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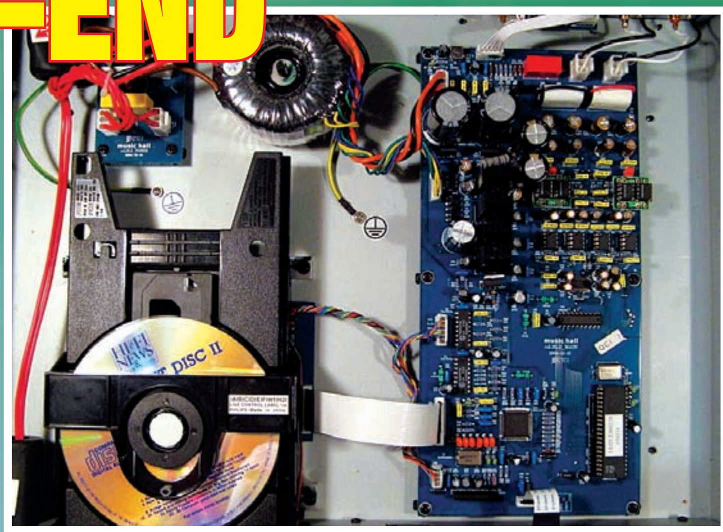

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By Kit Ryan

Modding the Music Hall CD25.2 CD Player

If you're looking for a project with endless possibilities and a big payoff in sound, modifying the Music Hall CD25.2 CD Player may be just the ticket.

PHOTO 1: Exterior view of CD25.2

Why people buy things has so much to do with personal taste that it's hard to justify a particular purchase on purely logical or technical grounds. Suffice it to say that I needed to replace my aging Rotel 855 player (which had been heavily modified already) with a more reliable one (the CD drawer wouldn't open automatically anymore). I considered spending thousands for a high-end player (if I had the money) or buying a less expensive player and trying to make it high end—thus the Music Hall CD25.2 (**Photo 1**).

It didn't hurt that I had once met Roy Hall and knew his taste for "musical" audio equipment. At $600 the player came with good credentials[1,2], but certainly not the ratings of expensive solid-state players, nor the tube circuitry in vogue with audiophiles. For additional reviews of its features and opinions, you can search the web. Buying it was as much a search for a project as it was a search for nirvana in the audio realm.

WHY THIS CD PLAYER?

Right out of the box, it blew away the Rotel, even with my improvements (all-new analog section). It presented a solid, dynamic sound but with a forward tilt and limited dimensionality. Still, it was a major step up. However, the sound quality of some more expensive players I had also just auditioned was ringing in my ears, so I opened up the case and looked inside to see what could be done.

PHOTO 2: Inside the CD25.2 with modified board.

Two notes first:

1. Making the following modifications could void your warranty.
2. The ideas here could very well be applicable to other CD players because the analog architectures after the D/A converter may be similar. Also, be extremely careful with live voltages with the cover off—you could receive a nasty shock or you could accidentally damage the circuitry.

UNDER THE COVERS

All CD25.2 audio components are mounted on a very nicely laid out board (**Photo 2**). Only two surface-mount devices are used for the transport control and D/A converter. Every IC has nearby decoupling electrolytics, each of which is also bypassed with a 0.1µF Mylar film capacitor.

The bottom side of what looks to be a four-layer board is almost a solid ground plane, well divided into sections that all terminate in a central ground point. The grounding is further enhanced with about a 3mm wide ground trace heaped with solder along its entire length, forming a semicircular mound of solder (**Photo 3**). Even though solder is only about 10% as conductive as copper, the net result is a decrease in ground resistance by a factor of about 35 over the copper trace, according to my calculations!

There are no fewer than seven regulated power supplies onboard, fed by a 50VA toroidal transformer. Even the bridge rectifier diodes are bypassed with the same Mylar caps as on the analog ±15V supplies. The audio outputs include a back-to-back pair of large electrolytics bypassed with a "For Audio" 1µF capacitor (never heard of the brand, but it's a big polypropylene cap). All-in-all, well done for this price range!

MODS-R-US

The CD25.2 seems to have been "designed to be modified." All components except two ICs are through-hole devices, all wiring to the motherboard is via removable connectors (making them easy to remove for soldering), all six op amps (two of which are duals) come already socketed for easy removal, and the motherboard is well marked with component IDs and values. If you Google modifications to this player, you will get several hits from highly capable companies who offer many upgrades[3,4]. One nice feature of these offers is that the original manufacturer's warranty is still honored. But I looked at those possibilities and said to myself, "Great, but what fun is that?" So here we are with a whole lot of possibilities to explore.

I needed to pare down the list of potential mods to something manageable. My criterion was primarily that I had to be able to "undo" and "redo" the mod quickly so that I could listen to the effects while my auditory memory was still relatively fresh. This necessitated putting sockets on many capacitors as well as on the digital voltage regulator.

An interesting sidelight to modifications is the answer to the question, "If these changes are so good, why didn't the manufacturer do them at the factory?" It turns out that the answer is simple economics. The typical markup from factory to buyer is about four times on parts and labor. This covers factory and distributor or retailer overheads, sales expenses, and so on. This means adding \$25 in parts increases the list price by about \$100. Whoa!

Say you replace eight op amps that cost \$2 each with OPA627Bs which cost \$20 each in quantity. \$20 × 8 × 4 = \$640, bringing the price of the CD25.2 from \$600 to \$1240. Well, you may say, it now sounds like a \$1200 player. Maybe so, but that's not the price point Music Hall was aiming for. By comparison, a modification house will charge only about two times on the parts and labor cost—perhaps \$300. Of course, doing it yourself can be even less, so stay tuned!

MEASUREMENTS (OR NOT)

One objective of this project was to try to relate sound improvements to improvements in known audio measures, such as frequency response and distortion. Alas, to my great disappointment, this was not to be because, with rare exceptions, the normal measurements were virtually unchanged throughout the modification process. Even using an Audio Precision One system, the differences between the stock CD player and the fully modified one are minuscule, as seen in the following series of figures for frequency response (**Fig. 1**—dark blue is stock), total harmonic distortion (THD) (**Fig. 2**—dark blue is stock), intermodulation distortion (**Fig. 3**—dark blue is stock), using the *HiFi News* test CD[5].

These results were both a great disappointment and a puzzlement, because I thought the audible differences were great. Perhaps the issue was that the technical performance of the CD player was already very near the 16-bit threshold limits to begin with, rendering measurement beyond my instrumentation's capacity. However, I believe there must be some measurements that will show the effects but clearly not the traditional ones. Figuring this out sounds like a future project.

For listening evaluation I originally chose at least a half dozen different types

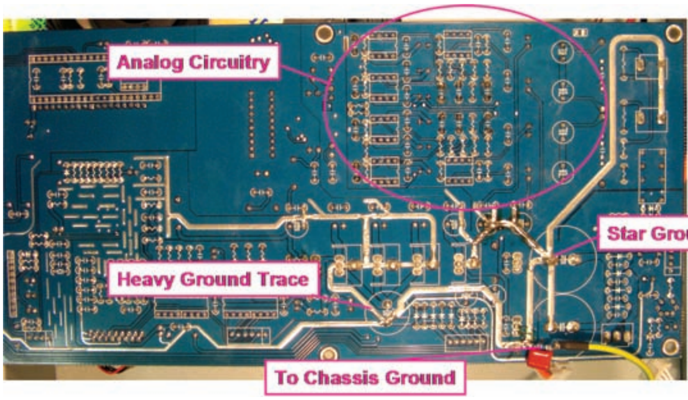


PHOTO 3: Bottom of main circuit board showing ground trace.

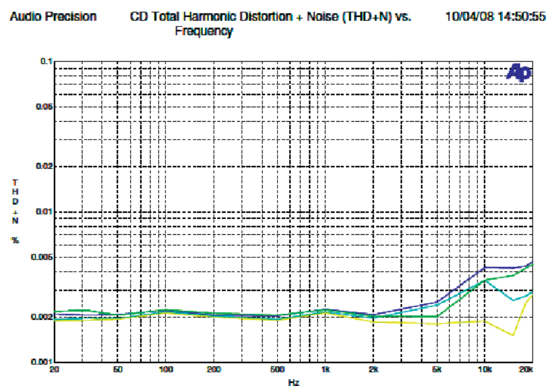


FIGURE 2: THD vs. frequency, stock and modified.

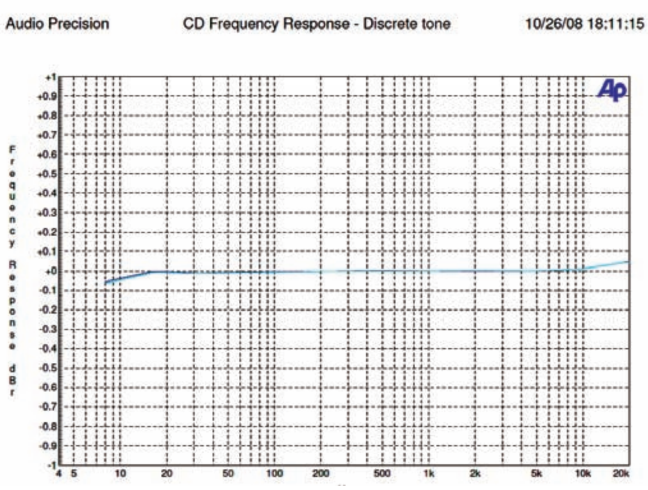


FIGURE 1: Frequency response.

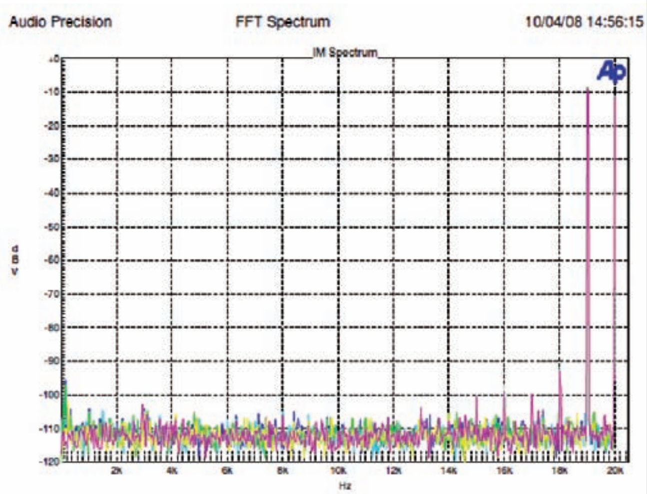


FIGURE 3: IM distortion, stock and modified.

of music, from folk to jazz to rock to classical, with and without voice. As I became more familiar with each piece and the effects of each mod, it was possible to detect the differences with just one or two CDs because the effect was similar on all the music. In fact, after a while I could hear the "before and after difference" in less than 10 seconds with a given track. That doesn't mean it is easy to describe. There is no way I can fill up five pages describing the nuances of each change as might be done in the audiophile magazines. I have tried to use the words "air," "soundstage," and so on as best as I can here, but you'll need to decide for yourself what you like.

Just so you know, my personal leaning is toward what's often called "musicality"—does it have good rhythm? Clean, clear, solid bass? Dynamics? My whole system has been solid-state for 20 years. That said, I have been on a quest to improve the air and soundstage depth of my system without sacrificing musicality, and I was reasonably successful with this project, so don't give up hope, tube fans!

