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Greening of the Behringer Ultra-Drive Pro DCX2496

By Jan Didden

You'll enjoy this mod project, which adds a replacement analog output board to this useful Behringer crossover unit.

Face it, the world is analog in nature. Yet, digital technology has its advantages. Witness the Behringer Ultra-Drive Pro DCX2496 (I'll call it the DCX from now on), with its two analog balanced stereo inputs, an AES/EBU digital input, six ADCs, a Digital Signal Processor (DSP), and six DACs. Plus a handful of op amps to size the analog signal up or down, and six analog balanced output channels. You can use the DSP for a variety of signal processing before sending a signal to any of the six outputs.

One obvious use of the six processing channels is for a digital three-way stereo crossover. In stereo use, you can link each pair of output channels (low, mid, high) so that any setting you make to one side is automatically copied to the other side. The crossover settings include filter frequency and filter type (Butterworth, Bessel, or Linkwitz-Riley, from 12dB/oct to 48dB/oct).

You can set the level for each output individually, set the input levels, and also insert bandpass or band reject filters, with varying Q and lift/drop levels to correct humps and troughs in your speakers' frequency response. You can modify the lf response to correct room errors. There is even an auto-delay mode (if you have the optional mike) to correct for unequal delays from speakers to the listening position.

There is no limit within the capabilities of the DSP, and the display will show you even how much, in %, remains of its processing power. You can save a particular setup with a name of your choice so you can have a series of setups if you use your unit at different places or systems (or music). And all that for a street price around $300 US. I have used this unit for two years now, and love its quality, flexibility, and well-thought-out user concept, and even bought a second one for my other system. But, being an audio DIYer, I couldn't leave well enough alone.

Because I stay away from DSPs and other

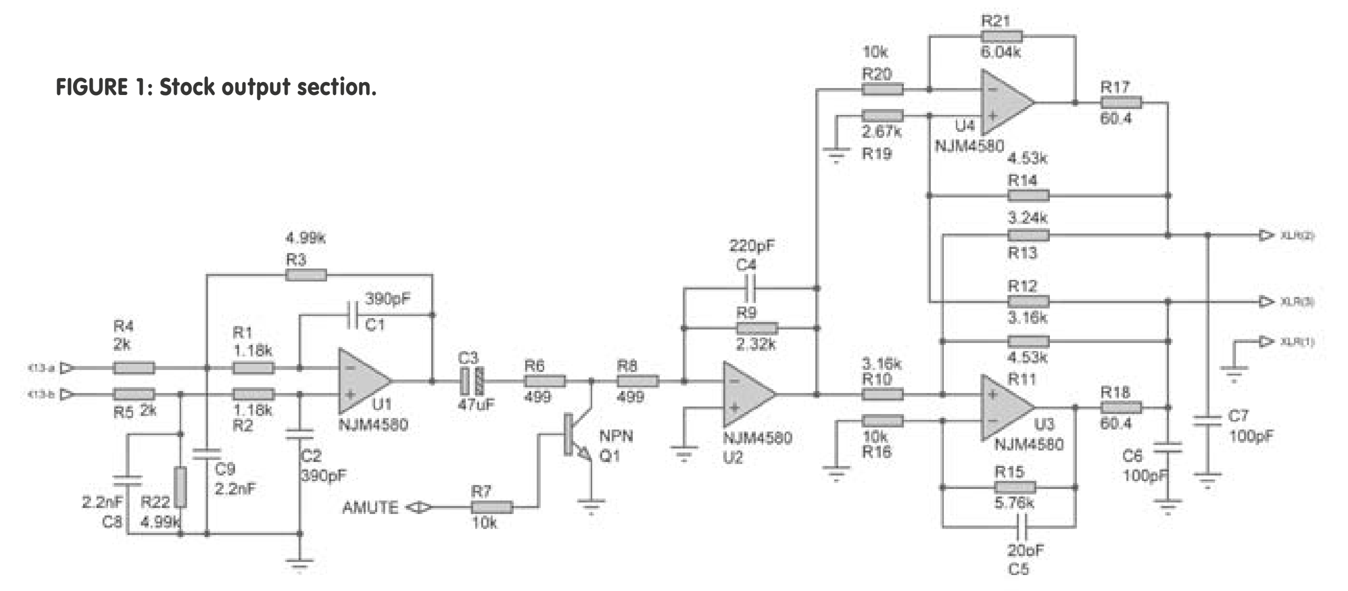

FIGURE 1: Stock output section.

