or Sony store
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2 1/2 lbs
List price \$100



In his recent online review (www.ham-radio.com/K6sti/xdr-f1hd.htm), Brian Beezley concluded the XDR-F1HD is the best tuner for analog reception. Brian is the only reviewer still making measurements of FM tuners. The catalyst for improved performance is the DSP signal processing that replaced the analog IF and MPX sections of the analog tuner. Complete measurements are on Brian's website (www.ham-radio.com/k6sti/x_dr-f1hd.htm) to allow you to compare this tuner to the best analog units made from the '60s to '80s.

CHIP SETS

The measured analog performance of the Sony is better than the Sangean HD radio tuners I previously reviewed in *Sensible Sound* (Nov./Dec. '07, p. 12). The Sangean uses a TI DSP chip set; the Sony uses a chip set from NXP (the old Philips Semiconductor). Of inter-

performance, but also software code options that the manufacturer selects change performance. Significantly more technical information is on Brian's website than appears here.

FILTER

According to Brian, the Sony has: "*Adaptive noise reduction* that forms a filter that fits the L-R audio spectrum and follows its changes. The filter suppresses noise between and beyond spectral peaks without restricting audio bandwidth." Brian also identified that the Sony has *threshold extension*, which "suppresses the impulse noise ordinary detectors generate for weak FM signals." Analog PLL detectors have some threshold extension, but with DSP more sophisticated detectors may offer increased extension.

Continuing from Brian's website: "To take full advantage of the processing, the XDR-F1HD does not switch or abruptly blend to mono at low RF signal levels. The Sony can deliver a clean stereo signal with wide channel separation at far weaker signal levels than can any other tuner. The only thing remotely comparable is a Carver tuner with Asymmetrical Charge Coupled Detector."

Take care when comparing an older analog tuner to these DSP units with low input signal levels. Brian reports that with a mono input the "XDR-F1HD begins to invoke a sliding, first-order, lowpass noise filter on L+R signal as the RF level drops below 26dBf. . . The filter is similar to the one in the HDT-1X, which begins at 29dBf and whose corner levels off at 2.9kHz at -3dB. The Sony corner stops falling at 3.2kHz." He found the unit was down 13dB at 15kHz with a 16dBf input.

"the XDR-F1HD is the best tuner for analog reception."

est is the HD functionality achieved in the NXP design by adding an extra chip to what was called *Car DSP* chip. I have long maintained that modern car receivers perform much better than what you find in a \$4000 audio video receiver.

I am doing little here but summarizing Brian's work. Because the functionality of the NXP and TI chips are in the DSP code, a service manual will tell you nothing. Making things even harder is that the datasheets for the HD Radio DSP chips are not publicly available. Brian thus had to do a significant amount of reverse engineering. Making evaluation more complex is the fact that not only do the chip set differences change

When the signal level is high enough to turn off the sliding filter, the stereo frequency response is flatter for the Sony in comparison to the Sangean. Brian's review of the Sangean is found at <http://ham-radio.com/k6sti/hdt-1x.htm>. Sony's *adaptive noise reduction* discussed previously also helps to suppress noise from co-channel interference and multipath distortion at all signal levels compared to the Sangean.

RECEPTION

The Sony and Sangean units received the same number of stations in HD mode at my home despite the differences in chip sets. Note that both tuners use simple version 1980s style RF sections that require alignment, and if the tuner is not correctly aligned you will not get as many HD stations. Making objective measurements of the tuners' HD performance is impossible because no cost-effective HD radio modulator exists.

The Sangean HDT-1X has an S/PDIF output that eliminates DAC – ADC interface to an AV receiver. However, the digital encoding used in HD radio (96kbits/sec with no secondary channel) may obliterate the sonic difference of a digital connection.

The Sangean has carrier-to-noise ratio readout that allows you to best aim your antenna for optimal HD reception in marginal reception areas. Because HD is digital, it is all or nothing. In a marginal area, the tuner may flop back and forth between HD and analog, which is worse than analog alone even with a bit of noise. That is why the Sangean HDT-1X offers forced analog, which is not available on the Sony. The Sangean also has forced mono.

Brian has modified the Sony for forced analog by cutting a trace and adding a switch (see his website) if you want the Sony to be able to have both HD and forced analog reception. Only the trace cut is required if you want to turn the tuner into an analog only unit. The retrofit is only for the experienced amateur or a reliable repair shop even for the one cut, because it is hard to get inside the unit and the traces are small. Obviously, your warranty is toast after the mod.

A cautionary note: The Sony resembles a table radio and becomes rather hot with all the electronics shoehorned into its compact shell. The Sony has ten flimsy buttons on top of the unit, which provide almost all functions. The remote adds only a couple more. The muting is marginal as you scan stations with a small burst of noise as it passes each station.

FEATURES

The Sangean is a full-sized component and feels more solid. All functionality resident on the front panel is a real luxury, though it requires navigating 21 touch controls. All the extra buttons on the Sangean allow for direct entry of a station's frequency, making it much easier to tune a station than using the scan buttons. Buttons also vary in size and shape on the Sangean. You can see where the \$120 difference in price goes.

Unfortunately, the Sangeans have a number of bugs that are identified on Brian's website. Sangean appears unwilling to clean them all up although several updates were issued (which required the return of the unit). A summary of the problems with the HDT-1 series is found on Brian's website. I never found that any of these bugs bothered me, but I did not set any presets or use the AM HD tuner. Brian notes the Sony forgets many user settings if the power goes out for more than a few minutes. Modifications on his website allow the tuner to retain data longer.

If you want to achieve a weak analog signal from a station with HD, you may need the Sangean despite its poorer analog performance to prevent the HD flip-flop problems, provided you do not want to perform dangerous surgery on the Sony. The price of the Sony is about \$120 less at street prices, however, and that difference might go a long way to putting in an outdoor antenna if you and your neighbors find the structure acceptable. A good antenna could turn flip-flopers into solid HD stations, but there are no guarantees. I, however, have found better indoor FM antennas (Audioprism 8500 and Terk FM Pro) do little to prevent the flip-flop, although they can improve your ability to bring in a weak analog signal over a floppy dipole.

The FCC may evaluate increasing the power of the HD signal, but this would increase the noise of adjacent analog stations and may block weaker ones. I would not expect any rule change soon.

Deciding between the Sangean HDT-1X and the Sony is arbitrated by your signal conditions and tolerance for a clock-radio type of thing proximate to your \$2000 preamp. Despite its looks, the Sony appears to be free of the operational bugs the Sangean units have. If you have HD flip-flopers, want a stock unit, and have no desire to install a better antenna, opt for the Sangean HDT-1X. If you want to get a weak analog signal with no HD interference, the Sony must be your choice.

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PS Audio Digital Link III

By Gary Galo, Regular Contributor

PS Audio International
4826 Sterling Drive
Boulder, CO 80301
(720) 406-8946
www.psaudio.com
List Price: \$699



PHOTO 1: Front view of the PS Audio Digital Link III. Operation is very simple, with two buttons toggling input selection and upsampling frequency.

PS Audio introduced their first outboard digital-to-analog converter, the original Digital Link, not long after the first CD players appeared. Like many of us with good ears, PS Audio found the first CD players appalling, and put much of the blame on the poor quality of the D/A converters used in those early players. The Digital Link, and its successor the Ultralink, showed many music lovers the potential of CD reproduction in a way that stand-alone players simply could not.

The Digital Link III is PS Audio's latest outboard DAC, and is a very simple device to connect and operate (Photos 1 and 2). Three digital inputs are included: S/PDIF coax via an RCA connector, Toslink optical, as well as a USB input for connection to a computer. The Digital Link III upsamples all incoming data to either 96kHz or 192kHz, user selectable. LEDs are used to indicate the selected input, the upsampling frequency, and a digital lock. PS Audio uses audiophile-grade, gold-plated, Teflon-insulated RCA connectors, and Neutrik XLR connectors with gold-plated contacts.

CIRCUIT DETAILS

The Digital Link III incorporates state-of-the-art digital hardware (Photo 3). The input receiver is a Cirrus Logic CS8416, a 192kHz/24-bit device with low-jitter clock recovery. The CS8416 supports eight digital inputs, eliminating the need for input switching ahead of the input receiver. I wish that PS Audio had included one or two more digital inputs on the Digital Link III, because many of us have more than two digital sources.

Sample rate conversion is accom-

plished with the Burr-Brown/TI SRC4192, possibly the finest asynchronous sample rate converter chip currently available. The SRC4192 will upsample any incoming data stream to as high as 192kHz, with distortion levels of -140dB from 20Hz up to half the sampling frequency. Like all asynchronous sample rate converters, the SRC4192 attenuates clock jitter, and its jitter performance will ultimately be determined by the stability of the reference clock. PS Audio claims 50pS of clock jitter after they recover the



PHOTO 2: Rear view of the Digital Link III. The DAC has three digital inputs: S/PDIF coax, Toslink optical, and a USB interface for computer connection.

clock "later in the process," which presumably means in the sample rate converter.

The SRC4192 has a bypass mode, which routes the input port directly to the output, bypassing the sample rate conversion. PS Audio has not implemented the bypass feature. However, all digital inputs are upsampled to either 96kHz or 192kHz, as selected on the front panel. Interestingly, the SRC4192 is pin-compatible with the Analog Devices AD1896 used in the Benchmark DAC1 USB.