THE MUSIC

I used two cuts for air and transient response:

1. Joni Mitchell's *Urge for Going*, which seems to be available only on her Hits album, Reprise #9-46326-2. The two guitars which start the piece show off plucking of strings, soundstage separation, and airiness very well. Of course, so does her voice. It's hard to believe this was made from a 40-year old analog recording.
2. The beginning of Handel's Harp concerto Opus 4 No 6 with Marisa Robles at the Harp, London 452585-2. Both the strings in the orchestra and the solo harp use plucked notes, making it easy to tell whether a mod has added clarity or mud to the mix. It's a delightful piece.

For socko (deep bass and kick drum togetherness)—Gorillaz' *Windmill* cut on their Demon Days album will almost shake out your fillings, but reveals much about how the bass is affected.

THE MODS

The sequence of making changes and testing was a quandary. The best approach seemed to be to work from the output in—that is, make changes from the "end" of the component chain inward, thereby unblocking any upstream improvements that would otherwise be constricted.

To make the changes easy, I first installed a number of sockets for the components that would be changed often. This required disconnecting the board and turning it over, using solder wick on the desired components, removing them, and reinstalling pin sockets. The board is ROHS compliant and the solder is much more difficult to remove than the traditional lead-based type. I used silver-bearing lead solder to install the pin sockets without difficulty. **Photos 4** and **5** show the board with stock components and with the pin sockets installed. Remember that the board comes with the six 8-pin DIP sockets for op amps installed.

OUTPUT CAPS

The schematic for the entire left channel of the analog section is shown in **Fig. 4**. By the way, all schematics here were done by reverse-engineering the circuit board and may not be absolutely accurate. Also, the component numbers do not necessarily correspond to those on the circuit board. Working from the outputs of the CD player inward, the first thing you find is a sizable output capacitor arrangement.

The outputs are normally shorted to ground by the relay until a CD track is actually playing. The 4.7Ω resistor allows for the output op amps to have some offset without excessive output current when shorted. I measured 3.5mV and 4.5mV offset on the left and right channels, respectively. You can only measure this by playing a completely silent CD track so the relays will open but no audio signal will disturb the meter readings.

I must admit that I kind of cheated on this mod. Instead of trying a whole bunch of different capacitors, I followed the dictum that "the best capacitor is no capacitor." **Figure 5** shows where they were bypassed with a silver wire. (Note that if you wire the output of the op amps directly to the output RCA jacks, the 4.7Ω resistor is also bypassed and the op amps are shorted directly to ground by the relays.)

Bypassing the output caps causes a very (very) slight "click" every time you start playing a new CD, because the input to your preamp goes from ground to a few millivolts DC offset. With the op amp substitutions noted later in this article, it is possible to install an offset adjusting potentiometer and bring the DC offset down to zero, but I didn't find this necessary.

Eliminating the output capacitors was a clear improvement overall. It was as though a veil had been lifted from all the music. For the first time I noticed that the violas in the orchestra behind the harpist on the Handel piece were actually plucking their strings instead of bowing them. A little air was added to all instruments, too. It's not a huge difference, but definitely worth doing.

OP AMPS

Each channel of the analog section of the CD25.2 consists of two individual current to voltage (I/V) converter OPA604s op amps followed by a dual OPA2134 output op amp. The schematic is almost an exact copy of that recommended in the Burr-Brown datasheet for the PCM1738 D/A converter, with the exception that an extra output op amp has been added at the end as shown in **Fig. 4**.

Why a dual op amp was chosen for the critical output section is a mystery. The companies that do mods generally replace this with two single OPA627B op amps using an adapter, so I started looking for the parts but immediately encountered bad news.

No one had any stock on the 8-pin plastic DIP op amps, which is the only way Texas Instruments (who now owns Burr-Brown) makes the coveted "B" model, and wouldn't have any for at least six months. Mouser's site said they would get 85 in December. Just 85?!? Aren't op amps supposed to be sold in lots of 1,000? Maybe the $27 price tag suppresses sales.

Judging from the web forums, this is a chronic problem. What to do? Well, eBay to the rescue—sort of. I ordered two batches of OPA627Bs from Hong Kong. As it turns out, I think both sets are counterfeit. They were too cheap, and one set had terrible characteristics

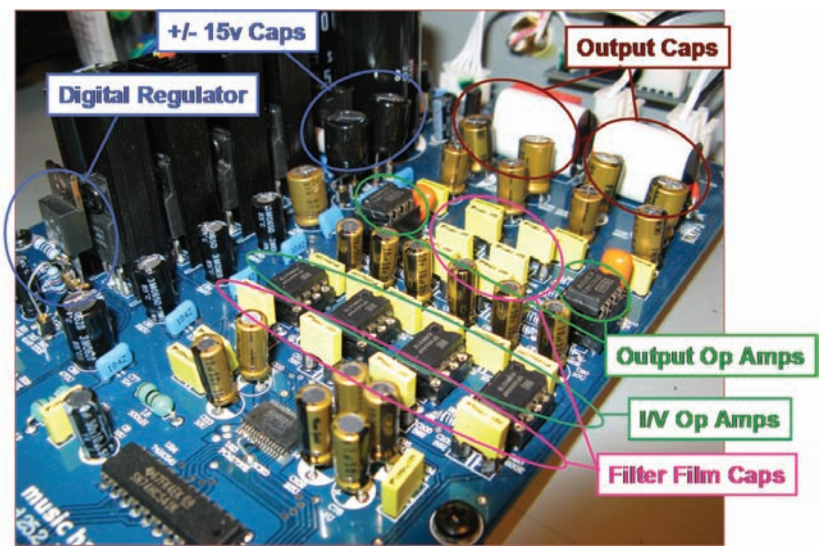


PHOTO 4: Main board with stock components.

PHOTO 5: Main board with pin sockets highlighted.

for distortion and noise. However, the other set did measure well and sounded great, so I used those but that was only enough for the four output op amps. The markings on the "B" type in **Photo 6** may help you to get at least something decent in the near term. I also purchased the 2-to-1 adapters on eBay, which were fine.

An alternative approach was required for trying OPA627s at the I/V position. I used genuine "A" variety op amps, available from Digi-Key and others, which can be had for around $18. Unfortunately, these models are only made

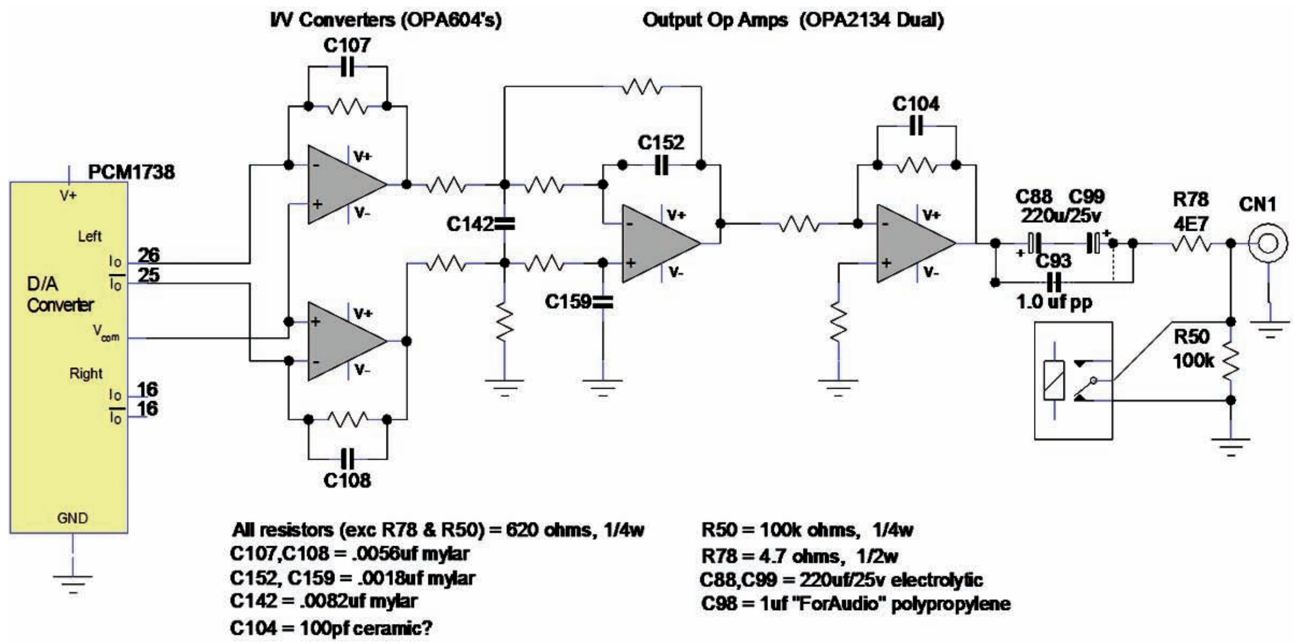


FIGURE 4: Analog section (left channel) schematic.

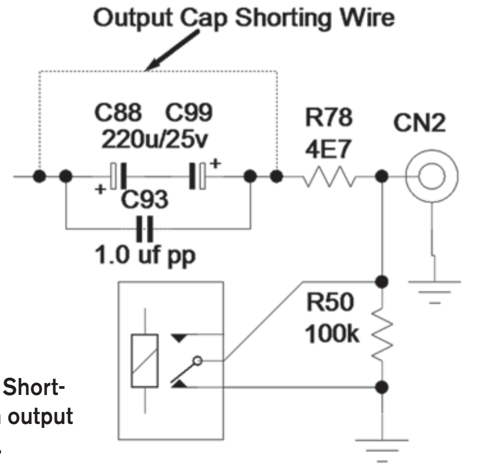


FIGURE 5: Shorting wire on output capacitors.

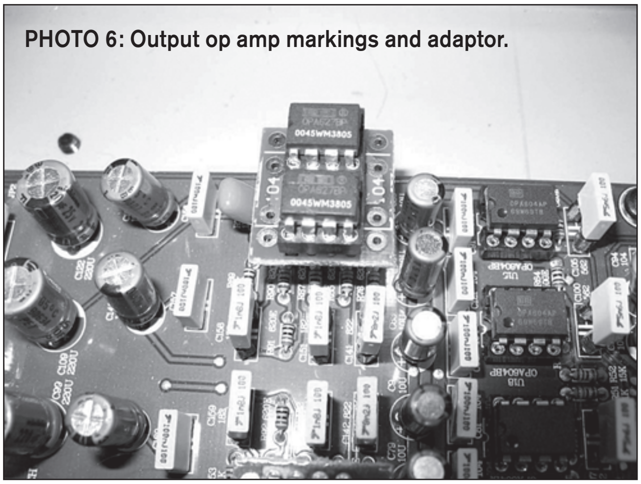


PHOTO 6: Output op amp markings and adaptor.

in SOIC surface-mount packages! Go figure. I found some very nice SOIC-to-DIP adapters with the right kind of pins on them to fit into the PC board's sockets and soldered away. **Photo 7** shows the various op amp configurations with the dual OPA627B adaptor on the left and two OPA627As soldered to the SOIC-to-DIP adaptors on the right.

There are so many "audio grade" op amps now being touted that I got a couple of others for comparison with the Burr-Brown 627s, including Linear Technologies LT1115 and National Semi's LME 49710. Thankfully, these are readily available in 8-pin DIP packages.

You can make a career out of listening to op amps, but in this case the choices quickly became clear. In going from the stock dual op amp to the pair of OPA627Bs on the outputs, my socks were blown off while listening to the Gorillaz Demon Days cuts. This was big time better. Bass notes especially were much cleaner and sounded an octave lower, too. You felt kick drums in the chest. The soundstage was nice and wide but only somewhat deeper than before. A slight mellowing of Joni Mitchell's voice and the overall sound was welcomed, too. Definitely much more "musical" if not super airy. At least half the battle was won.