magic, this article is about improving the analog portion of the DCX, in particular the output amplifier section. This is a low cost unit, so those op amps aren't so hot, and the judicious use of electrolytic signal coupling caps (arrgh!) and—Ohm forbid!—the use of a transistor for muting means that there is much that can be done better. But, I also wanted it to be fully reversible.

The end result is a plug-in replacement board between the DAC outputs and the outside world. The original unit is fully balanced, and I have kept that, but I must confess that—until now—I use it single-ended.

A STACK OF OP AMPS

Figure 1 shows the output section of the stock DCX. This is repeated six times for each of six output channels. Take a walk through it. The DAC is a balanced voltage-output AK4393, and its outputs are connected to X13-a and X13-b, onward through a whole lot of active circuitry, to the balanced XLR output connectors on the rear panel [XLR (1), (2), and (3)]. X13 is an existing flatcable header from the DSP board to the output board, so you can see that by tapping this flatcable you can access the analog signal at will without any hardware changes.

Look at the detailed circuit. Op amp U1 is a second-order active filter, as is usual after a DAC, but it also amplifies the signal 2.5 times and converts the balanced signal to unbalanced or single-ended, if you will. Transistor Q1 is the mute transistor. If the signal AMUTE is high level, the transistor is fully conducting, thus presenting a very low impedance to the signal line and effectively shorting it to ground.

So far so good, but even when the signal is not muted, Q1 is still in the system. Its collector represents a nonlinear impedance and nonlinear capacitance, and together with R6 this will form a nonlinear signal dependent attenuator. I know from experience—for instance, in the early days—when this low-cost mute was often used with CD players, that this can cause distortion of the signal. So, that transistor needs to go!

By the way, that C3 is also not a good idea, but is required here. The DAC outputs the signals on a common-mode voltage of half its supply voltage, or 2.5V DC. Any small DC difference between the two inputs becomes amplified also, and could cause a serious offset further down the line, thus C3.

Next is U2, a buffer with some 2.3 gain, feeding the single-ended to balanced converter U3, U4, which also slightly attenuates the signal (the overall gain of this chain, by the way, is just above 10dB). U3 and U4 are cross-coupled. If you use the output single-ended and ground the unused pin, this will zero the cross-coupled feedback and increase the level of the used output by 6dB.

I wondered about the conversion: first- to single-ended and then again back to balanced. My guess was that it would be much more difficult to implement a mute on a balanced signal at 2.5V DC, and my later experiments confirmed it.

All in all, there is much gain in this chain, and the reason is that this is a pro unit, and pro signal levels are in practice 14dB above home hifi levels. It's a pain because it means you have less control range available on your volume control. So, for home use you really don't need all that gain.

I would like to keep the mute function and the filtering, of course. My new board modification therefore consists of a passive output network with the filtering, and the mute function with reed relays.

GOING PASSIVE

Figure 2 shows the diagram of one channel of the new output stage. The signal from the X13 header is connected to a first-order symmetrical passive filter (fancy name for R3, R4, and C1) through coupling caps. More on these caps later.

Then there is a second single-order filter implemented with R1, R2, and C2. RL1 is the mute relay, which shorts both signals to ground when not activated. A resistor from each signal phase to ground, R5 and R6, provides a DC path to ground in case the equipment after the DCX is AC-coupled. They can also be used to attenuate the signal if needed.

The values of the filter resistors are chosen to present at least a resistance of more than 1k to each DAC output even if the output is shorted. You can load the DAC with 600Ω for AC, but the DC load should not be lower than 1k; I selected 950Ω total for a safety margin. Select the filter capacitors to get as close as possible to the second-order active filter shape in the original unit.

Figure 3 shows the frequency and phase response of the original version and my implementation (simulated), which is a tiny bit lower in level at 20kHz but has less phase shift. At any rate, I think the differences are too small to be of audible consequence. Finally, **Fig. 4** gives the full circuit.