The D/A converter chip is a Burr-Brown/TI PCM1798, a 192kHz/24-bit device incorporating an 8× oversampling digital filter. The PCM1798 supports Red Book de-emphasis in the digital domain, and Chuck Hansen's measurements confirm that it has been implemented in the Digital Link III. Kudos to PS Audio for enabling the de-emphasis function, ensuring backwards compatibility for your entire CD collection, no matter how old. A Burr-Brown/TI PCM2902 USB codec is used for the USB interfacing, and the TOSLINK optical receiver is an RX142L. The digital chips chosen for the Digital Link III are outstanding performers and are used in many well-regarded professional and audiophile products.

The PCM1798 has true differential current outputs for each channel. PS Audio doesn't like IC op amps, noting that "feedback and speed issues" account for the bright and edgy sound found in op amp based current-to-voltage (I/V) converters. PS Audio believes that slewing-induced distortion (SID) is a problem even with exotic high-speed op amps used for I/V conversion. Their approach consists of a single-transistor I/V converter with no feedback, which they claim completely eliminates the speed and SID problems associated with op amps.

The rest of the analog circuitry reflects the same design philosophy. The I/V converter feeds a passive analog filter which PS Audio says "eliminates the hardness associated with analog filtering." Gain is provided by a fully discrete Class-A amplifier using FETs at the input and low-noise high-current

bipolar transistors for the voltage gain and output stages. The Digital Link III is DC coupled. There's a Texas Instruments TL082 op amp near each output stage, which is probably used as

a DC servo amplifier.

PS Audio has paid careful attention to the power supply in the Digital Link III, beginning with a 32VA toroidal transformer shown in **Photo 3**.

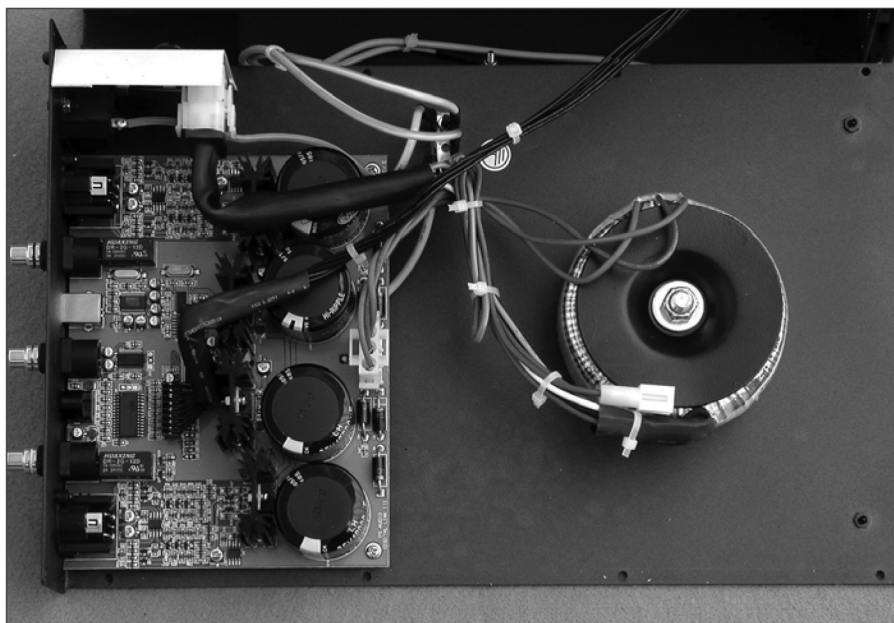


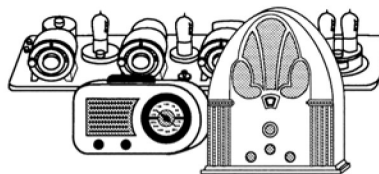
PHOTO 3: Inside view. The Digital Link III uses state-of-the-art digital hardware, a robust power supply, and discrete, zero-feedback analog circuitry.

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The transformer is not as large as the one in the photo on the manufacturer's website, however.

All rectifier diodes are Vishay/General UF5404 high-speed, soft-recovery types—four for the analog supplies and two for the digital. Raw filtering is done with four 12,000 μ F/35V electrolytic capacitors. Regulators include a 7815/7915 pair for the analog supplies, along with a 7805CT and 78M33C for the digital supplies. Two LM317 regulators, near the output stage in each channel, probably provide local regulation, or regulation for the DC servo.

PS Audio recommends isolating the Digital Link III from mechanical vibration. There are many types of isolation feet on the market, some of them fetching extremely high prices. Back in the late 1980s AudioQuest brought Sorbothane isolation feet to the audiophile market. I still use a set from that vintage, and they continue to work extremely well. AudioQuest subsequently replaced the Sorbothane isolators with

a material they call SorboGel. These and several other isolation components are available from The Audio Advisor—www.audioadvisor.com (they also sell the Digital Link III).

Perhaps the “best buy” in audio isolators are the Sorbothane Bumper Hemisphere Mounts sold by Edmund Optics—www.edmundoptics.com, part no. NT35-264 (**Photo 4**). These bumpers are 50mm in diameter, 26mm thick, and cost only \$16.50 for a package of four. Because I've found them to be excellent when used with CD and DVD players, I decided to use them on the Digital Link III as well. I suggest removing the feet supplied by PS Audio and putting these in their place.

PS Audio also recommends power line filtering and a high quality power cord, and suggests their own in both cases. They caution against simple power line filters, noting that they may “bleach” the sound and rob the music of its life and dynamics. I found that my own power line filtering scheme worked extremely well with the Digital

Link III.

Regarding power cords, I use a three-conductor 14AWG “SJT” type power cord fitted with a Marincos 8215 Hospital Grade plug and their 320IEC15 IEC connector (the Digital Link III has a standard IEC power connector on the rear panel). Both are available from Parts Express (www.partsexpress.com). I have no complaints about the sound of the Digital Link III used with my home-brew power cord.

ABSOLUTE POLARITY

The Digital Link III inverts absolute polarity at both the RCA and balanced XLR outputs. The latter is a bit strange because all that's required to correct it is the reversal of pins 2 and 3 on the connector (PS Audio confirmed that pins 2 and 3 are identical, except for polarity, because of the completely symmetrical signal path from the DAC's I/V outputs through the analog circuitry). If you only play digital sources, and they're all routed through the Digital Link III, simply reverse your speaker connections. Otherwise, I suggest making an XLR-to-RCA cable, taking the XLR output between pins 3 and 1, leaving pin 2 disconnected.

Another possibility is using transformer coupling using the “audiophile” version of the Jensen IsoMax CI-2RR that I described in *aX* 1/08 (“Lundahl and Jensen Meet the Monarchy M42”). With this configuration, simply use the inverting inputs to the isolation transformers.

The Digital Link III has a USB interface for connection to a personal computer. Most computer sound cards are mediocre at best. Connecting the Digital Link III to a computer allows you to bypass the sound card completely. As was the case with the Benchmark DAC-1 USB, the improvement in sound quality is stunning.

Because the Digital Link III does not have its own volume control, I recommend either a preamp or a passive volume control between its analog outputs and your power amp's inputs. Otherwise, the playback level will be way too high, and the computer's internal volume control will be operating at the bottom of its range all the

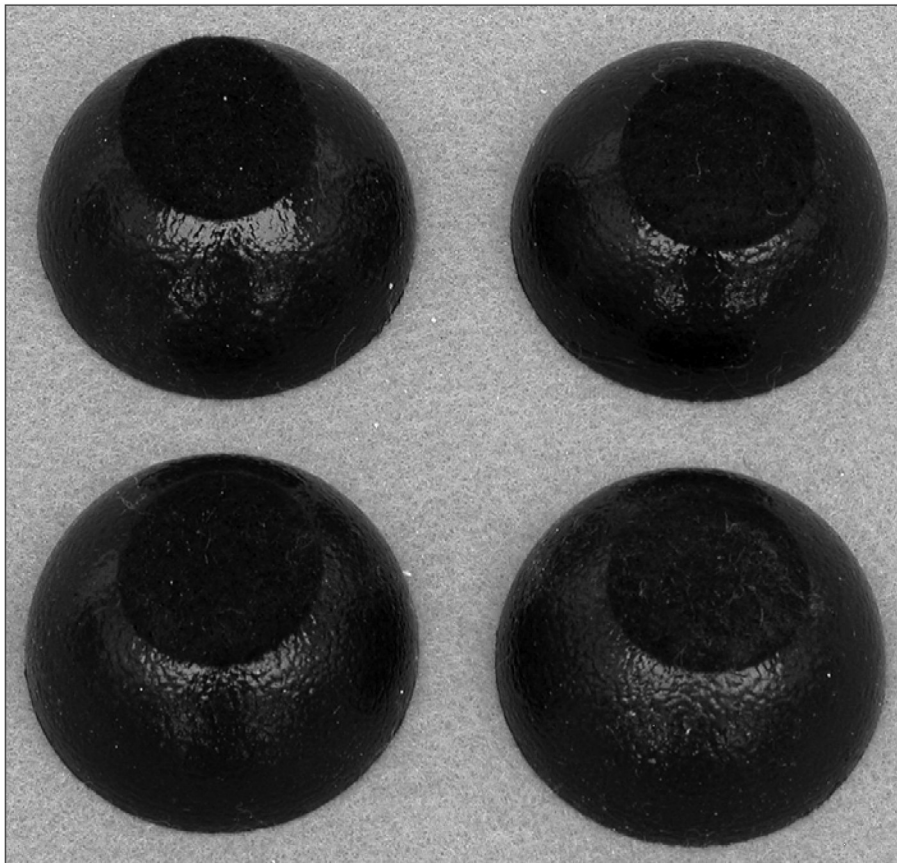


PHOTO 4: Sorbothane Bumper Hemisphere Mounts from Edmund Optical are a cost-effective solution to isolation from mechanical vibration.

time, so you'll have very little control over the playback level. The PS Audio DAC has a power switch on the rear panel, but it can be left on all the time, or connected to a switched power line filter.