During these "op amp wars," the "B"s handily beat the competition in the output position. While all the single op amps were better than the stock dual, the "A"s came in a noticeable second with a slightly more analytical sound and less air.

The LT1115s came in a clear third with even less air, narrowed soundstage, and a more "forward" sound overall. Good dynamics, though, with as solid a bass as the OPA627Bs. National Semi's LME49710s were fourth with a harsher sound. For comparison, the stock dual op amps have a very two-dimensional soundstage and forward sound that was further behind these in my ranking. When I put the OPA627Bs back in again after listening to the others, it was no contest. You could tell the harpist on the Handel concerto was really excited about playing!

With that settled, the I/V op amps were next. To make a long story short, the stock OPA604s had a little more air than the OPA627As I tried (I didn't have any more "B" models available to try). Not much difference really, so I left it stock. Apparently the output op amps make a much bigger difference to the sound.

FILM FILTER CAPS

From **Fig. 4** you can see that five small capacitors are part of the high-frequency filters that help roll off the output above 20kHz (C5-C9, ignoring C4, which is just for stability of the op amp and doesn't affect the audio range). The stock CD player uses Thompson Mylar (polyester) caps in these positions. These seemed ripe for replacement with the often exalted polypropylene and polystyrene caps.

Digi-Key had Panasonic polypropylene caps and Mouser had suitable polystyrene caps (one value had to be made up by paralleling two separate capacitors). In removing the stock capacitors, I was careful to safeguard their short leads for reuse in the pin sockets.

While the polypropylene caps could just have their leads cut to fit and work well in the sockets, the small space available on the board meant squeezing the physically larger polystyrene caps in vertically. These caps also come with incredibly thin leads, so I soldered on thicker pieces of leads cut from spare

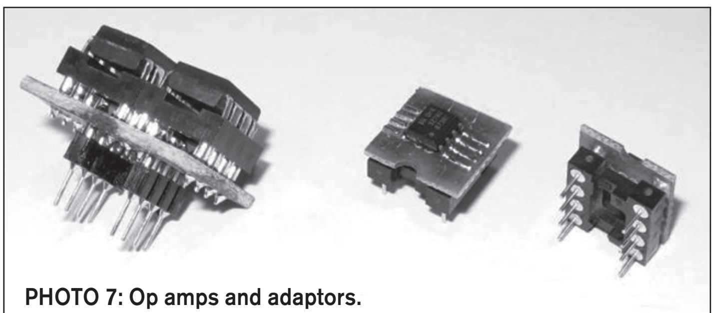


PHOTO 7: Op amps and adaptors.

PHOTO 8: Film filter capacitors.

¼W resistors to get them to fit snugly into the sockets. **Photo 8** shows these different cap types (from left to right: stock Mylar, polypropylene, and polystyrene).

I spent much more time changing these capacitors than with any other mod. The effect on the sound was enormous and confusing at the same time. Both polypropylene and polystyrene added large amounts of soundstage depth and air to every piece of music. This also has a sweetening effect. Some of my notes say "lush," "smooth," "mellow," "sheen," and so on. That's the good news.

I'm afraid there's bad news, too. They both diminished the dynamics of virtually everything. Joni Mitchell sounded as though she was all of a sudden playing a nylon string guitar; Gorillaz' kick drum had a "hangover" on each beat, plus the treble was dulled; the orchestra on Handel's concerto was waaaaay back there (the big soundstage), and the harp sounded as though it had the sustain pedal stuck in the "on" position. Oh boy, not an easy choice.

For those of you who like the warm tube sound, it's a no brainer. Definitely go with the polystyrenes—they're noticeably more musical than the polypropylenes. Personally I wish I could easily install a switch depending on the type of music. But for now, the stock Mylar caps are staying in my unit. It's just too big a penalty in transient response for me. You must come to your own conclusions—I'm exhausted with this one.

ANALOG REGULATOR CAPS

The whole analog section of the player is powered by two standard three-terminal voltage regulators at ±15V. Music Hall did a nice job of bypassing the output capacitors on these regulators with 100nF Mylars and, in addition, put more electrolytics and bypass Mylar caps at the power pins of each op amp. A serious attempt at power-supply quality. The analog regulator and distribution circuitry is shown in **Fig. 6**.

The main output electrolytics at the regulators are 220µF/25V Nichicons—Gold with black stripe. I found sources for the much regarded Nichicon Muse and Elna Silmic (silk impregnated) capacitors but only at 50V ratings. Panasonic EEU audio capacitors from Digi-Key were also included with a 35V rating.

All the new caps are physically larger with thicker leads than the stock types. For use in the sockets, the leads had to have thinner lead pieces soldered onto them to allow for swapping, the exact reverse problem of the polystyrene filter caps. The capacitors are compared in **Photo 9** from left to right: stock, Panasonic EEU, Nichicon Muse, and Elna Silmic.

The Panasonics were eliminated right away—my notes just say "bad!" That's two strikes against Panasonic. Not sure whether they're really interested in audio.

There must be something about the silk in the Elnas because they definitely made the sound smoother but at a noticeable loss in dynamics. On the other hand, the Muse caps made everything clearer and more musical. More bass socko, more ringing on the guitars while preserving most of the air that had been added by the op amp switch. Despite the added punch, there was no additional harshness. The overall effect was modest but there. The Muse caps were a keeper.

DIGITAL SUPPLY REGULATOR

The three major digital ICs are fed 3.3V from a single LM317 regulator. Each IC is well isolated and filtered as you can see in **Fig. 7**. One of the mod companies makes a big deal out of changing this regulator, so I thought it worth a look.

The stock circuit consists of a garden-variety LM317 regulator followed by some very thoughtful filtering. Perhaps noise on this regulator could affect the D/A conversion process and may be audible in the way of jitter on the main 16.394MHz oscillator. Digging into IC regulators netted a crop of lower noise types such as the LT3080 that could be adapted for a swap. Because they are not pin-for-pin replacements for the stock units, I had to remove two resistors and a diode from the PC board and solder two of them onto the stock regulator pins.

Using two pin sockets for the voltage setting resistor, it is then possible to use the LM317 with its 330Ω resistor, then pull them both out and substitute the LT3080 with its 330k resistor. **Photo 10** shows how the two regulators were

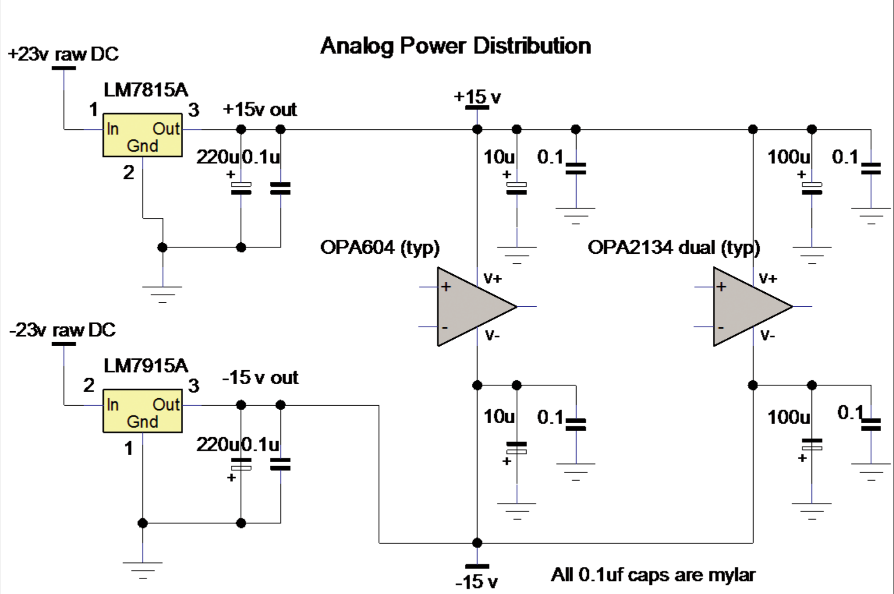


FIGURE 6: Analog regulator and distributor schematic.

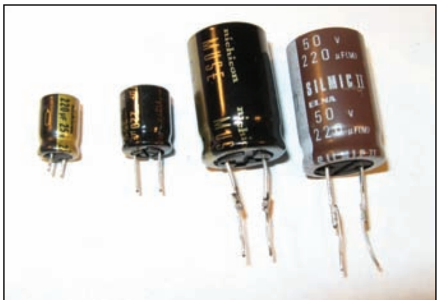


PHOTO 9: Analog regulator caps.

configured for rapid change-out with the stock unit on the right. **Photo 11** shows how the board was modified with the removed components and added pin sockets. The setting resistor goes between the two pin sockets where R25 used to be.

I made a surrogate measurement on jitter, namely phase noise on the master clock as it is seen by the D/A converter. It has a clock output pin that will drive the 50Ω input to my spectrum analyzer, and I tested it with the two different regulators (**Fig. 8**). The A trace is the Tektronix 495P residual phase noise, B is with the stock regulator, while the C is with the low-noise LT3080 regulator. It looks as though there's about a 2dB reduction in phase noise for most of the audio region with the LT3080.

A couple of back and forths on these parts and the verdict was obvious—the difference was audible and the lower noise regulator was better, although this had nothing to do with audible "noise" (e.g., hiss). Bass notes seemed even lower now, while the dynamics were maintained and the soundstage was widened and deepened somewhat. Overall, it brought good benefits with no penalties. I don't know whether a hand-made, discrete component regulator would do even more, but this was sure a good $4

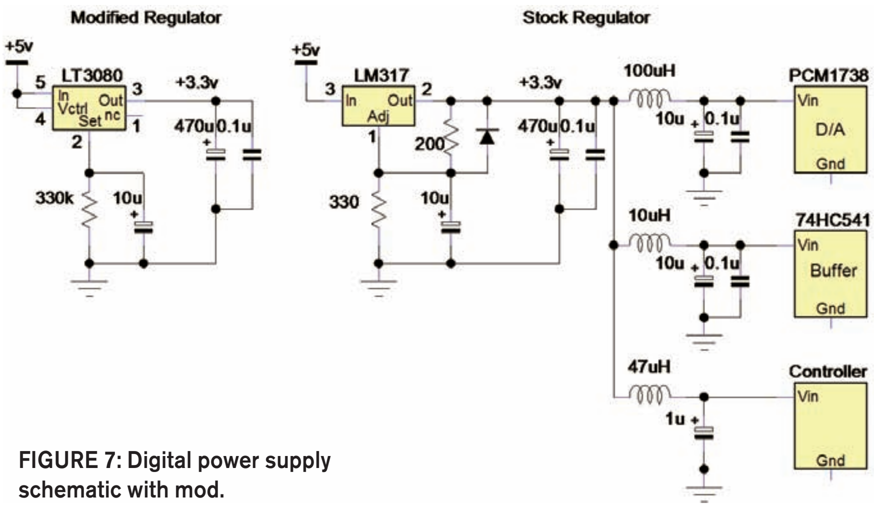


FIGURE 7: Digital power supply schematic with mod.

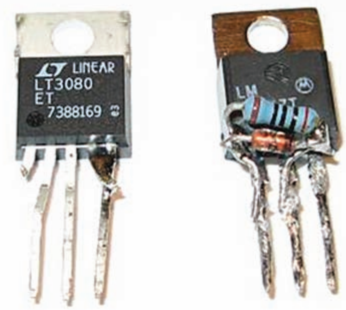


PHOTO 10: Digital regulator physical configuration.