THE MUTING

The muting circuit gave me some headaches. When you turn on the power to the unit, the AMUTE line (**Fig. 1**) is set high by the DSP and the signal is muted. After a few seconds, the mute is released and sound will be heard. (This is independent of the software mute available for each output which you can activate at the front panel.) In the stock unit, there are no audible clicks at switch-on or switch-off.

I had a choice: mute by shorting either the two output phases together or each

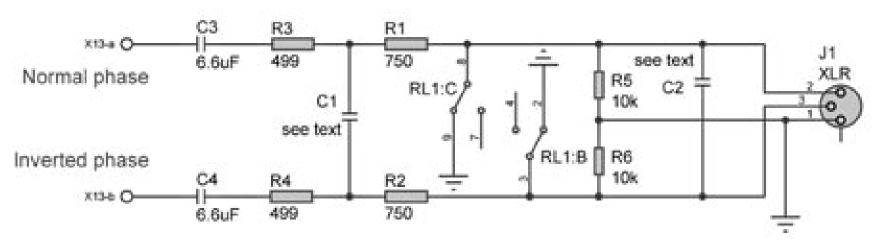

FIGURE 2: New output section.

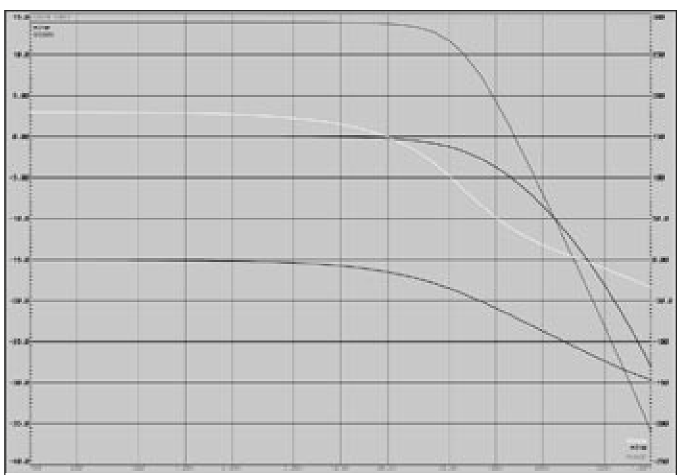

FIGURE 3: Frequency and phase response of the stock and the new output section (simulated).

phase to ground. I tried the first possibility because it needed only half a relay per channel, but I couldn't get rid of the switch-on noise. The mute signal from the DSP just came too late (it first needs to boot after power-on, I guess). Therefore, I used the relays in normal-on mode (**Fig. 4** again). Both signal lines are shorted to ground until the relays are activated.

Two conditions need to be met for this. First, C26 must be charged up through R26. This takes a second or so, and bridges the time between power-on and setting of AMUTE by the DSP. After the DSP has settled and releases the mute signal, the relays switch and sound is heard. Importantly, at switch-off, or if the power drops out, there is absolutely no noise at all.

BUILDING THE NEW PCB

The final PCB with stuffing guide is shown in **Fig. 5**. **Table 1** is a parts list. I was fortunate to locate (after a long search) the same exact XLR male output connectors Behringer uses, making for a true form-fit and function replacement board. There is only a handful of other passive components. Because you will most probably use this unit as an electronic crossover, you can size the coupling caps with the expected frequency range in mind.

Because it makes sense to use channels 1 and 2 for the low-frequency range, you need to use relatively large caps there (C14, C15, C20, C21). But as they will not be asked to carry a signal more than a few thousand hertz at the very most, they don't need to be exotic types. I used two parallel 3.3µF/50V polyester film types, which will give a low-frequency cutoff of about 5Hz with a 5k load. If the load is higher, of course, the lf –3dB point decreases. Similarly, I use larger passive filter caps (C1, C2 in **Fig. 2**) for the low-pass channels, 1500pF versus 820pF for the mid- and high channels. For the latter, rolloff should only start after 20kHz, but for the low-pass channel we don't care if rolloff starts at 10kHz or so, it only helps to limit hf noise.