THE SOUND

Readers will recall my review of the Monarchy M24 (*aX* Oct. 2007, with measurements by Chuck Hansen in June 2007) and the Benchmark DAC1 USB (*aX* Jan. 2009). Getting right to the point, the sound of the PS Audio Digital Link III is stunning, and is easily the most satisfying of the D/A converters I've auditioned over the past year and a half. With a price tag of just under \$700, it's a truly amazing achievement.

Although the Digital Link III sounds excellent with the upsampling set at 96kHz, it is even better at 192kHz. Chuck's THD measurements show the presence of low-order THD but, unlike the Monarchy M24, the PS Audio's THD is entirely second harmonic. The sound of the Digital Link III is both rich and warm, with palpably realistic instrumental timbres, yet the sound is not overly euphonic. Instrumental timbres simply sound more like live music than on any DAC I've heard, including gorgeous reproduction of massed orchestral strings in the Mercury CD of Respighi's *The Birds*.

The Digital Link III produces a spacious, airy sonic presentation with excellent inner detail. Superficially, the Benchmark DAC1 USB has greater inner detail than any DAC I've heard, but at least part of that is an illusion produced by the somewhat bright sonic presentation of the Benchmark. The Benchmark certainly excels in the area of clarity and detail, but when the tonal balance differences are taken into account, the PS Audio easily holds its own.

The Digital Link III has an extremely neutral and smooth treble region—perhaps the smoothest I've heard—and is more musically satisfying and listenable over the long haul than the more analytical Benchmark. Recordings that lean toward the bright side, such as the Mercury of Schoenberg's *Five Pieces for Orchestra*, can be

too bright on the Benchmark. On the Digital Link III the Schoenberg has the most natural treble region that I've heard from this CD.

The Digital Link III reproduces a soundstage that is larger, with more precise localization, than any DAC I've auditioned. What amazed me was a sense of height information on certain recordings that was elusive on the Benchmark and Monarchy DACs.

Fritz Reiner's recording of *Scheherazade*, particularly the last movement, has always sounded as though the trumpets were placed on two levels on the risers. But, that top-to-bottom localization is much more obvious on the PS Audio DAC. Producer John Culshaw attempted to make the Rhinemaidens appear to come from below the soundstage in Decca's pioneering 1958 recording of *Das Rheingold*. This effect, always obvious on a first-class LP playback system, is beautifully rendered by the Digital Link III.

Dynamics are outstanding. The Digital Link III has no trouble reproducing the biggest orchestral climaxes. Bass has superb weight and impact, and is tight and very well controlled. But to get the best performance you should operate it with correct absolute polarity. The bass drums on the David Hancock-engineered Rachmaninoff *Symphonic Dances* are the best I have heard from this recording. Telarc's recent transfer of their 1979 Soundstream digital recording of Stravinsky's *Rite of Spring* is similarly impressive.

In fact, the Digital Link III narrows the gap between the CD and SACD layers on this disc to some degree (I play the SACD layer on my NAD M55). An inexpensive SACD player is likely not to sound as good as the CD layers when played through the Digital Link III. You need a really first-class SACD playback system to show the superiority of that medium over conventional CDs played on the PS Audio DAC.

PS Audio claims extremely low clock jitter for the Digital Link III, and the CS8416 input receiver and SRC4192 sample rate converter chips are certainly capable of outstanding jitter performance. In my review of the Benchmark DAC1 USB, I wrote that

their UltraLock clock system seemed to virtually eliminate sonic differences between CD transports. In that review, I noted that my reference NAD M55 multi-format player was very difficult to distinguish from the Marantz PDM-340 CD player. The Marantz is normally a rather unrefined transport compared to the NAD.

The PS Audio DAC did not eliminate the differences between the two transports. With the Marantz, the soundstage became shallower, with less precise localization. The Marantz sounded slightly brighter than the NAD, and less detailed. However, the Marantz sounded better with the Digital Link III than it has with any other DAC I've tried. Any outboard jitter suppressor, including my Parts Connection D2D-1 Sample Rate Converter, will probably degrade the sound. (There's a reason why outboard jitter suppressors and sample rate converters have all but disappeared from the market—DACs employing current digital hardware don't need them.) When it comes to wiping out jitter from digital sources, Benchmark still leads the competition, however.

I mentioned the possibility of transformer-coupling of the DAC's unbalanced outputs to maintain absolute polarity via the RCA outputs. The Jensen transformers are so neutral that the Digital Link III sounds fabulous with the RCA output connected to the inverting inputs of my transformer box. The PS Audio DAC doesn't need any additional analog filtering, which the transformer coupling provided for the Monarchy M24, and I wouldn't necessarily spend the money on the Jensen transformers if I didn't already have a pair on hand. But, in listening comparisons I conducted, I found the treble region even smoother with transformer coupling than with the outputs direct-coupled from pin 3 of the XLR connectors.

CONCLUSIONS

The Digital Link III takes the virtues of the Monarchy M24 to the next level, bringing the excellent soundstaging and detail of the M24 into the 192kHz/24-bit world. The PS Audio DAC really combines the strengths

of the Monarchy and the Benchmark DAC1 USB, without the bright tonal balance of the Benchmark.

The Digital Link III provides reference-quality performance at an amazingly low price. As such, it's one of today's best values in audio. It is easily my favorite of the DACs I've

auditioned over the past two years and has found a permanent place in my system. PS Audio is also funding a website called The Audiophile Wiki (www.audiophilewiki.org), which is designed as a knowledge base for audio topics, and is well worth a visit. ■

Table 1 Reference Recordings

CDs

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

Ravel: *Alborada del Gracioso*. L'Orchestre de la Suisse Romande conducted by Ernest Ansermet. London 433 717-2.

Rimsky-Korsakov: *Scheherazade*, Op. 35. Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68568-2 (UV22-Encoded Limited Edition Gold CD version of 68168-2).

Mussorgsky/Ravel: *Pictures at an Exhibition*, especially track 2, "Gnomus." Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68571-1 (UV22-Encoded Limited Edition Gold CD version of 61958-2).

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

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

Wagner: *Der Ring des Nibelungen*, especially "Siegfried's Death and Funeral March" from *Götterdämmerung* (CD 4, Tr. 10-11), and the "Forging Scene" from *Siegfried* (CD 2, Tr. 3-5). Birgit Nilsson, Wolfgang Windgassen, et al. Vienna Philharmonic Orchestra conducted by Georg Solti. Decca 455 555-2.

Stravinsky: *Le Sacre du Printemps*. The Cleveland Orchestra conducted by Lorin Maazel. Telarc SACD-60563 (DSD transfer of 1979 Soundstream digital recording; CD layer only).

DVDs

Rachmaninoff: *Symphonic Dances*, Op. 45. Dallas Symphony Orchestra conducted by Donald Johanos. Classic Records DAD 1004. (96kHz, 24-bit PCM transfer of 30-ips analog tape, or possibly a 15-ips copy, engineered by David Hancock.)

Wagner: *Die Walküre*—Wotan's Farewell and Magic Fire Music. Houston Symphony Orchestra conducted by Leopold Stokowski. Classic Records HDAD 2029. (192kHz/24-bit DVD-Audio transfer of original Everest 35mm 3-track tape.)

PS Audio DLIII DAC Measurements

By Chuck Hansen

SETUP

I used an RCA-terminated S/PDIF coax cable between my Alesis ML-9600 Masterlink recorder digital output and the PS Audio DLIII DAC AES/SPDIF 75Ω RCA input. The default AutoScan mode properly illuminated the COAX LED on the left side of the front panel. In order to provide a suitable warm-up before testing, I played over one hour of 16-bit and 24-bit digital audio test tracks.

The DLIII DAC easily locked onto 16-bit 44.1kHz digital signals, as well as 48kHz, 88.2kHz, and 96kHz with 16-bit to 24-bit-depth signals (I don't have any 192kHz digital test data). There is no indication of the received data bit-depth or sample rate on the front panel, just the one LED that indicates "LOCKED."

The RCA jack analog output signal has inverted polarity with respect to the digital data, positive-going test signals appearing negative-going on my analog oscilloscope. The analog outputs produced a rather high 2.736V RMS (L) and 2.759V RMS (R) with a 1kHz 0dB digital test signal, or a significant 2.79dB maximum above the Red Book standard of 2.0V RMS. The balanced XLR outputs produced 5.485V RMS (L) and 5.530V RMS (R). Again, the analog output signal has inverted polarity using the convention of pin 2 as the signal hot connection.

The output impedance of the RCA jack analog outputs measured 99Ω over the audio band, while the balanced XLR outputs measured 196Ω over the audio band. Hum and noise measured -87dB relative to 2V RMS, and -96dB A-weighted. The analog outputs had a negligible DC offset of 2.7mV. Separation between the stereo channels was excellent at -127dB at 10kHz.

RESPONSES

I recorded the frequency response shown in **Fig. 1** from the RCA analog outputs into a 100k load, using 16-bit/44.1kHz Red Book data, as well as 24/96kHz data into 100k. The 0dB line in this graph is representative of the higher 2.759V RMS output level. The DLIII DAC response drops off by -0.40dB at 20kHz (44.1kHz data), and -1.07dB at 43kHz (96kHz data), with a slight 0.04dB rise at 10Hz with all digital data formats. When I decreased the analog output load (44.1kHz data) to a difficult 600Ω, the frequency response fell by -1.32dB at 1kHz.