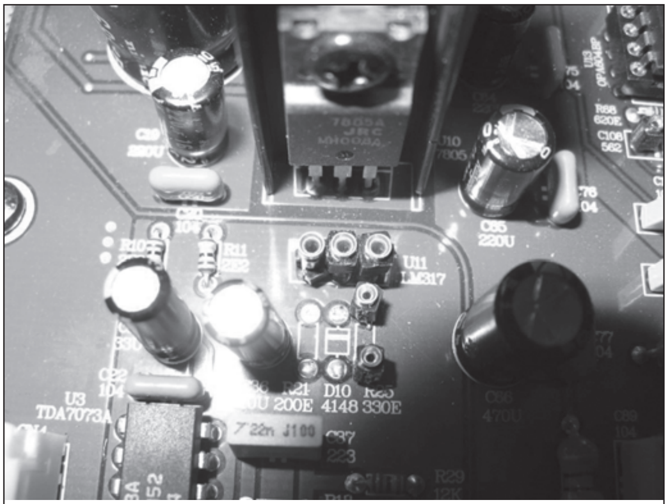


PHOTO 11: Digital regulator modifications.

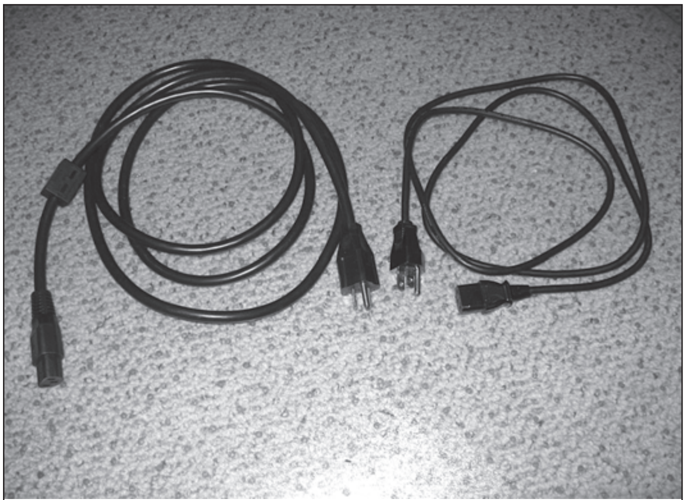


PHOTO 12: Power cords—14 gauge vs. 18 gauge stock.

investment.

POWER CORD

I don't know why I even thought of changing the power cord. Every engineer knows that it can't have any effect on the sound after power goes through a transformer, rectifier, filter caps, regulated power supply, more filter caps, and then to an op amp that has 100dB of PSRR (power supply rejection ratio). Let's get real here.

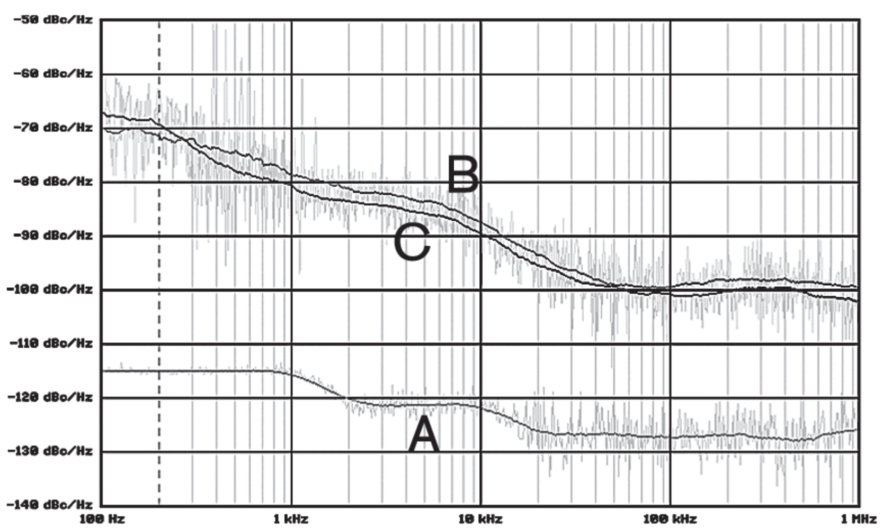
However, I ran across some cords on eBay for just $10 each and tried one out. These were 14g and much thicker than stock. They came with clamp-on RF filters which are easily removed (which I did just to be sure they weren't interfering with the sound). The impedance at 120Hz measured 0.081Ω for an 8′ length compared with 0.196Ω for the 6′ stock unit. Based on that ratio, the stock cord was probably 18 gauge. See the size difference in **Photo 12**.

All I can say is that when I changed power cords, I was shocked (no pun intended). Everything was better—air, soundstage, dynamics, and so on. Typical CD harshness was reduced, too. Musicality was unaffected. Is this really possible? Must be.

But the story was more complex than that, actually, because at first I compared the stock cord plugged into the power strip I have used for years (computer type with surge protection) with the new cord plugged directly into the wall socket. The wall socket in my system has a dedicated 20A circuit breaker with 12 gauge wire running straight to the hi-fi system. When I plugged the stock cord directly into the wall, it was better, too, but not nearly like the heavy gauge cord.

Two lessons to take away from this: first, if you have a cheesy power strip like I did, remove it from your system immediately. Everything will sound better. Second, buy a heavy power cord because it makes a difference no matter what the experts and instruments may say. For $10 how can you miss?

As an aside, this change was so dramatic to me that I've begun rewiring all my components with heavy cords and a direct connection to a heavy-duty, non-protected power strip. It seems that every one of these changes brings an improvement to the overall system.



Trace	Carrier Hz	Carrier dBm	dBc/Hz at 200 Hz	RF Atten dB	Instrument
Test Phase Noise	16 936 000	-20.00	-126.5	0	TEK 495P
Test Phase Noise	16 936 000	-20.00	-69.4	0	TEK 495P
Test Phase Noise	16 936 000	-20.00	-72.0	0	TEK 495P
Test Phase Noise	16 936 000	-20.00	-97.0	0	TEK 495P
Test Phase Noise	16 936 000	-20.00	-96.2	0	TEK 495P
Test Phase Noise	16 936 000	-20.00	-115.0	0	TEK 495P

FIGURE 8: Digital phase noise on master clock.

CASE DAMPING

To be efficient and not have long-time lags between listening sessions for these mods, I removed the top of the player case throughout these changes. I put two layers of damping material available from Parts Express on the inside of the top, which clearly deadened the case ringing, but I really didn't notice any difference in the sound with the case on or off, damped or not. Still, it's cheap insurance and gives the unit a more solid, expensive feel. By the way, it was difficult to fit two layers of the material in because there's not much clearance between the metal top and the mechanical workings of the CD tray. You may just want to do one layer.

SUMMARY

All the results of my experimentation are charted in **Table 1**, which indicates the direction and extent to which a change was considered to affect the sound. A big asterisk and footnote goes to the op amp filter capacitor option because this is such a big trade between soundstage and dynamics. The chart also indicates the cost of each option. Amazing improvements can be made for just a few dollars, the exception being the real OPA627Bs if you can find them.

There are literally thousands of combinations of these mods plus many others that could be tried. For example, I did not replace the power supply diodes with soft recovery types, put special feet on the case, try super-duper resistors, or try different bypass caps on each op amp. When I heard an improvement with a change I left it in and went on, but you can experiment with other mixes to your heart's content. I am quite happy with the sound now but am still looking for more depth to the soundstage without compromising the musicality. Happy modding! *aX*

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1. Audio Advisor review online: www.audioadvisor.com/prodinfo.asp?number=MHCD25.2.

2. Elusive Disc review online: www.elusivedisc.com/prodinfo.asp?number=MHCD252.

3. Reference Audio Mods at www.referenceaudiomods.com

4. Underwood HiFi at www.underwoodhifi.com

5. *HiFi News & Record Review* Test Disc II, HFN 015, 1989.

PARTS SOURCES

Digi-Key
OPA627A op amp (SMT package only)
LT1115 op amp
LME49710 op amp
Panasonic polypropylene and electrolytic caps
LT3080 adjustable low noise regulator

Handmade Electronics,
(www.hndme.com) (handmade@hndme.com)
Elna Silmic electrolytics
Nichicon Muse electrolytics
Silver wire (20g)

Mouser
polystyrene caps

Speakersformusic, (eBay vendor)
(speakersformusic@windstream.ne)
14g power cords

Logical Systems (www.logicalsys.com)
(shopadmin@logicalsys.com)
SOIC to DIP adapter, 8-pin (#PA-SOD3SM18-08)

Tube_buyer (eBay vendor) (raychow830@yahoo.com.hk)
Dual to mono DIP adapter with OPA627B op amps
(these were the good-sounding ones)

Parts Express
Damping material (#268-015)

Component / Effect	Output Op Amps	I/V Op Amps	Output Cap	Bandpass caps	15v Supply caps	+3.3V Digital Regulator	Power Cord
Better	OPA627B						
	OPA627A						14 gauge
	LT1115						
			bypass		Muse		
	LME49710					LT3080	
Stock	OPA2134	OPA604	ForAudio	Mylar *	Nichicon	LM317	18 gauge
				Polystyrene			
		OPA627A			Silmic		
				Polypropylene			
Worse							
Cost for best upgrade	$110	$0	$1	$0	$2	$4	$10

* Capacitor effect is huge and a trade-off between soundstage and musicality

Table 1

By Ron Tipton

DIY Microphone Calibration

This hands-on tutorial shows you how to inexpensively calibrate a microphone.

PHOTO 1: Half-inch electrostatic actuator.

PHOTO 2: Microphone Calibration Workstation, model 9350C from the Modal Shop, Inc., www.modalshop.com. When the lid is closed, some sound isolation is provided for the microphone calibration.

Microphone calibration has been the subject of many articles and technical papers, some of which I've included in the References and Bibliography sections. I won't cover so-called "primary" calibration because it is too expensive and too time consuming for those of us who just want to have a reasonably accurate measurement microphone. I will discuss two methods of "secondary" calibration: the electrostatic actuator and microphone comparison.

THE ELECTROSTATIC ACTUATOR

This is a non-acoustical method, and although the equipment is a bit expensive[1], it can yield excellent results. Because of the way this method has evolved, it is applicable to only the class of measurement microphones that come in 1/4, ½, and 1″ sizes—that is, the outside diameter of the microphone barrel. Bruel and Kjaer[2] measurement microphones are well known, but my favorite is the inexpensive model ECM-8000 made by Behringer[3], which I discuss later in this article.

To use this method, remove the microphone cartridge protection grille and replace it with an electrostatic actuator. They come in 1 and ½″ sizes, and a ½ to ¼″ adapter is available for ¼″ microphones. **Photo 1** shows a ½″ actuator. Place the microphone and actuator in an enclosure to isolate the measurement from external noise sources (**Photo 2**). In the measurement setup (**Fig. 1**), the swept sine wave input can be from a computer sound card and the spectrum analyzer can also be software based.

The actuator driver is probably beyond the realm of the home-constructor because it amplifies the input signal to about 25V RMS and then imposes it on 800V DC to drive the actuator. The actuator does not produce a sound wave (a non-acoustical method) but rather electrostatically couples the diaphragm in the actuator to the diaphragm in the microphone. Energy is transferred through the electrostatic field and the microphone "thinks" it's in a sound field.