On the third and fourth channels (mid frequency band), where I don't expect any signal below say 100Hz, I use a single cap. Finally, for the last two channels (high-frequency band), I used 0.68µF. Because these have a low value, using high quality types here doesn't cost you an arm and a leg. At any rate, I made the capacitor outline with multiple holes to accept several types and sizes of caps.

The resistors used for the filters are run-of-the-mill 2% metal film. Because the reed relay normally is not in the circuit, it also doesn't need to be exotic; however, the board is laid out for Omron type G6H-2-DC12 because I had a box full of those (thanks, Leo). But, since most reed manufacturers use similar layouts, you can probably use other types, too. Just make sure they are dual normally-on 12V types and don't consume too much current (12mA each for the Omrons).

GETTING CONNECTED

The new PCB is connected to the DSP board using existing headers. There are two headers on the DSP board—X1 for the input and output signal lines and X2 for the RS232, midi, and power supply lines. You will need to make a new set of flat cables with connectors to hook up the new PCB to the DSP PCB. The new cables from the DSP board are split and

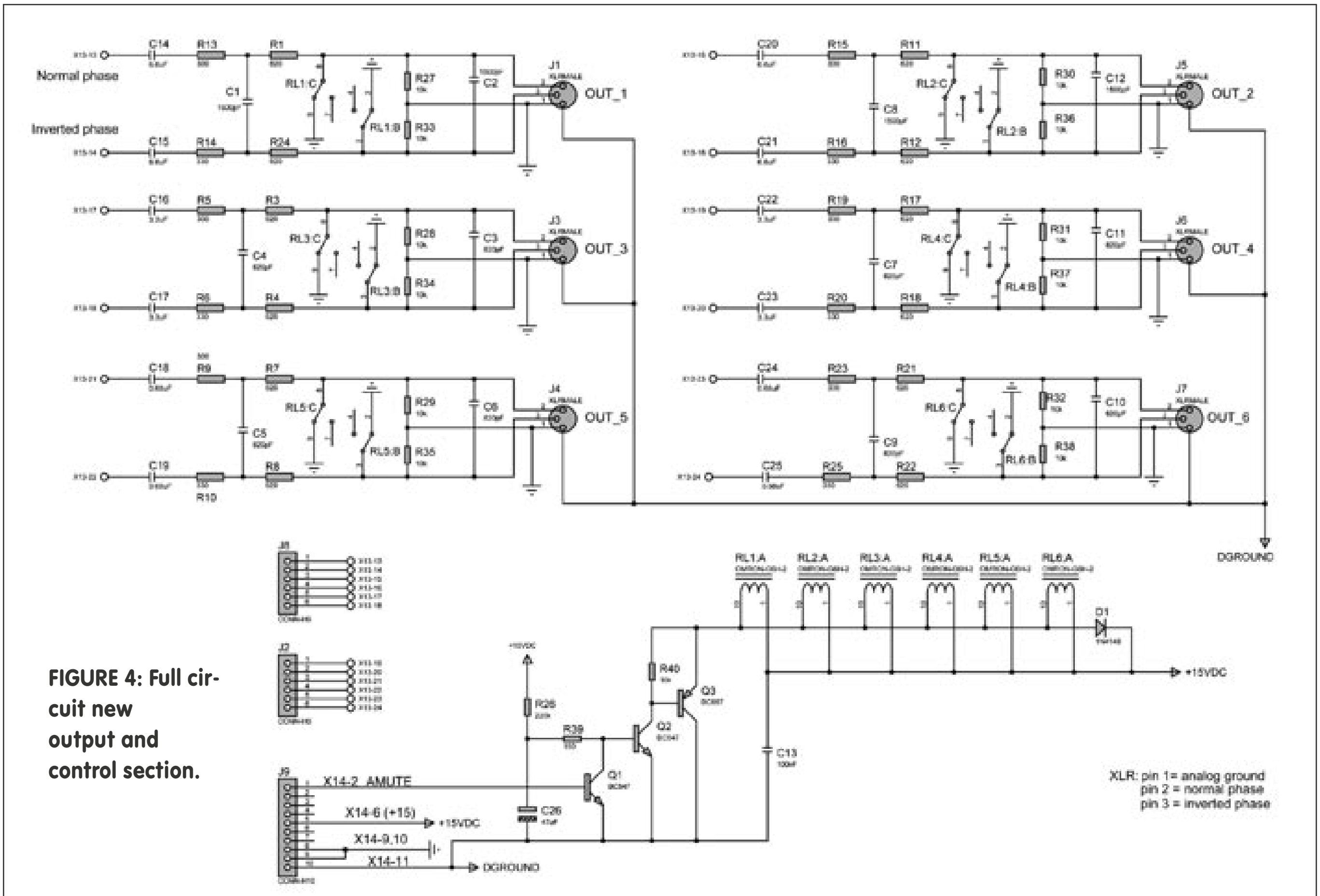

FIGURE 4: Full circuit new output and control section.

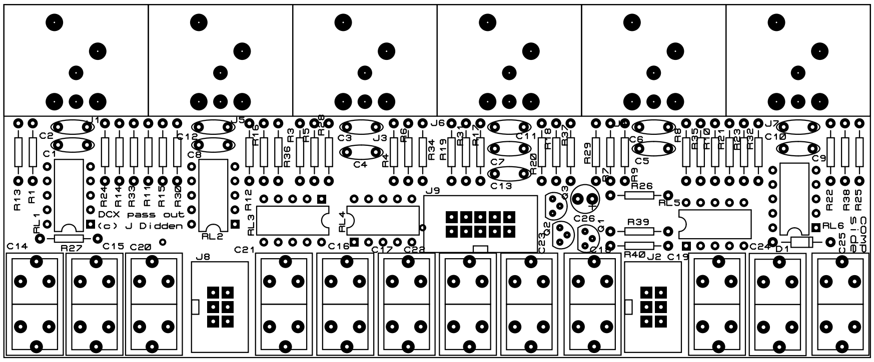

FIGURE 5: Output PCB.

fixed to several connectors to join the new output board and the new serial interface board.

The cable plan is shown in **Fig. 6**, while **Photos 1** and **2** show the cable run in the modified unit. **Photo 3** shows the cable set you need to build. Not all wires are used; for instance, there is no need to run all supply lines to the new board (except +15V for the reed relays), and the analog input lines are also not used (but see next section). So you should be careful with the cable alignment in the headers, because in some cases the flat cable has fewer wires than the header pins and the cable needs to be aligned left or right. **Figure 6** and **Photo 2** should make that clear.