Figure 2 shows the deviation from flat response when pre-emphasized Red Book 16/44.1kHz digital data from Gary Galo's detailed De-emphasis Test CD is fed to the digital input. The de-emphasis curve is displaced by -2dB from its normal measured values for clarity. You can see that the DLIII produces a nearly ideal de-emphasis response, unlike many CD players and DACs these days that ignore the de-emphasis bit.

The THD+N versus frequency for the

DLIII analog output using 16/44.1kHz and 96kHz digital data is shown in **Fig. 3**. I used the steep distortion test set 22kHz low-pass filter to remove out-of-band noise. The THD+N is essentially flat over the entire audio band. Decreasing the load from 100k to 600Ω (dashed line) showed only a slight increase in THD+N above 1kHz. Switching the upsampling frequency between 96kHz and 192kHz did not produce any noticeable change in the THD versus frequency measurements.

Figure 4 shows THD+N versus output voltage for the analog output at 1kHz, for both 100k and 600Ω loads. I found it interesting that the THD initially decreases from the 0dB point, reaching a minimum about -4dB. This is about 0.63V RMS below the maximum output measured with a 0dB test signal, so there is a slight penalty for running the maximum analog output signal so much higher than the Red Book 2.0V RMS.

On the other hand, because music rarely peaks at the maximum digital 0dB, perhaps the PS Audio engineers set their

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The DLIII output spectrum of a 50Hz sine wave at 0dBfs into 100k is shown in **Fig. 5**, from zero to 1.3kHz. The THD+N measures 0.0224%, with only the subjectively benign 2nd and 3rd harmonics showing above the -100dB noise floor. There are also no discernible AC power line artifacts. Raising the frequency to 1kHz and expanding the spectrum range to 20.5kHz shows a similar result (**Fig. 6**), where the THD+N measured 0.0239%. Decreasing the load to 600Ω left the THD+N level at both frequencies essentially unchanged despite the -1.32dB drop in analog output.

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Photo: Jürgen Ultee

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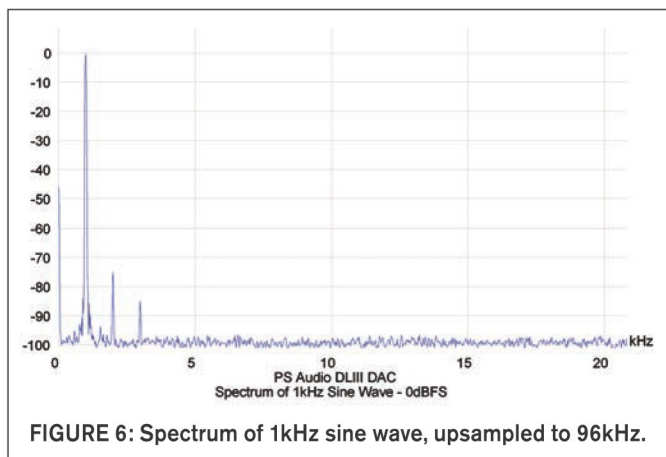
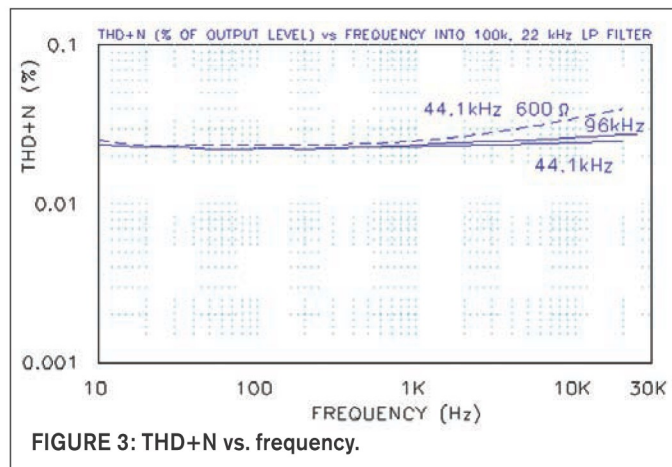
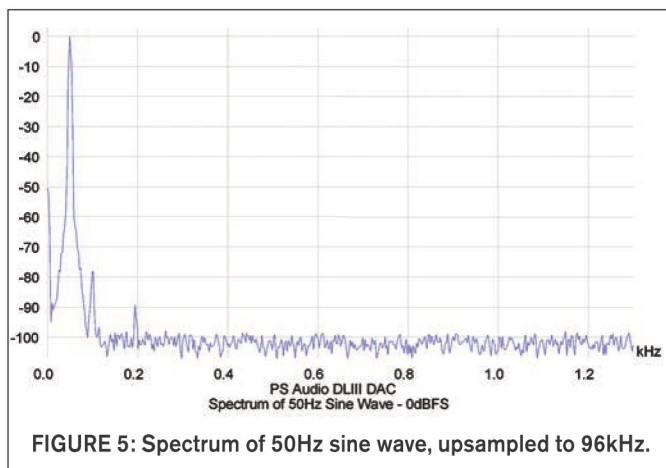
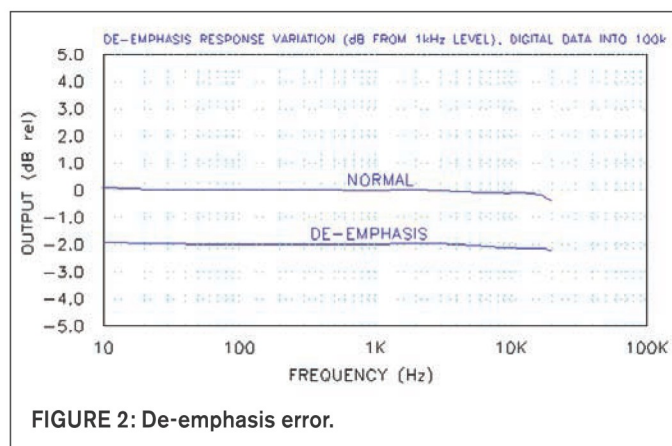
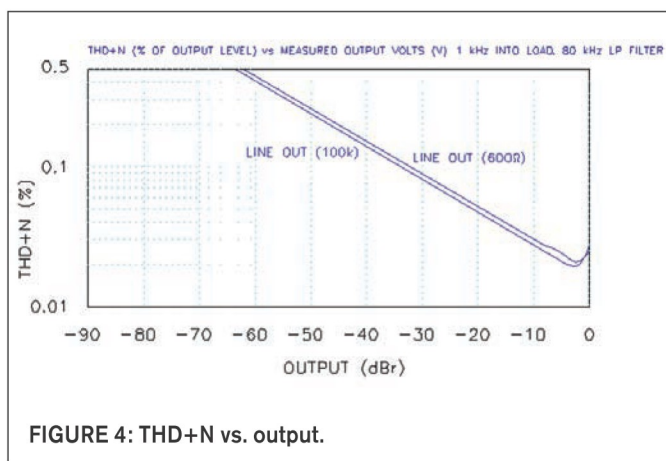
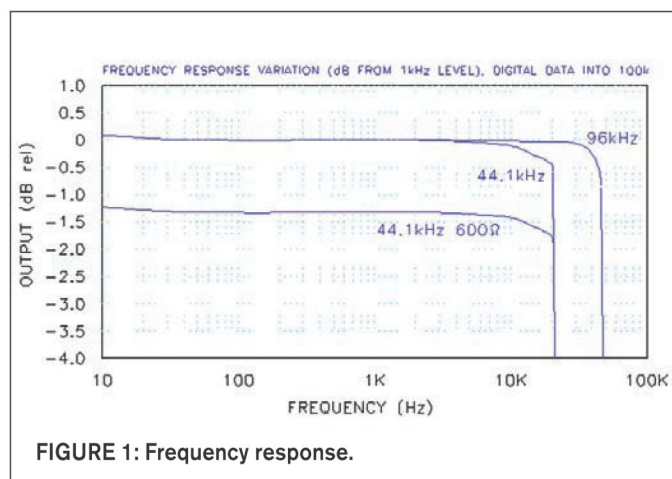
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Manufacturer's Response:

From everyone at PS Audio, we thank

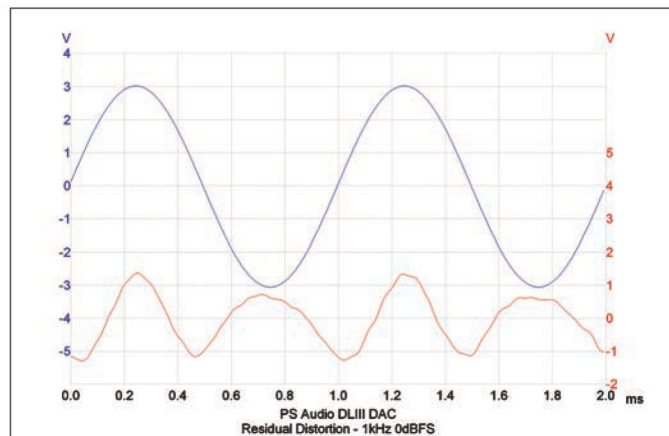


FIGURE 7: Residual distortion of 1kHz sine wave, upsampled to 96kHz.