My primary reason for introducing the actuator is because this is how measurement microphone manufacturers calibrate the product we will use as the reference microphone in the comparison method. It would be useful to be able to periodically check the calibration of the

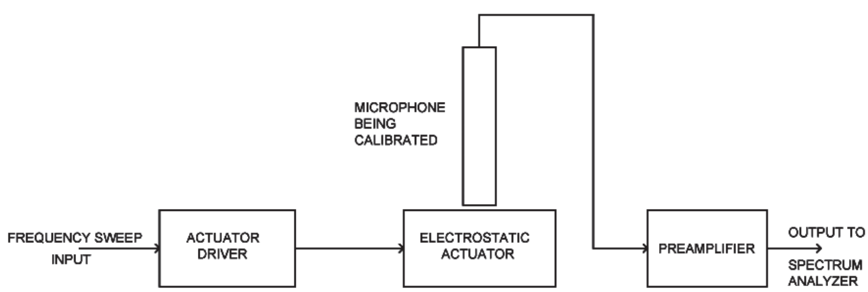

FIGURE 1: Electrostatic actuator method block diagram. In practice, the actuator and microphone would be enclosed to reduce ambient noise pickup.

reference microphone, but the actuator and driver are a bit pricey. It turns out there is a less expensive way to verify the calibration.

THE COMPARISON METHOD

Briefly stated, this is done by comparing the microphone to be calibrated with a reference microphone. But as the saying goes: "The devil is in the details."

One reference I found[4] made a convincing case for needing a sound source (loudspeaker) with a nearly flat frequency response. Hence, I spent quite a bit of time designing and building the coaxial speaker shown in **Photo 3**. The response is fairly flat but the microphone placement is so critical that it was impossible for me to produce repeatable results. (Unless you are doing these measurements out of doors or in an anechoic chamber, you are always seeing the combined response of the speaker and the room.)

At this point, I wasn't sure where to go with this project. I finally decided to just try a speaker and see what happened. I picked one of the four that I had built for the MaxxBass article (*aX* 11/04). It has a Hi-Vi F6 driver in a 0.35 ft^3 sealed box with a measured Q(tc) of 0.71. The frequency response is down about 40dB at the low and high ends of the spectrum compared to the midrange, but it worked well enough. It is essential to make the measurements at a high enough signal level so the signal-to-noise ratio is adequate over the full spectral range, but this is not difficult to do.

Photo 4 shows this speaker and the microphone holder I built. The wooden frame is about 18 inches square, but this is not critical. I did round the speaker-facing edges of the boards to minimize

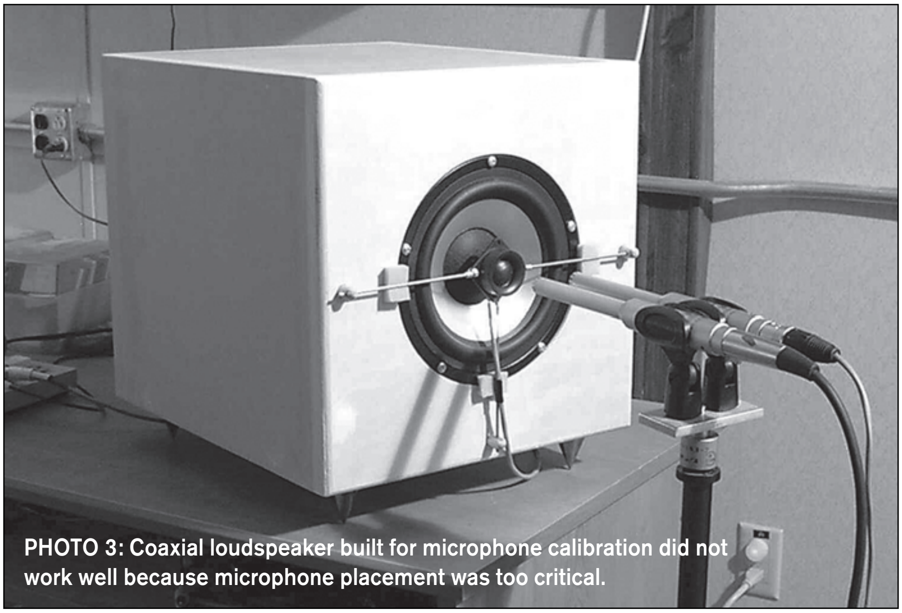

PHOTO 3: Coaxial loudspeaker built for microphone calibration did not work well because microphone placement was too critical.

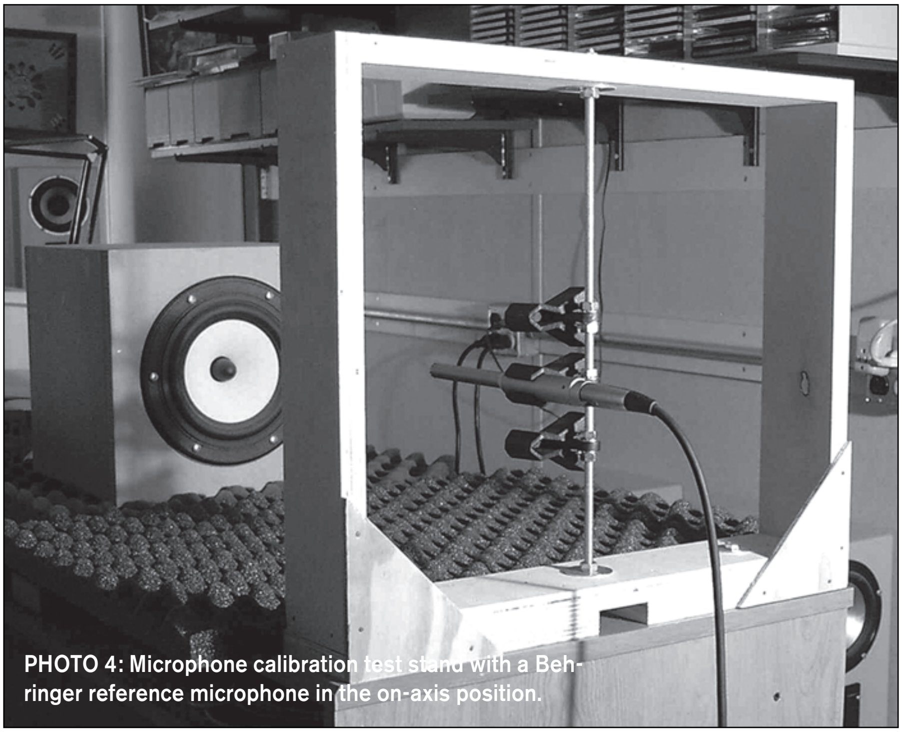

PHOTO 4: Microphone calibration test stand with a Behringer reference microphone in the on-axis position.

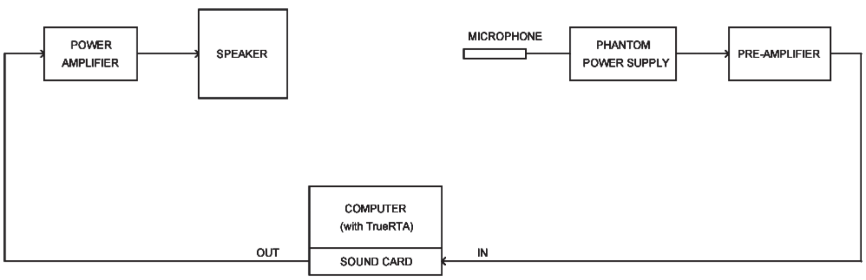

FIGURE 2: Comparison method block diagram. The microphone to be calibrated is substituted for the reference microphone.

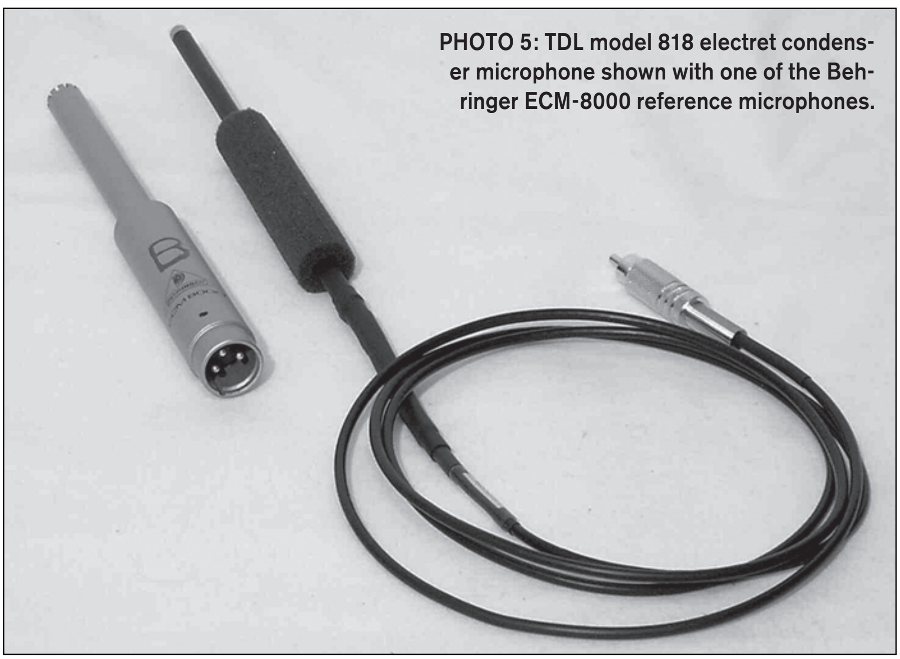

PHOTO 5: TDL model 818 electret condenser microphone shown with one of the Behringer ECM-8000 reference microphones.

reflections. The spring-loaded microphone holders are type MUS-MY200 from American Musical Supply[5] supported on a 5/16″ diameter threaded rod. (I have discovered that this holder is also available from Parts Express[6] as catalog number 242-025). I used three holders to investigate the off-axis response. The best results are attained by substituting the microphones in the on-axis center holder. The speaker cone to microphone spacing is 15 to 18″ with the setup shown in the photo—this distance is not especially critical.

The comparison block diagram is shown in **Fig. 2**. A switch box for the preamplifier inputs and outputs would be helpful, but so far I have just switched connectors. First, I will list the needed equipment and then I will give you the step-by-step details for calibrating the unknown microphone, in this case a TDL® model 818 (**Photo 5**).

1. Reference microphone, Behringer ECM-8000[6]. (It would be helpful to buy three, as I will explain later.) You will notice that the microphone spec sheet indicates +15 to +48V of phantom power is needed, which is true. But at 15V the sensitivity has decreased and the decrease isn't specified. I have measured it at about -2.5dB, but it varies from microphone to microphone.
2. Phantom power supply for the microphone, 48V DC. (I used a TDL model 492 because I had one available.) The model 432 preamplifier supplies the 818 microphone.
3. Low-noise microphone preamplifier with at least 40dB of gain. Even though the microphones are substituted, both channels are needed because the right channel also supplies power to the 818 microphone. (I used a TDL model 432.)
4. Audio power amplifier. Not critical, a few watts up will do. The one I used has a gain of 18dB, which required a *TrueRTA* output setting of -25dB or higher for a good measurement signal-to-noise ratio.
5. Microphone and audio cables. I like the *Dayton Audio* cables from Parts Express[6].
6. *TrueRTA* software[7] and a computer with a sound card. (*TrueRTA* will run under Windows 98SE and up.)