Double-check the wire numbers before fitting the connectors. It is not difficult to do, but it's a pain to redo if you make an error. Both my units ran fine on the first try.

TABLE 1.

PART REFS	VALUE
Resistors (all 0.25W 1% metal film)	
R1, R3, R4, R7, R8, R11, R12, R17, R18, R21, R22, R24	620
R5, R6, R9, R10, R13-16, R19, R20, R23, R25	330
R26	220k
R27-R38, R40	10k
R39	150
Capacitors	
C1, C2, C8, C12	1500pF (mica or silver film)
C3-C7, C9-C11	820pF (mica or silver film)
C13	100nF/50V
C14, C15, C20, C21	6.6µF (2 parallel WIMA MKS2 pol. 3.3µF/50V DC Farnell # 107-423)
C16, C17, C22, C23	3.3µF (WIMA MKS2 pol. 3.3µF/50V DC Farnell # 107-423)
C18, C19, C24, C25	0.68µF (EPCOS BS32652 polyprop. 0.68µF/250V, Farnell # 400-3718)
C26	47µF/25 electrolytic
Transistors	
Q1, Q2	BC547 or equivalent small-signal NPN
Q3	BC557 or equivalent small-signal PNP
Diodes	
D1	1N4148
Miscellaneous	
J1, J3-J7	XLRMALE (Neutrik NC3MAH, Farnell # 724-543)
J2, J8	CONN-H6 (flatcable conn 2 × 3 pins 0.1″ grid)
J9	CONN-H10 (flatcable conn 2 × 5 pins 0.1″ grid)
X13	header 26 pin (flatcable conn 2 × 13 pins 0.1″ grid)
X14	header 20 pin (flatcable conn 2 × 10 pins 0.1″ grid)
RL1-RL6	OMRON-G6H-2 or equivalent (see text)
20″ flatcable 26-way 0.05″ spacing	

FINAL CONSTRUCTION

Mounting the completed board in the unit is straightforward, once you remove the old board (at which point you void the guarantee, I guess). Remove the existing flat cables and all the screws in the various plugs and sockets on the rear panel. You can then take out the complete I/O board. The new board is fixed with the original screws used to hold the XLR output connectors.