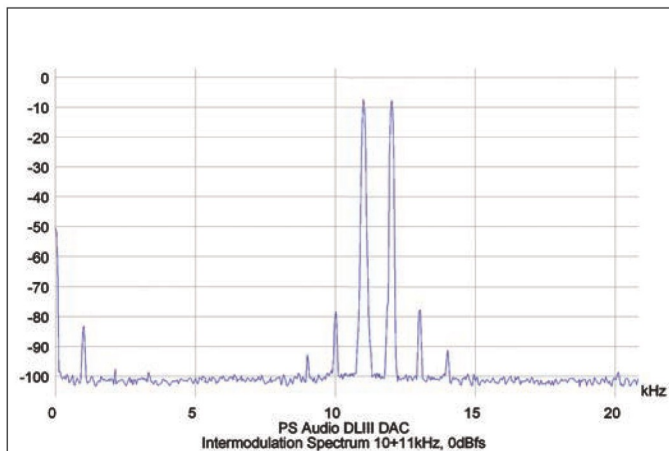


FIGURE 8: 11kHz + 12kHz intermodulation distortion.

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Mr. Galo and Mr. Hansen for the thorough and thoughtful effort they put into this review. We are thrilled that Mr. Galo enjoyed the product so much that he purchased it after his extensive listening.

Please note that our new top-line

D/A converter, the PerfectWave DAC, does not have the analog outputs in reverse polarity. However, absolute polarity is switchable from the front panel so listeners may easily decide which they like best.

Again, thank you very much for the

great review!

Dave Kakenmaster
Vice President of Sales and
Marketing
PS Audio
aX

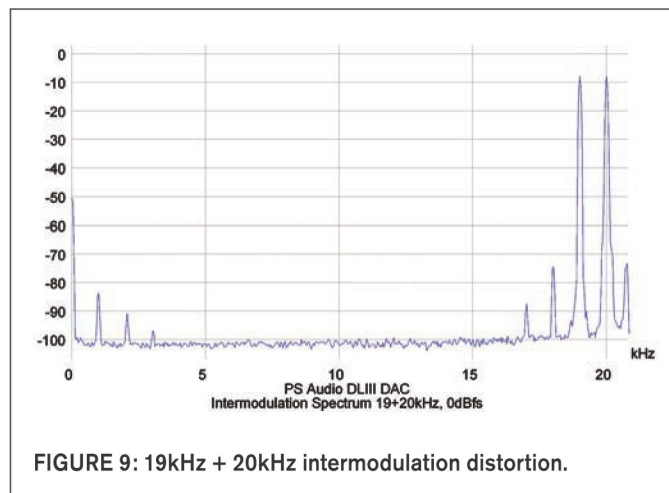


FIGURE 9: 19kHz + 20kHz intermodulation distortion.

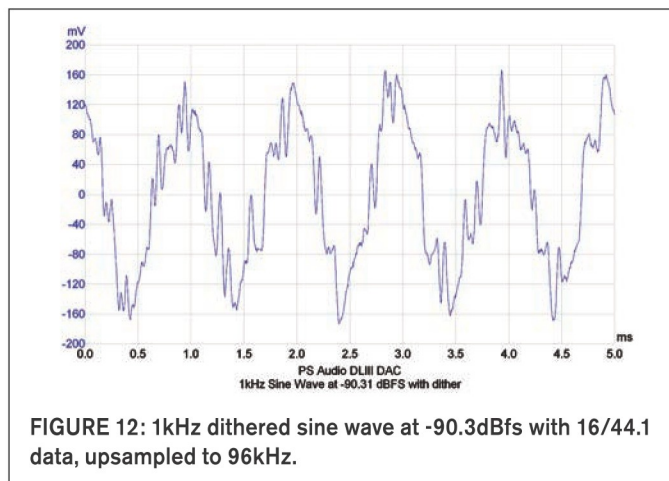


FIGURE 12: 1kHz dithered sine wave at -90.3dBfs with 16/44.1 data, upsampled to 96kHz.

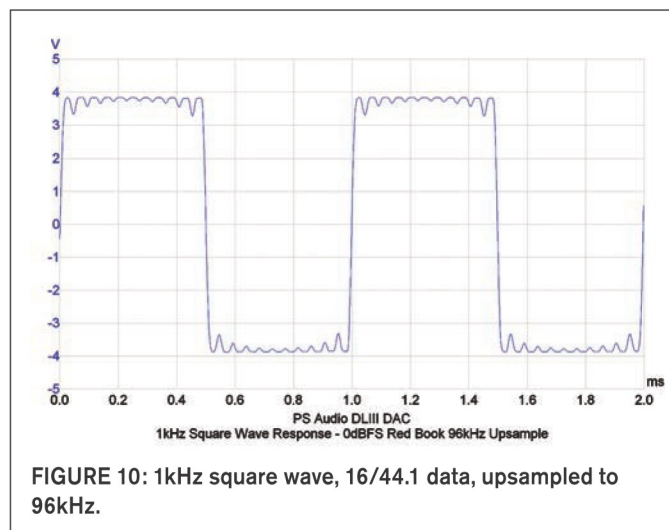


FIGURE 10: 1kHz square wave, 16/44.1 data, upsampled to 96kHz.

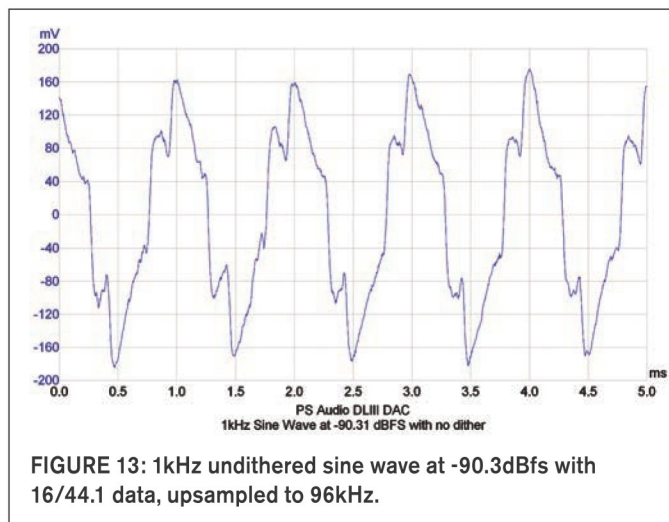


FIGURE 13: 1kHz undithered sine wave at -90.3dBfs with 16/44.1 data, upsampled to 96kHz.

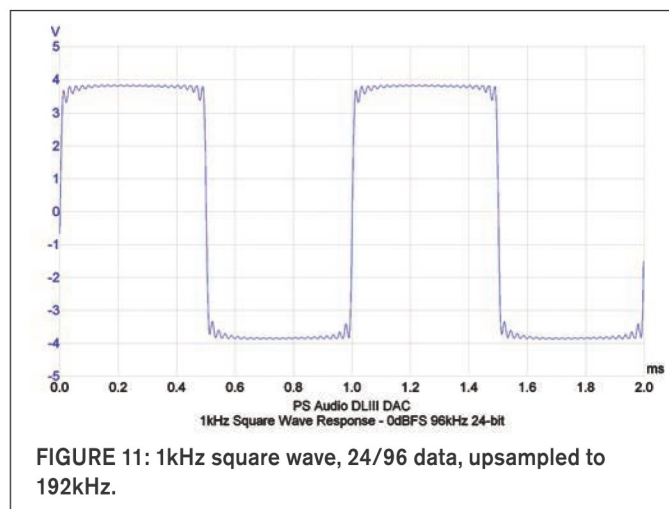


FIGURE 11: 1kHz square wave, 24/96 data, upsampled to 192kHz.

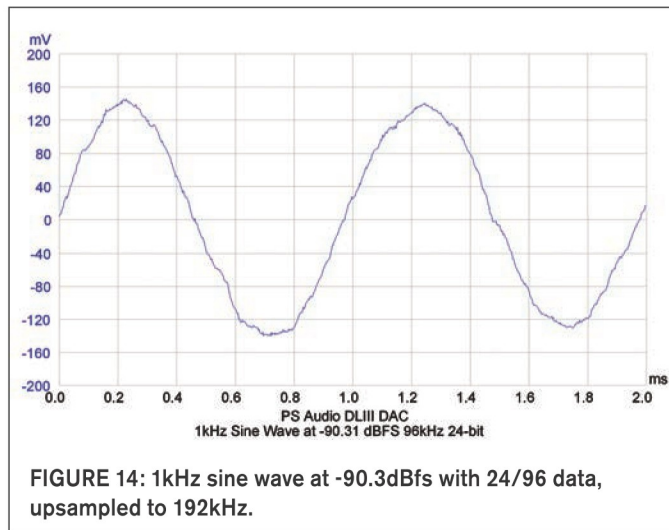


FIGURE 14: 1kHz sine wave at -90.3dBfs with 24/96 data, upsampled to 192kHz.

of the Monarchy and the Benchmark DAC1 USB, without the bright tonal balance of the Benchmark.

The Digital Link III provides reference-quality performance at an amazingly low price. As such, it's one of today's best values in audio. It is easily my favorite of the DACs I've

auditioned over the past two years and has found a permanent place in my system. PS Audio is also funding a website called The Audiophile Wiki (www.audiophilewiki.org), which is designed as a knowledge base for audio topics, and is well worth a visit. ■

Table 1 Reference Recordings

CDs

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

Ravel: *Alborada del Gracioso*. L'Orchestre de la Suisse Romande conducted by Ernest Ansermet. London 433 717-2.

Rimsky-Korsakov: *Scheherazade*, Op. 35. Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68568-2 (UV22-Encoded Limited Edition Gold CD version of 68168-2).