CALIBRATION PROCEDURE

1. Try to keep your measurement environment as quiet as possible. Listen for noises that you are accustomed to hearing and turn them off if possible or wait for them to stop. These noises include air conditioning or furnace blowers, fans, refrigerators, and so on.
2. Turn on all the equipment and launch *TrueRTA*. The combination of the speaker and room responses will probably be rather "ragged" because QuickSweep is performed in 1/24-octave mode.
3. Put the reference microphone in the holder and make sure its cables are connected correctly. Sit quietly in your chair and tap the QuickSweep key. You will hear the "chirp" from the speaker and the frequency response will appear on the screen. Save this response to Memory 2 (Alt-2).
4. Substitute the microphone to be calibrated for the reference and change the cable connections. The model 818 gets its operating voltage (+5V DC) from the right channel of the 432 preamplifier. Sit quietly, hit QuickSweep, and save the response to Memory 3 (Alt-3).
5. You will probably find that the two curves on the screen have an offset between them. For the 818 with its lower sensitivity this was about -3dB. Bring up the *TrueRTA* Utility menu and select Shift. Shift the lower curve up so that it coincides as closely as possible with the upper curve. The Shift works repeatedly, so keep at it until you get a good match in the center portion of the spectrum.

   Bring up the Utility menu again and select Difference. Subtract one input from the other. You may need to try it both ways and keep the one that shows a rolloff (rather than a "roll-up") at the low end of the spectrum. The difference is your calibration curve; save it to Memory 4 (Alt-4).

   Because of the jaggedness of the 1/24-octave curves, I tried 1/3-octave smoothing of the microphone data before differencing. But this didn't seem to work too well because it exaggerated the amplitude differences at the low and high ends of the frequency spectrum. It seems better to smooth the difference curve, and

1/2-octave smoothing seems about optimum. Why is it fair to do this? Because microphones tend to have smooth responses with only gradual roll-ups or rolloffs.

6. Bring up the File menu and select Save. This lets you save the Workbench (all three curves) to your choice of file name and directory. From the View menu select Hide All Memories. Then click on the Memory 4 icon. The difference curve alone will reappear.

   From the File menu select Export Data and save the calibration data as a text (ASCII) file to your choice of file name with a .txt extension. This file can be used in a variety of programs, including *TrueRTA*, as a microphone calibration file. **Figure 3** shows the 1/24-octave microphone data and the difference with 1/2-octave smoothing.

Now you have a relative calibration curve, but is there a way to get an absolute reference value so you could use the microphone for making sound pressure level measurements (SPL)? Yes, there is.

The Behringer ECM-8000 has a nominal (but pretty close) sensitivity of 10mV/Pascal with a 48V phantom supply. Put the ECM-8000 back in the holder and substitute a physical voltmeter for the input to the computer sound card. (You can probably use *TrueRTA* for this, but I prefer a "real" voltmeter.) Using a 1000Hz sine wave output (from *TrueRTA*), adjust the output level and preamplifier gain for a 1V RMS reading on the voltmeter.

Now substitute the newly calibrated microphone and measure its output level, keeping the level and gain controls unchanged. Suppose the voltmeter now reads 680mV RMS. This gives your microphone a sensitivity of 6.8mV/Pascal. Knowing that 1 Pascal equals 94dB SPL lets you use your microphone for SPL measurements.

Earlier I suggested you buy three Behringer microphones. Why? First, they are inexpensive—less than $50 each. But more important, with three you can verify that they are all functioning correctly or if not, which one has gone bad. With just two, you can inter-compare them and if there is a difference all you know is that one of them has "taken a dive." But with three, you inter-compare A with B, A with C, and B with C; and thus find the "bad apple" if one exists. True, this is an expenditure of another $100, but this is much cheaper than buying an electrostatic actuator and accessories for the same purpose.

SOME FINAL WORDS

Useful microphone calibration and verification is well within the means and ability of any audio enthusiast who cares

FIGURE 3: Reference and model 818 microphone data shown with the difference curve. TrueRTA using QuickSweep. Half-octave smoothing was applied to the difference data.

to take it up. The fact is, with careful work, most any mid-range speaker is up to the task. Also, using the Shift and Difference functions built into *TrueRTA* makes easy work of getting the calibration data.

I tried two other speakers with comparable results. One of them, a 3″ TangBand model W3-879S with a response of ±5dB from 100Hz to 20kHz, gave a very smooth difference from 50Hz up but very rough below 50Hz. Because of its small size, its response is down about 60dB at 20Hz. But this could still be a good choice if you are primarily interested in the mid and high frequencies. (Although the W3-879S has been discontinued, there are current TangBand models with similar specs.)

I have also tried using MLS software for microphone calibrations such as *Praxis*[8], *Sample Champion*[9], and *WinMLS*[10]. But I have found that windowing the impulse response to get the frequency response is critical. I have had better results with less work using the QuickSweep feature in *TrueRTA*. However, you may have better results with MLS.

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1. For example, a minimal system from G.R.A.S. Sound & Vibration costs about $3600 and consists of a type 14AA High Voltage Actuator Amplifier, a type AL0010 Calibration Stand, and a type RA0014 ½″ Actuator. You can take a look at the specs at www.gras.dk.

2. Bruel and Kjaer is a Danish company founded in 1942 to develop instruments for acoustic measurements. You can find a wealth of product and technical info on their website: www.bksv.com (Denmark) or www.BKhome.com (USA).

3. Behringer is a German company that makes all kinds of audio instruments. Take a look at their website: www.behringer.com.

4. Case Study: Secondary Microphone Calibration by Substitution Method. www.fouraudio.com.

5. American Musical Supply, 1-800-458-4076, www.americanmusical.com.

6. The Behringer microphone (catalog number 248-625) and Dayton Audio cables are available from Parts Express, 1-800-338-0531, www.parts-express.com.

7. *TrueRTA* software is available from True Audio, www.trueaudio.com.

8. *Praxis Audio Measurement System*, Liberty Instruments Inc., www.libinst.com.

9. *Sample Champion*, www.purebits.com.

10. *WinMLS*, Morset Sound Development, www.winmls.com.

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1. Microphone Comparison. www.purebits.com.

2. Jobling, B., "Electrostatic Actuators—A Role in Calibration. www.iop.org/EJ/article/1751-8121/41/16/164033/a8_16_164033.pdf.

3. Nedzelnitsky, V., "Laboratory Microphone Calibration Methods at the National Institute of Standards and Technology, U.S.A. ts.nist.gov/MeasurementServices/Calibrations/acoustics.cfm. Scroll down to near the bottom of the screen and click on the report name, it's a pdf file.

4. Electrostatic Actuator Power Supply User's Guide, www.bswa-tech.com.

5. Bruel, P. V. and Harry Zaveri, "Of Acoustics and Instruments—Memoirs of a Danish Pioneer." Part 1 was published in the February 2008 issue of *Sound & Vibration* magazine with Part 2 following in the August 2008 issue. Sound & Vibration, PO Box 40416, Bay Village, OH 44140. 1-440-835-0101, www.SandV.com.

T-reg: A High-Voltage Regulator for Tube Amps

Tube amplifiers can benefit from regulated and stabilized power supplies, especially sensitive preamplifier stages, but also (single-ended) power amps. But traditional high-voltage regulators add several extra tubes, complexity, and power losses, while good performance is not easy to obtain. This article describes a novel design that is very simple yet offers excellent performance. Output voltage is fully adjustable with a single resistor without any compromises in performance.

Why is a regulated supply for (tube) circuits a good thing? If you look at a circuit, you see a DC source powering the stages. The implicit assumption is that the DC power source is just that—DC power, with no hum, noise, and ripple from the signal frequency. But that's not the case, of course.

There's ripple and noise from the mains through the rectifier and smoothing circuits. Even if you get fancy with C-L-C and so on circuits, it's never completely clean. And the load current that varies with signal frequency flows through the internal resistance of the supply, and Mr. Ohm showed us that current × resistance causes a signal-related voltage on the supply. Is that bad?

Every amplifier has a property called power supply rejection ratio (PSRR), which is a measure of how much of the ripple and hum on the supply bleeds through to the output. It varies with amplifier topology, of course, but it's always there. Single-ended circuits are especially sensitive to this. So, a regulated supply that minimizes ripple, noise, and hum on the supply line is quite useful for clean, undistorted sound. Especially if it can be much better, smaller, and cheaper than a classic C-L-C circuit.

THE SOLUTION

The idea for this project came to me when I was working on a broken 1970s Lambda 250V lab power supply. After spending a few hours on it, I found the problem: a defective voltage regulator IC, a 14-pin Lambda FBT-00031. Tech support at Lambda kindly provided a single spec sheet for the chip. It took several weeks for someone on the Internet to identify the chip as a re-branded Motorola MC1466L (**Fig. 1**), which was obsolete, of course.

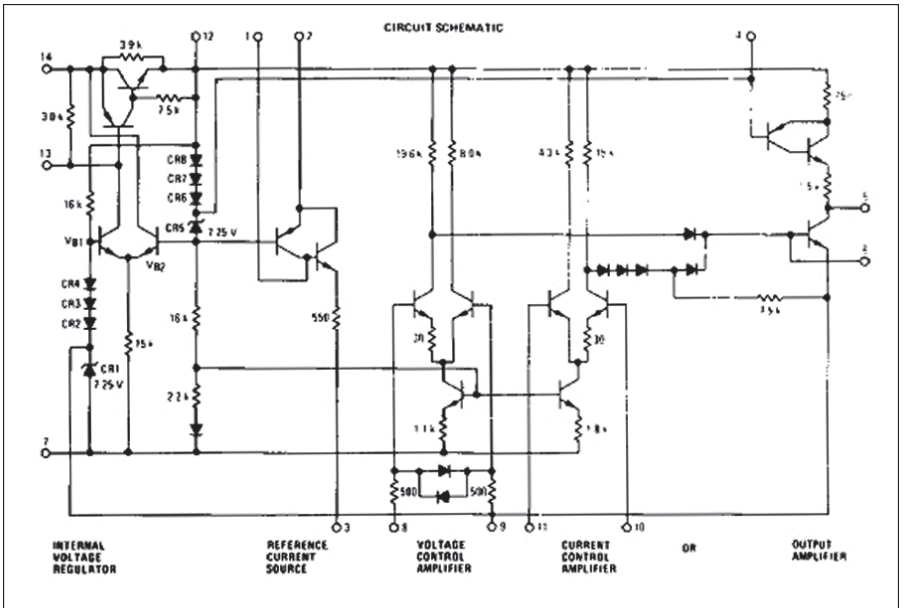

FIGURE 1: MC1466L simplified circuit.

I found two NOS sources: one in the Far East to the tune of $280 each for a minimum order of four, the other in nearby Germany for 6 euros each. I ordered a couple, replaced the chip, and the supply worked perfectly. I found the conceptual approach of this chip, with a floating error amp and reference setting with a single resistor, so intriguing that I decided to see whether I could apply it to a high-voltage tube regulator.

The concept of my circuit is shown in **Fig. 2**. This is a high-voltage regulator, but I'd like to avoid as much as possible relatively rare and expensive high-voltage devices and parts. That's why there is a separate low-voltage supply, floating on top of the high-voltage output. This low-voltage supply, powered from the rectified pass tube heater, runs

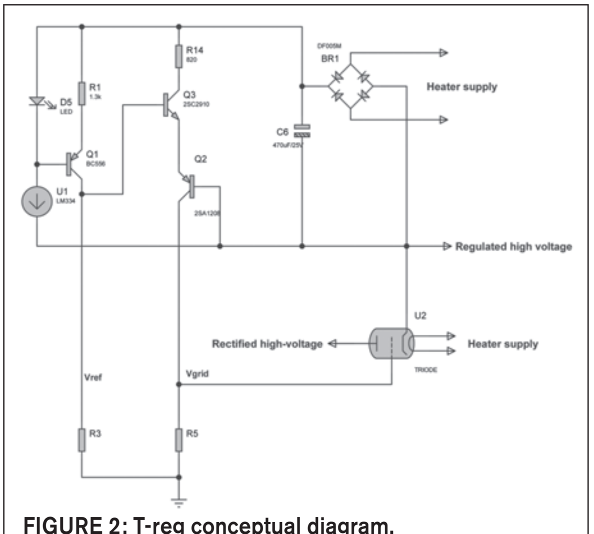

FIGURE 2: T-reg conceptual diagram.