I taped a plastic sheet to the bottom of the enclosure to avoid any contact of the solder joints with the metal as the distance is quite short. The small serial port board also uses the original screws that held the RS232 socket at the rear panel. Next put in the new cables—first the analog one, then the one for the control—and serial I/O lines.

WHAT DO I LOSE?

My modification replaces the entire original back panel circuit board, which includes several sections. One section containing the six analog output channels is replaced by my analog passive output PCB. There are two control input sections.

One is for a serial connection to a PC, which allows you to control the DCX remotely, to store and save different setups on your PC and to download firmware upgrades. This functionality is maintained via a small additional PCB (**Fig. 7**). The other is a jack to connect multiple DCX units in series so that they can be controlled by a single PC (this is pro stuff, remember!). That functionality is lost but would most probably not be used in home setups anyway.

The original board also contains analog and digital (AES/EBU) inputs. In fact, in the stock unit the analog and digital inputs share female XLR jacks and are selected from input channel A via a normal low-level relay. Don't ask me what that does to the digital signal integrity!

Because I use the DCX only with digital inputs, I didn't replace the analog input section. Just to keep my (and your) options open and also to have an analog input channel for measurements, I put a flying header on the B- and C- analog inputs (**Photos 1** and **2**). If you want to use the DCX with a high-quality analog source such as a turntable, I recommend a Behringer SRC2496, which is a flexible unit that does all kinds and combinations of A/D and D/A and sample rate conversion and costs less than a DCX.

Currently, I use a WBT RCA jack in one of the unused XLR input mounting holes to connect the digital input (pin X13(1) to RCA center pin, X13(2) to RCA ground pin; see **Fig. 6**). As mentioned before, you also lose about 10dB of excess gain, so this is a good thing for home use.

The next step up is a new digital input PCB to improve cable matching and jitter performance. I am not sure, though, that it can be done without hardware changes to the DSP board, which I would like to avoid. But stay tuned.

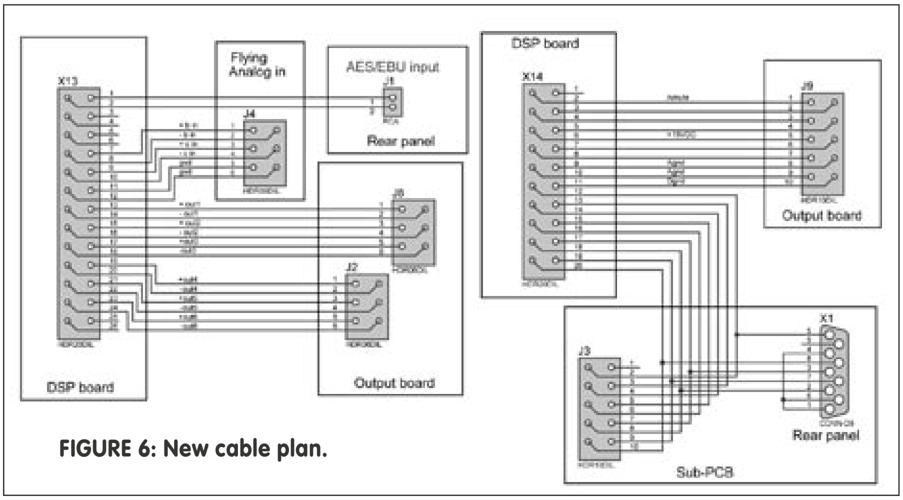

FIGURE 6: New cable plan.