Mussorgsky/Ravel: *Pictures at an Exhibition*, especially track 2, "Gnomus." Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68571-1 (UV22-Encoded Limited Edition Gold CD version of 61958-2).

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

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

Wagner: *Der Ring des Nibelungen*, especially "Siegfried's Death and Funeral March" from *Götterdämmerung* (CD 4, Tr. 10-11), and the "Forging Scene" from *Siegfried* (CD 2, Tr. 3-5). Birgit Nilsson, Wolfgang Windgassen, et al. Vienna Philharmonic Orchestra conducted by Georg Solti. Decca 455 555-2.

Stravinsky: *Le Sacre du Printemps*. The Cleveland Orchestra conducted by Lorin Maazel. Telarc SACD-60563 (DSD transfer of 1979 Soundstream digital recording; CD layer only).

DVDs

Rachmaninoff: *Symphonic Dances*, Op. 45. Dallas Symphony Orchestra conducted by Donald Johanos. Classic Records DAD 1004. (96kHz, 24-bit PCM transfer of 30-ips analog tape, or possibly a 15-ips copy, engineered by David Hancock.)

Wagner: *Die Walküre*—Wotan's Farewell and Magic Fire Music. Houston Symphony Orchestra conducted by Leopold Stokowski. Classic Records HDAD 2029. (192kHz/24-bit DVD-Audio transfer of original Everest 35mm 3-track tape.)

PS Audio DLIII DAC Measurements

By Chuck Hansen

SETUP

I used an RCA-terminated S/PDIF coax cable between my Alesis ML-9600 Masterlink recorder digital output and the PS Audio DLIII DAC AES/SPDIF 75Ω RCA input. The default AutoScan mode properly illuminated the COAX LED on the left side of the front panel. In order to provide a suitable warm-up before testing, I played over one hour of 16-bit and 24-bit digital audio test tracks.

The DLIII DAC easily locked onto 16-bit 44.1kHz digital signals, as well as 48kHz, 88.2kHz, and 96kHz with 16-bit to 24-bit-depth signals (I don't have any 192kHz digital test data). There is no indication of the received data bit-depth or sample rate on the front panel, just the one LED that indicates "LOCKED."

The RCA jack analog output signal has inverted polarity with respect to the digital data, positive-going test signals appearing negative-going on my analog oscilloscope. The analog outputs produced a rather high 2.736V RMS (L) and 2.759V RMS (R) with a 1kHz 0dB digital test signal, or a significant 2.79dB maximum above the Red Book standard of 2.0V RMS. The balanced XLR outputs produced 5.485V RMS (L) and 5.530V RMS (R). Again, the analog output signal has inverted polarity using the convention of pin 2 as the signal hot connection.

The output impedance of the RCA jack analog outputs measured 99Ω over the audio band, while the balanced XLR outputs measured 196Ω over the audio band. Hum and noise measured -87dB relative to 2V RMS, and -96dB A-weighted. The analog outputs had a negligible DC offset of 2.7mV. Separation between the stereo channels was excellent at -127dB at 10kHz.

RESPONSES

I recorded the frequency response shown in **Fig. 1** from the RCA analog outputs into a 100k load, using 16-bit/44.1kHz Red Book data, as well as 24/96kHz data into 100k. The 0dB line in this graph is representative of the higher 2.759V RMS output level. The DLIII DAC response drops off by -0.40dB at 20kHz (44.1kHz data), and -1.07dB at 43kHz (96kHz data), with a slight 0.04dB rise at 10Hz with all digital data formats. When I decreased the analog output load (44.1kHz data) to a difficult 600Ω, the frequency response fell by -1.32dB at 1kHz.

Figure 2 shows the deviation from flat response when pre-emphasized Red Book 16/44.1kHz digital data from Gary Galo's detailed De-emphasis Test CD is fed to the digital input. The de-emphasis curve is displaced by -2dB from its normal measured values for clarity. You can see that the DLIII produces a nearly ideal de-emphasis response, unlike many CD players and DACs these days that ignore the de-emphasis bit.

The THD+N versus frequency for the

DLIII analog output using 16/44.1kHz and 96kHz digital data is shown in **Fig. 3**. I used the steep distortion test set 22kHz low-pass filter to remove out-of-band noise. The THD+N is essentially flat over the entire audio band. Decreasing the load from 100k to 600Ω (dashed line) showed only a slight increase in THD+N above 1kHz. Switching the upsampling frequency between 96kHz and 192kHz did not produce any noticeable change in the THD versus frequency measurements.

Figure 4 shows THD+N versus output voltage for the analog output at 1kHz, for both 100k and 600Ω loads. I found it interesting that the THD initially decreases from the 0dB point, reaching a minimum about -4dB. This is about 0.63V RMS below the maximum output measured with a 0dB test signal, so there is a slight penalty for running the maximum analog output signal so much higher than the Red Book 2.0V RMS.

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The DLIII output spectrum of a 50Hz sine wave at 0dBfs into 100k is shown in **Fig. 5**, from zero to 1.3kHz. The THD+N measures 0.0224%, with only the subjectively benign 2nd and 3rd harmonics showing above the -100dB noise floor. There are also no discernible AC power line artifacts. Raising the frequency to 1kHz and expanding the spectrum range to 20.5kHz shows a similar result (**Fig. 6**), where the THD+N measured 0.0239%. Decreasing the load to 600Ω left the THD+N level at both frequencies essentially unchanged despite the -1.32dB drop in analog output.

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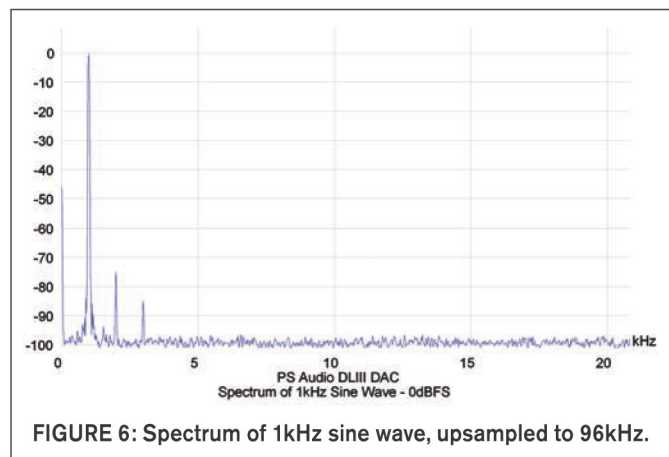
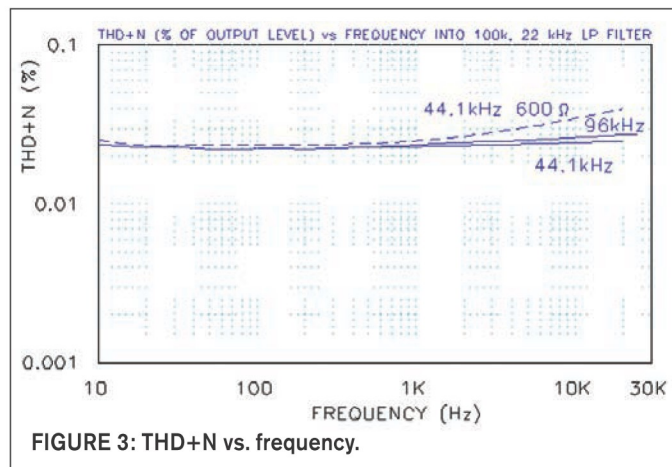
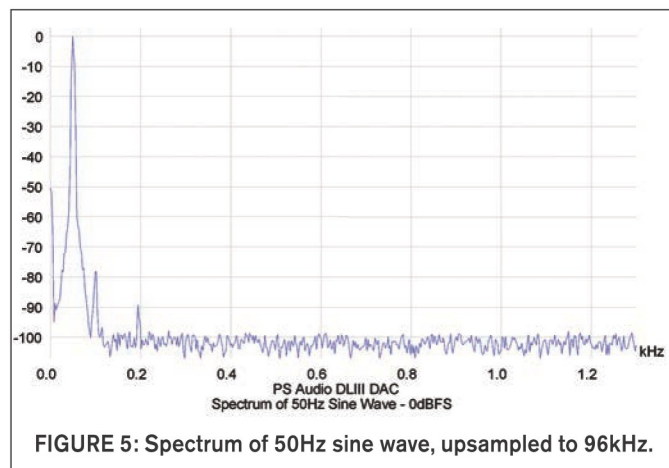
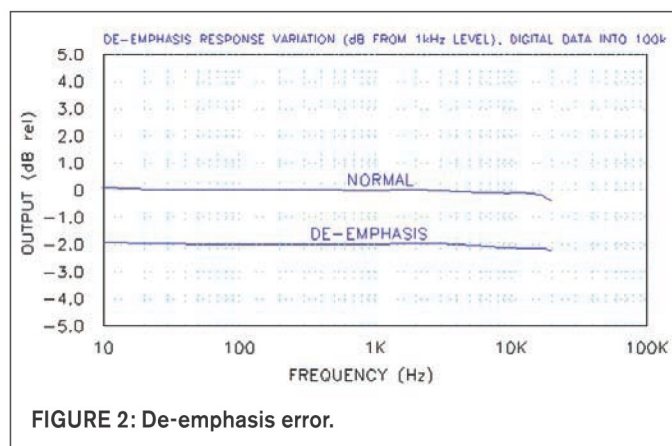
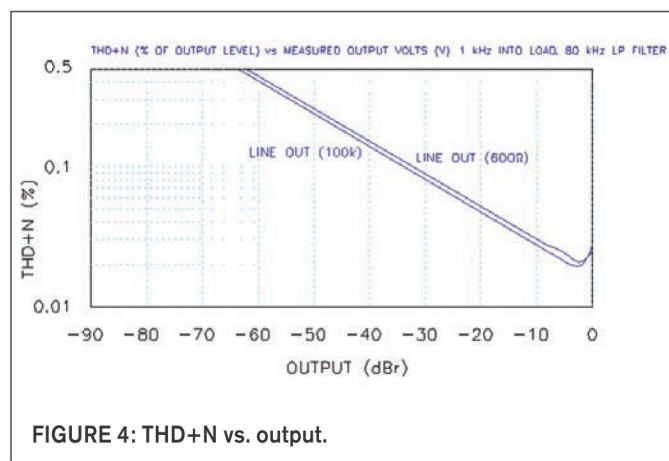
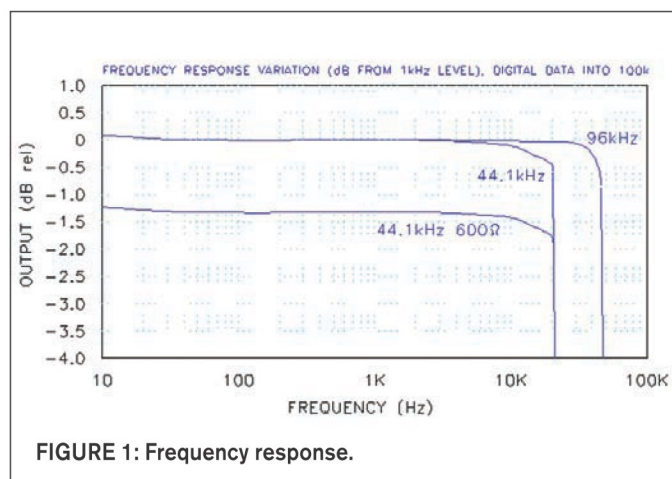
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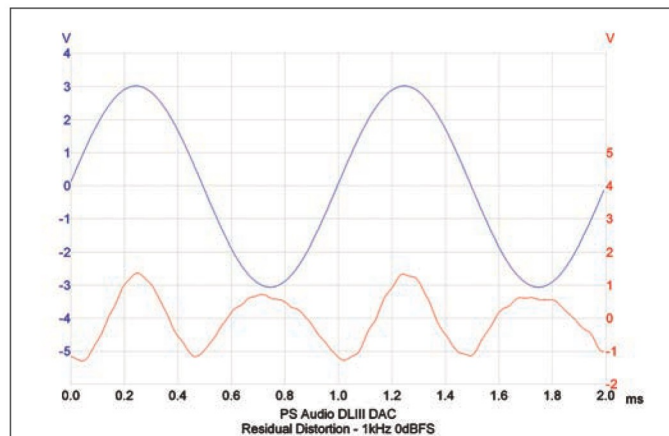