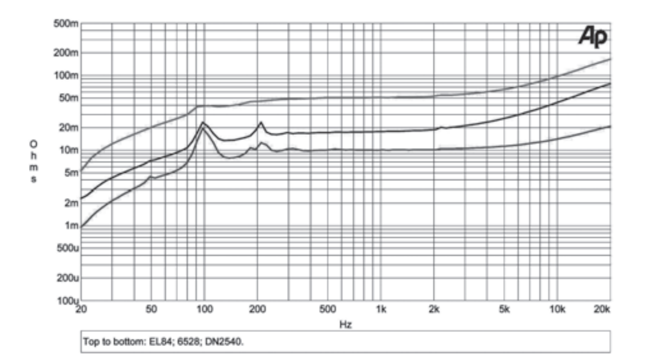

FIGURE 3: T-reg output impedance plots.

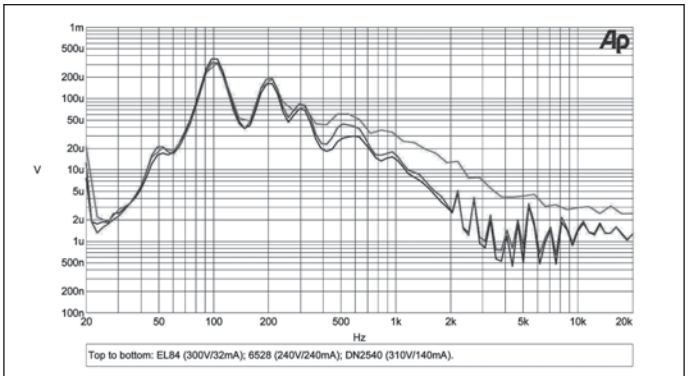

FIGURE 4: T-reg output hum and noise plots.

the reference and control circuits, which can be low-voltage.

This type of regulator consists of the same modules. There is always a reference voltage and an error amplifier that compares the reference voltage to (a sample of) the output voltage. The error amplifier then drives the pass element to control the output.

HOW IT WORKS

In my circuit, the reference voltage comes from resistor R3, driven from the floating constant-current source Q1. Q2 and Q3 form the error amplifier. It is a differential amplifier, but with complementary devices rather than the more usual difference amp with two similar devices. But it works the same: the reference is input at Q3, and regulated the output voltage is input at Q2.

If the output voltage falls below the set value, Q2 will start to conduct more, and the drive voltage for the pass device across R5 will increase, which, in turn, increases the output until it is again at the set value. Similarly, when Vout becomes too high, the drive is reduced, lowering Vout back to the set value. Because Vout is equal to Vref except for a 2-Vbe offset, in principle you can set the output voltage to anything you want by just selecting R3. This is a great advantage to more traditional setups.

Traditionally, you would have a reference voltage that is some fraction of Vout, and then you divide Vout down to Vref before feeding it into the error amplifier. If you want to change Vout, you must change the division ratio. The disadvantage is that this also changes the loop gain of the control loop, which again means that the performance and stability change with output voltage. By using a reference voltage equal to Vout, as I have done, the stability and performance of the circuit does not change with output voltage.

You may wonder why the error amplifier is such a simple circuit. Certainly, using a high performance op amp here increases performance, but not enormously. And such op amp-based very high gain circuits bring their own issues with stability and compensation, for instance. In fact, the circuit as-is is already pretty high performance, as shown in **Figs. 3** and **4**.

One reason for the good performance is that the load resistor for the error amp is the resistor from Vdrive to ground, R5. Using a tube as a pass device, this resistor forms the grid resistor, which should be around 500kΩ, depending on the device. So, even a very small error between B and E of Q2 will lead to a relatively quite large "correction" current through R5, and thus to a quite large correction voltage at Vdrive. This loop gain is pretty large notwithstanding the simple circuit, so I decided not to use a high-gain op amp.

OTHER CONSIDERATIONS

This being a tube regulator, there are a few other factors to consider. One is the delayed application of the anode voltage, not just for the pass device, but also for the amp to be supplied. The full circuit (without the pass device) is shown in **Fig. 5**. U1 is a standard 555 (CMOS) timer that pulls down the LED in U4 some time after you switch on the supply.

The delay is set by R8 and C3, and with the given values is about 30 seconds. You can shorten it by lowering the value of C3, or make it longer by increasing R8. Depending on the leakage of C3, 1MΩ is probably the maximum value for R8. Once the LED in U4 is on, the opto-triac will fire and switch on thyristor U3, which will apply the rectified high-voltage to the pass tube (J4). Because the pass tube needs a heater voltage, I used that also to power the reference and control circuits through rectifier BR1 and capacitor C6.

An important factor is the "cleanliness" of the reference voltage. If there is any hum or noise on the reference, it will be faithfully duplicated by the error amp on the output. Current source Q1 gets its reference Vbe through the LED D5. With R1 being 1.3kΩ, the reference current is almost exactly 1mA, which makes the selection of the reference resistor R3 easy: just 1kΩ for every volt output you want.

LED D5 acts as a voltage source and is pretty clean. However, the first prototype did not have current regulator U2, but just a resistor to set the LED current. The hum on the control supply caused hum across the LED biasing resistor, which caused very small hum across the LED, but enough to appear in the reference. I also noticed that the output voltage decreased several 100mV with load changes of 100mA. The reason: both the high-voltage supply and the heater supply came from the same transformer.

Increasing the load made the transformer secondary voltage drop slightly, including the heater supply that supplied the reference and control circuit. This was enough to slightly lower the current through the LED, and therefore the reference voltage. In other words, increasing the load lowered the reference!

I then tried a voltage regulator for the reference and control supply. That worked, but I lost some flexibility with the heater voltages because a (5V) regulator also needs some dropout reserve. Finally, I settled for the current regulator U2, an LM344, which performed even slightly better than a voltage regulator on the supply.

There are some other parts on the schematic for protection purposes. The most obvious one is the fuse FU1 in series with the high voltage. Next there's protection for the transistors. For example, if the high voltage is switched on by U3, the reference voltage at the collector of Q1 needs time to rise up by charging C1. During that time, transients on the output line could be much higher than the safe value Vce for Q1. D2/D8 limits this value.

Protection for Q2 is also required for the same reasons: under transient conditions, Vout may peak above the required value. Q2 will try to correct that which may cause the Vc of Q2 to drop momentarily to a very low value. Vce of Q2 is limited to 100V by D1. This means that Vgrid can be maximum 100V negative with respect to Vout, the pass tube's cathode voltage. This value should be sufficient even with low-mu triodes and low load current. R14 limits the maximum current through the small-signal transistors during startup and transient conditions.

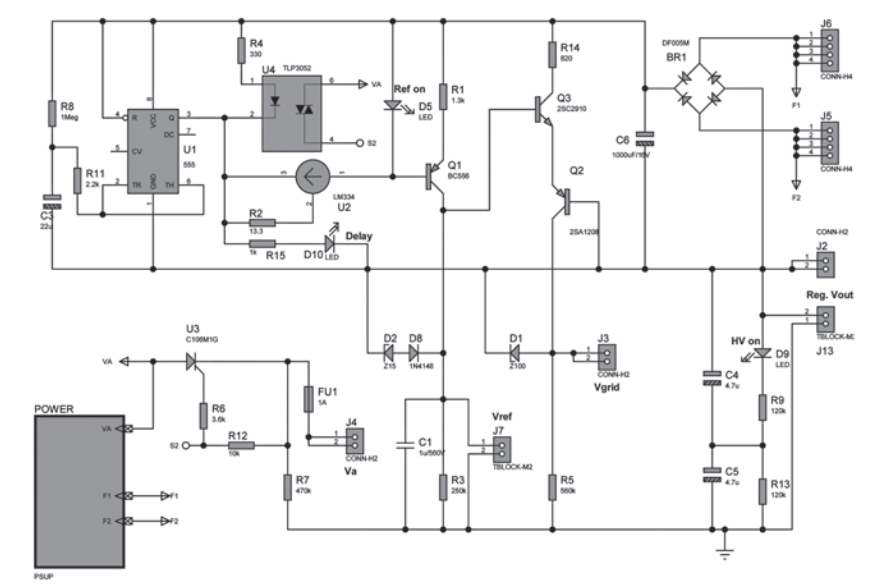

FIGURE 5: Complete motherboard circuit diagram.

CONFIGURATIONS

Because this supply is pretty universal as to the output voltage (in principle from the minimum Vgk of the pass tube up to 500V, limited by the voltage rating of the high-voltage rectifier and capacitors), I thought I'd also make it flexible as to the current it can supply. The rectifier, reference, and control circuits fit on a "motherboard" type of PC board (**Photo 1**). Then there is a small plug-in-board that carries the pass device and some resistors, and that is plugged into the motherboard. This way, you can use the regulator with different pass devices depending on your load current.

Currently, I have three plug-in-boards (**Photo 2**). For power amps, the plug-in-board with a 6528 dual triode tube can deliver up to 600mA (with a suitable power transformer, of course). This tube is specifically developed for series regulation.

A 6080 tube also fits on this plug-in board. Due to its lower mu, it has a slightly higher Zout and slightly higher hum and noise, but if you happen to have one of those lying around you can use it with good results. The plug-in fitted in the photo is a board with an EL84/EL86-type triode-connected small pentode for preamps and similar loads, up to 50mA.

Last, for those who want a "green" version and save 10W or so on the pass tube heater, there is a board with a depletion-mode MOSFET, a DN2540. This particular plug-in-board can accommodate two DN2540s and their heatsinks for an output load of more than 1A, depending on the dissipation from the Vds drop.

The circuits for the plug-in-boards are

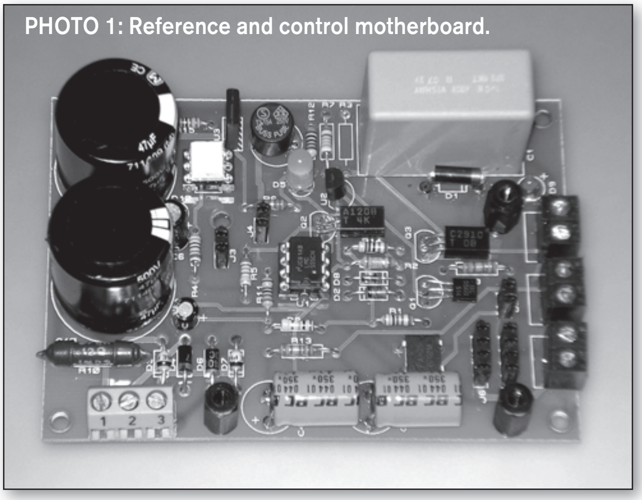

PHOTO 1: Reference and control motherboard.