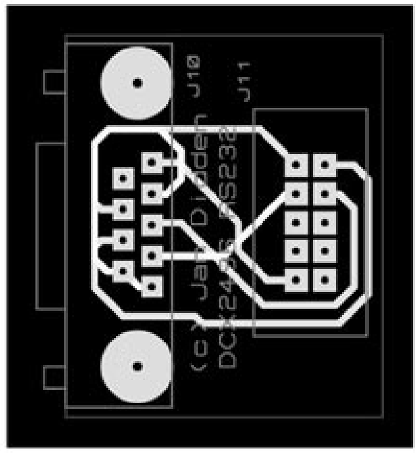

FIGURE 7: Serial port sub-PCB.

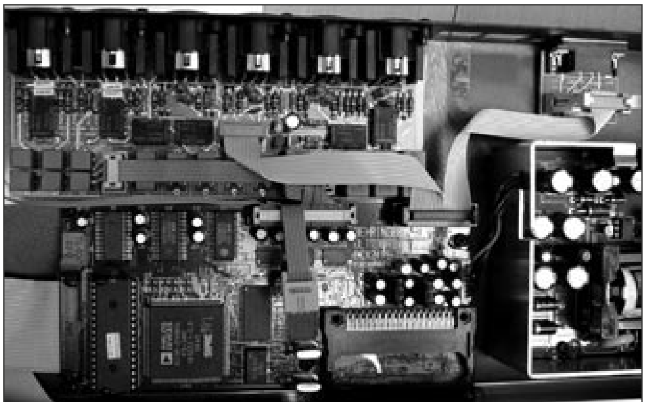

PHOTO 1: Cable layout with new board installed.

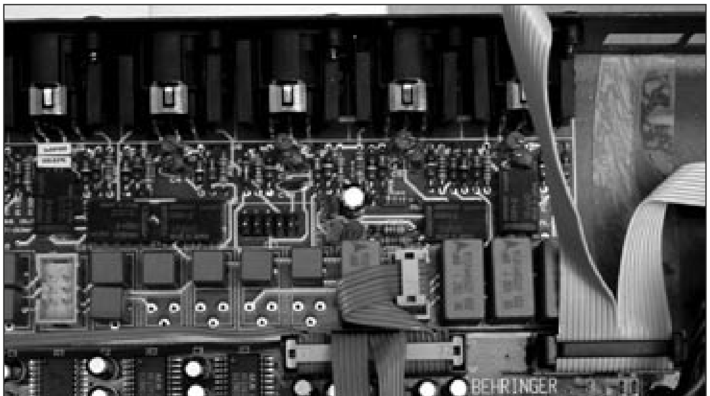

PHOTO 2: Cable layout detail.

WRAPPING IT UP

This is a relatively simple project with a great reward. The mod is fully reversible, and not having to touch the hardware minimizes the risk that you damage anything on the DSP, which would make the unit worthless. Best of all, the sound definitely improved after removing all those op amps. The levels are better matched to a home hifi system as well.

You have several options to do this mod. If you have power amps or a preamp that is AC coupled (has an input coupling cap), you can delete the coupling caps on this board (C1... C12), saving considerable expense. In that case, make sure you jumper each capacitor. There is, however, one caveat to this: depending on the exact pre/power amp input circuit, most probably the mute will no longer function correctly when you use DC coupling. In that case, you might as well delete the mute option, saving the expenses for the six relays and associated components (Q1 . . . 3, D1, R26, R40, C13, C26). If you do delete the mute option, make sure you always switch on the DCX before the power amps and switch it off after the power amps, or, as I do, just leave it on always. (You could, of course, delete the mute option anyway, independent of whether you use AC or DC coupling). Each of the deletions mentioned probably saves some 30% of the cost, depending where you buy, and has, of course, no impact on the final sound quality. *aX*

RESOURCES

You can obtain the PCBs from Roger Pilgham Audio (www.roger-pilgham.nl). And the first one who identifies who Roger Pilgham was gets one set for free! The resistors and PCB headers and connectors are quite standard. The XLR connectors are critical for a correct fit on the replacement board as well as in the chassis, so I recommend using the types indicated in the text. The capacitors are a tight fit, but you may have adequate replacement types. If not, use the indicated Farnell part numbers (www.farnellinone.com).

PHOTO 3: New cable set.