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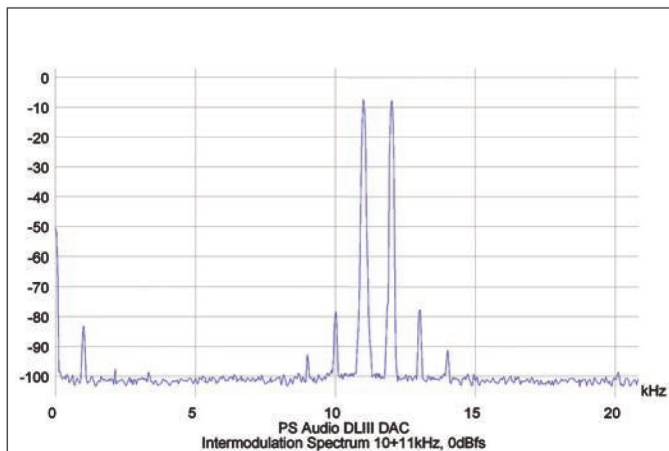


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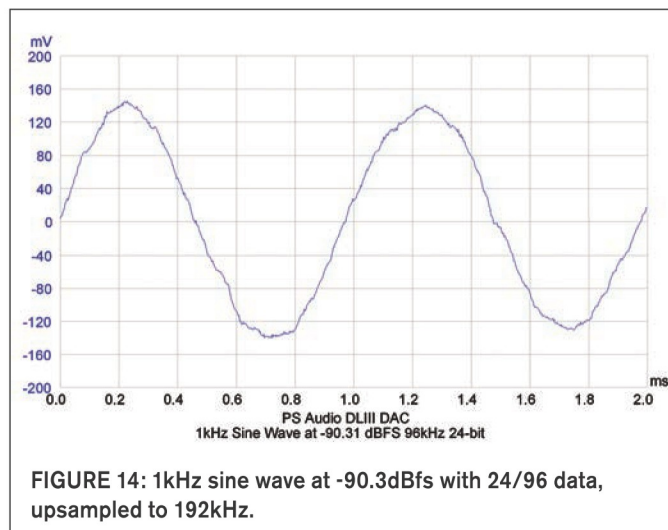
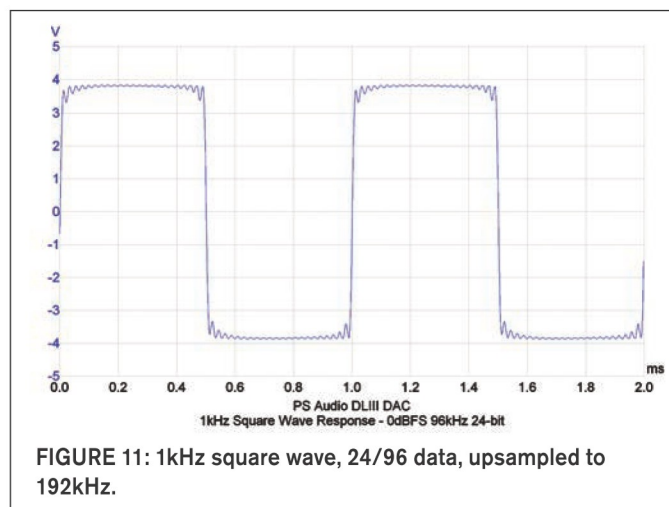
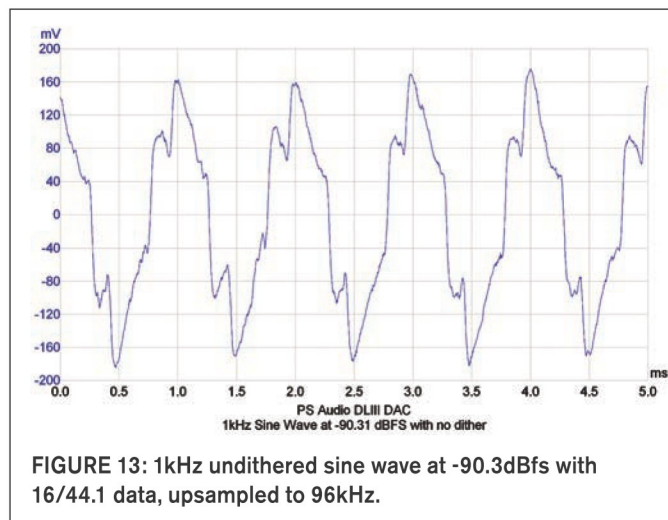
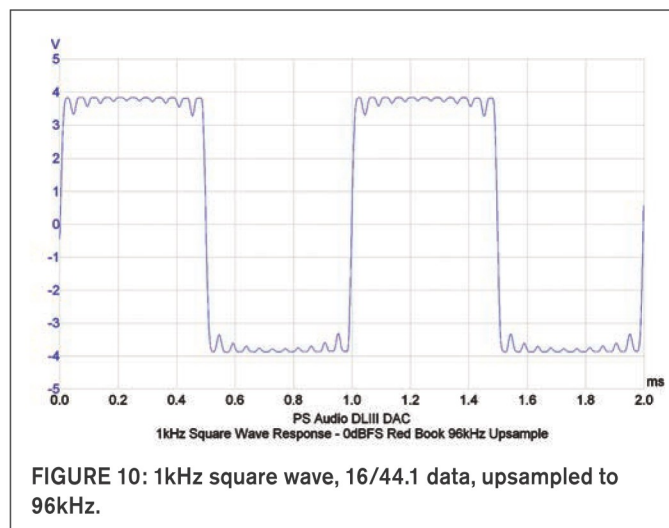
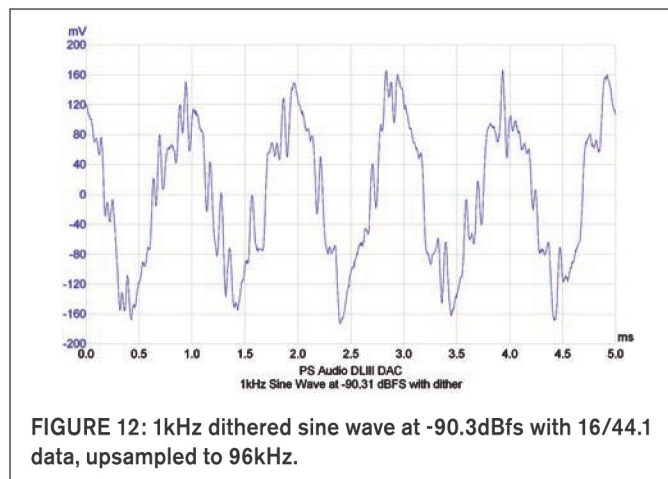
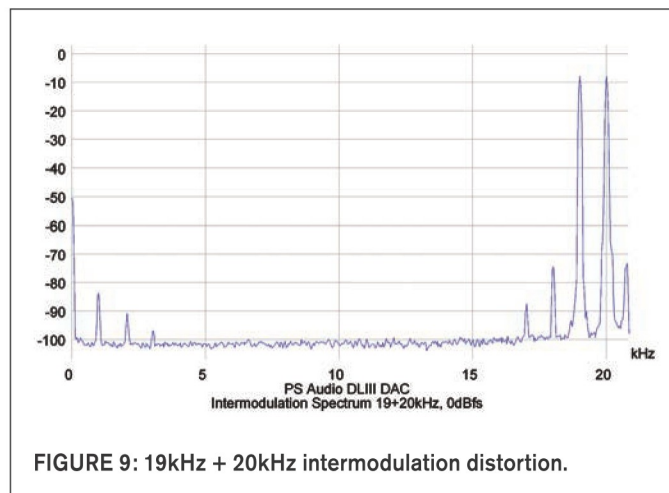
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Again, thank you very much for the

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Dave Kakenmaster
Vice President of Sales and
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PS Audio
aX



IMPEDANCE DISPUTE

Having read the exchange between Mr. Paul and Mr. Yaniger regarding the cathodyne/split-load phase inverter (April and June '09, Xpress Mail), I must support Mr. Yaniger before another audio myth is further perpetuated. It is a common misconception among audio amateurs that the output impedances of a cathodyne phase inverter are inherently unbalanced. This is *not* true, and this persistent falsity should be put to rest once and for all.

Perhaps this confusion arises when considering the output impedance from *one* output at a time, in which case the output impedances do indeed appear unequal. However, because a phase inverter which only uses one of its outputs is fairly pointless, this definition of output impedance is, too. As Mr. Paul correctly points out, provided the anode and cathode are at all times equally loaded, mutual feedback between each output and the grid causes the output impedances to become identical, and therefore so does the frequency/amplitude balance, at least over the normal audio range.

For some reason, Mr. Paul supports his argument by citing page 330 of the *Radio Designer's Handbook* (4th ed.), where it clearly states that "the equivalent source impedance [*for both outputs*]. . . is given by":

$$R_o = \frac{r_p R_L}{r_p + R_L(\mu + 2)}$$

where $R_L = R_p = R_k$

Perhaps Mr. Paul did not read all the way down page 330. In any case, this mathematical puzzle was also solved over half a century ago in:

Jones, G. E., (1951). Analysis of Split-Load Phase Inverter. *Audio Engineering*, 35 (12), pp 16+40-41.

And later by:

Priesman, A. (1960). Notes on the Cathodyne Phase Splitter. *Audio*, (April), pp 22-23.

I will not repeat the mathematical derivations here, as both seminal papers may be accessed on www.thermionic.info, and www.aikenamps.com/TI_Other.htm, respectively.

Mr. Paul also argues that "no justification is given for [the requirement for identical test loads] in any engineering textbook that I have ever seen." This is not surprising, because it is implied. If you test the circuit with only one output loaded, then you are not using the circuit as a phase inverter; if you do not apply all the necessary loads, then you are not testing the same circuit! No textbook would bother to make such an obvious statement.

As an academic curiosity, a similar process applies to an op amp arranged for balanced-to-single-ended operation, with equal value resistors used throughout. Mutual feedback to each input forces the input impedances to become equal despite their "apparent" inequality when considering only one input.

I can also supply an electronic copy if Mr. Paul wishes to read:

Peters, D. P. (1957). In Defense of the Split-Load Phase Inverter. *Radio & TV News* (November), pp 182+184.

Merlin Blencowe

merlinblencowe@hotmail.com

I read with interest the discussion between Mr. Yaniger and Mr. Paul about the output impedances of the phase splitter described in Mr. Yaniger's article. Mr. Yaniger states that the output impedance at the plate and cathode of the stage are equal as long as the load impedances are equal. Mr. Paul disputes that claim. At first blush it would seem that Mr. Paul's assertion must be correct, because it is well known and provable that a common plate stage (cathode follower) has a much lower output impedance than a common-cathode stage. However, the special case described by Mr. Yaniger—i.e., equal output impedances—is intriguing, so why argue about the point when some simple circuit analysis will provide a definitive answer?

For clarity, let me review what I mean by output impedance. A common way to model the output impedance of any volt-

age source is to consider such a source as consisting of a perfect voltage source—one with zero output impedance—in series with a resistor.

In **Fig. 1**, R_g represents the voltage source's output impedance, and R_L represents the load. For simplicity, I've made the assumption that the source's impedance is not frequency dependent. I think it will be clear to most readers that the value of R_g can be calculated by varying R_L and noting how V_{out} changes. If

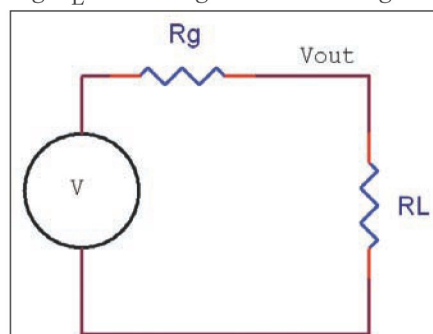


FIGURE 1: Impedance model.

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you had two voltage sources and wanted to compare their output impedances, it would be sufficient to test them both with the same varying load resistances. If their outputs changed equivalently, then, by definition, their output impedances are equal.

Now on to the phase-splitter stage. Let's model the stage using a simplified hybrid-pi model (**Fig. 2**).

V_g represents the grid voltage. The current source represents the tube's transconductance, so the value of the current equals $g_m V_{gc}$. R_p is the plate

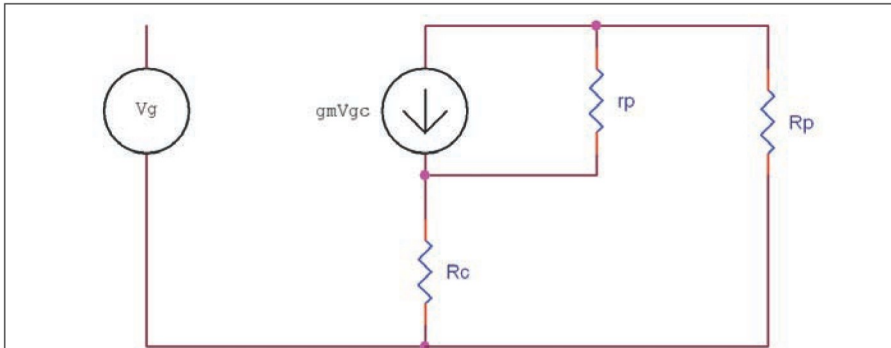


FIGURE 2: Example phase-splitter stage.

CONTRIBUTORS

Jan Didden ("paX, Part 1," p. 6) built his first OTL amp with 807 tubes 35 years ago. He has built speakers, preamps, and tape recorders, but is most interested in power amps, especially using error correction as discussed by Hawksford. Many of his projects have been published by *audioXpress*. Now retired from a career with the Netherlands Airforce and NATO, he tries to complete all those half-finished projects accumulated for lack of time. He now also has the time to travel to interesting audio events and interview audio luminaries. His projects are documented on his Linear Audio website.

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Chuck Hansen (PS Audio DLIII DAC Measurements, p. 40) is an electrical engineer and holds five patents in his field of engineering. He began building vacuum-tube audio equipment in college. He plays jazz guitar and enjoys modifying guitar amplifiers and effects to reduce noise and distortion, as well as building and restoring audio test equipment. He enjoys sailing and has over 150 magazine articles to his credit.

resistor, R_c is the cathode resistor, and r_p represents the tube's internal plate resistance (the reciprocal of its output conductance). The principle simplification in this model is that none of the tube capacitances are shown. Strictly speaking, the model is therefore only accurate at DC. I am making the untested assumption that, within the bandwidth of interest, the capacitive reactances are so much larger than the plate and cathode resistors as to be of little import.

A quick look at the model shows that R_p and R_c are in series with each other, and in parallel with r_p . With that insight, the equations for the circuit can be easily derived. For better or worse, I have named voltages for the resistors they appear across. Also, I never design with tubes, so please excuse any labels that don't follow accepted convention.

$$\text{Equation 1: } V_{r_p} = g_m V_{gc} \frac{r_p(R_p + R_c)}{r_p + R_p + R_c}$$

$$\text{Equation 2: } V_{R_c} = V_{r_p} \frac{R_c}{R_c + R_p}$$

$$\text{Equation 3: } V_{R_p} = V_{r_p} \frac{R_p}{R_c - R_p}$$

$$\text{Equation 4: } V_{gc} = V_g - V_{R_c}$$

It's not necessary for the question at hand, but from these equations, you can derive the transfer function

$$\frac{V_{r_p}}{V_g} = \frac{g_m r_p (R_c + R_p)}{r_p + R_c + R_p + g_m R_c R_p}$$

Now, remember what I previously said, that if two voltage sources behave equally when tested with varying load resistors, those two load sources, by definition, have equal output impedances. Referring to equations two and three, you can see, in fact, that for any value of R_c and R_p , that V_{R_c} and V_{R_p} are equal to each other as long as R_c and R_c are equal to each other. Therefore, I have proven Mr. Yaniger's assumption that if the impedances at the plate and cathode are equal, then the stage does have equal output impedance at cathode and plate. Let me note for completeness that the equal impedances at both nodes would consist both of the discrete plate and cathode resistors as well as the input impedances of the following stage each node is connected to.

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aX