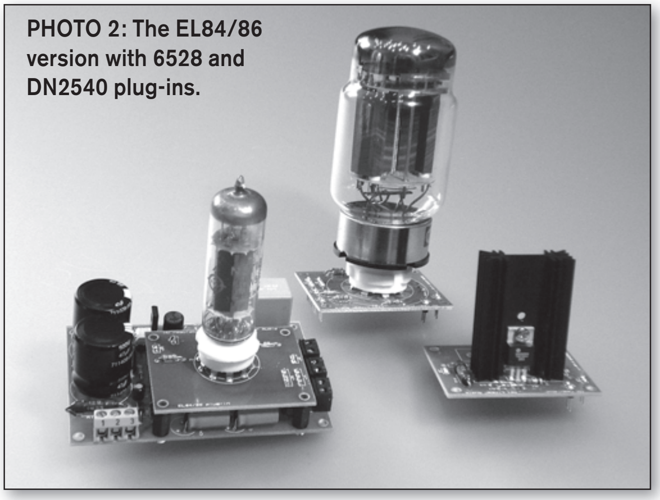

PHOTO 2: The EL84/86 version with 6528 and DN2540 plug-ins.

shown in **Fig. 6**. Because the 6528 (6080) are dual triodes, there are small value cathode resistors to equalize the currents at higher loads. The same is used on the DN2540 board in case you use two devices for increased load capacity (you can short the source resistor if you use only a single DN2540).

The plug-in-boards also have grid and gate stoppers for added stability. The DN2540 board has an additional 100V/5W zener diode (D5) to protect the MOSFET during switch-on and load transients as well as Vgs protection zener diodes. It also accepts a pair of standard heatsinks; tubes are definitely at an advantage in the dissipation department. The boards are connected with standard 0.1″ pitch PCB connectors and headers cut to size. Note that when you use the MOSFET pass device you don't need the heater connectors between the two boards. M3 machine screws and 12mm standoffs secure the plug-in to the motherboard.

PERFORMANCE

Figures 2 and **3** show the output impedance and output hum and noise of the various versions over the audio band. The performance is quite respectable for a high-voltage tube supply. The best all-round performer is the MOSFET, because of its high transconductance; however, differences are small. Hum and noise under load are typically below 500µV RMS in a 20kHz bandwidth for all versions (mostly line-related hum components).

You may wonder why I actually built three versions. Why not just use the DN2540? It is the best performer, and it has another advantage: a lower dropout voltage. The DN2540 works fine with just 10 or 15V Vds (depending on the input ripple), while the tube versions require several tens of volts Vak to operate cleanly. This means that for a given transformer the DN2540 provides 10 to 15V more DC output and less dissipation. But there's more to it than just measurements.

Although I would be hard-pressed to

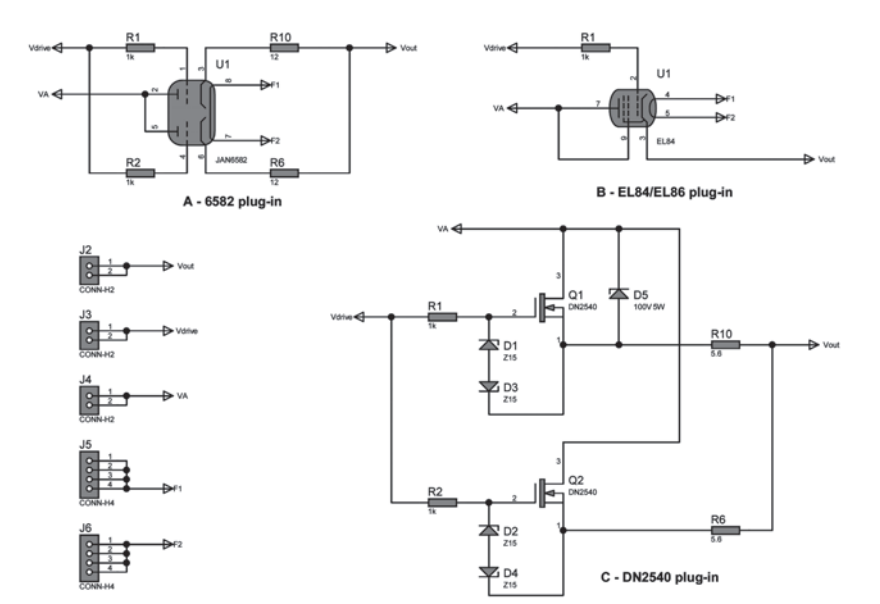

FIGURE 6: Plug-in board/pass device circuits.

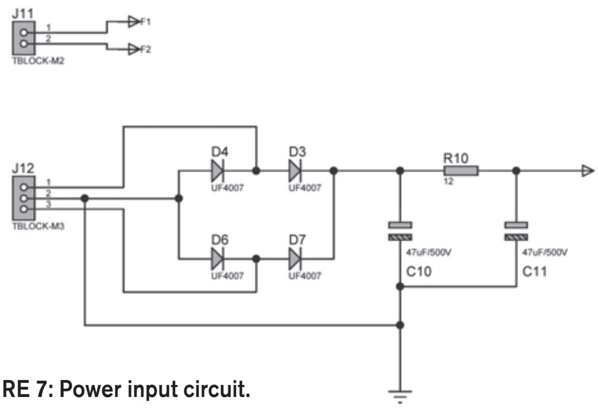

FIGURE 7: Power input circuit.

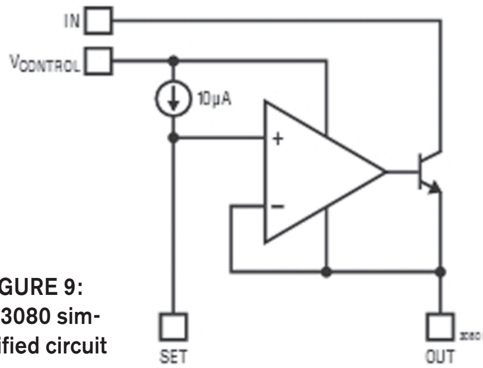

FIGURE 9: LT3080 simplified circuit diagram.

hear any difference between those regulators powering an amp, there's always the "X-factor." You may prefer a tube regulator in a tube amplifier for perfectly good reasons. So, the choice is yours!

CUSTOMIZATION

The input voltage is limited by the voltage rating of the rectifier diodes and capacitors—500V peak in my version. That limits Vout to about 485V with the DN2540. If you need more output voltage, you can use higher-rated diodes for D3, 4, 6, 7, and C10, 11 (**Fig. 7**).

Note that the diodes should be rated for at least twice the rectified peak voltage, preferably more. It is always wise to keep a healthy margin for the diode and capacitor ratings to cope with unloaded output voltages and high mains levels. You can use a transformer with a single secondary connected to J12 pin 1 and 3, or a center-tapped secondary with the tap at J12 pin 2. In the latter case you can omit diodes D4 and D6, of course.

The board has provisions for a small resistor between the two reservoir capacitors (R10) to take the high-frequency edge off the ripple voltage. The 12Ω is probably a bit too high for load currents above 100mA, so you should experiment with it, or short it out altogether. You can also use off-board reservoir capacitors and/or chokes. Lowering the ripple on the rectified voltage will not, however, bring a 1:1 improvement on Vout because the ripple on Vref will start to dominate. As-is, the performance is pretty well balanced.

As noted, the output voltage is 1.2V below Vref. You can easily set Vref by selecting R3. With the reference current approximately 1mA, Vref (in volts) = R3 (in kΩ). Note that you can use an external resistor or pot meter at J7 to set Vout. If you use a pot, be sure to use one that can withstand 500V! Provided that you have enough rectified input voltage, you can set Vout to any value within reason. The minimum Vout depends on the negative grid/gate drive voltage. If you make Vref = 0 by shorting R3, the DN2540 gets pretty close to zero as well, but a 6528 will not go lower than tens of volts (depending on the load) because of the necessary negative grid bias. The output at the cathode will need to be above the drive voltage at the grid, except for quite high output currents.

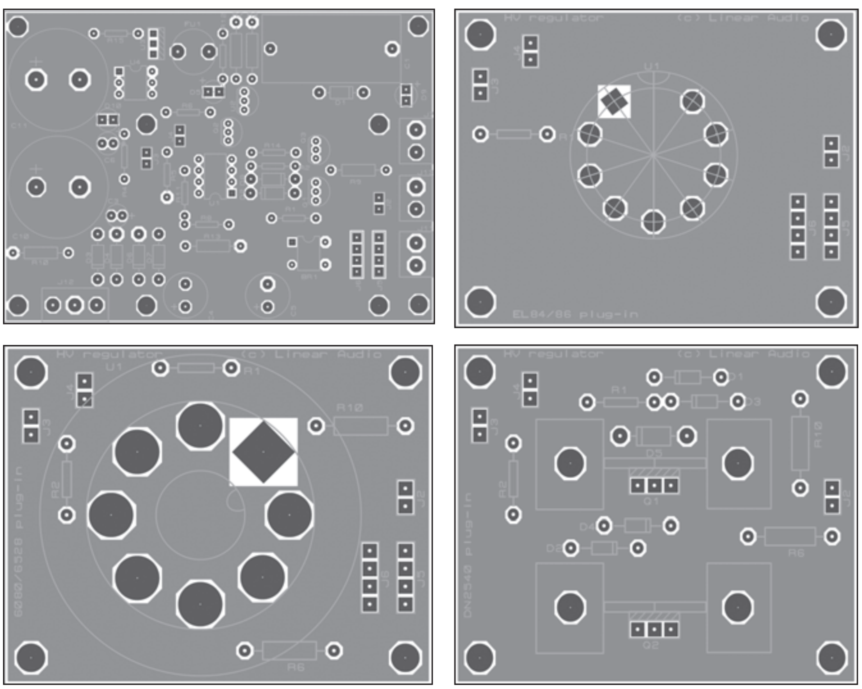

FIGURE 8: PCB stuffing guides.

CONSTRUCTION

Before starting construction, note that this supply contains points with lethal voltages and currents, including the heatsinks for the DN2540s! You are responsible for your own safety, not me, nor the folks at *aX*. Double-check that the high-voltage input is disconnected or turned off before working on the board.

Discharge the high-voltage capacitors with a 1k resistor. Make it a practice to keep one hand in your pocket when touching live boards with a meter probe or anything at all to avoid body currents. Even when switched off, the high-voltage capacitors may still have a dangerous charge. Discharge them.

Warning: the heater supply for the pass tube is also used for the reference and control circuits, and floats on the regulated high-voltage. Do *not* use this heater winding for any other tubes in the equipment to be powered. If you use the DN2540 plug-in board, you still need a separate floating supply for the reference and control circuits. In any case, you could also use a small separate 5 or 6V AC transformer, provided it has enough secondary insulation to allow it to float to up to the maximum Vout.

Construction is straightforward when using the motherboard-plug-in board approach. **Figure 8** shows the component stuffing guides for the motherboard and the plug-ins. The complete parts lists are in **Tables 1A-1D**.

It is best to start with the small parts on the motherboard, such as resistors, diodes, transistors, and headers, leaving the larger caps for last. Take care to mount the pins and headers straight upright. The hole locations on the plug-in boards correspond exactly to those on the motherboard, but if the pins and headers are slightly off vertically, they mate less smoothly. The transistors are below the plug-in board. Because of the height limitation, you should fold them horizontally on the board as shown. Do the same with the output electrolytics C4 and C5. The layout has open space for that.

You can test the motherboard without plugging it in, to verify that the delay and thyristor circuits work. LED D10 should light when switched on, while D5 should come on after the delay time. (D9 will not come on unless you have a pass device plugged in.) The input voltage at J4 should also come on after the delay. Next, populate one of the plug-in boards, mount it on the motherboard, and check out the whole assembly.

Well, that's it. I'd appreciate some feedback about your experiences with this supply. You can contact me via my website www.linearaudio.nl. I will also post any additions and tips there. Happy building and happy listening!

POSTSCRIPT

After completing the prototype, I received an e-mail copy of the June 2008 issue of *LT Magazine*, which featured the new LT3080 Linear Regulator (**Fig. 9**) that looks conceptually *exactly* like an MC1466L that was declared obsolete more than a decade ago. What's more, this chip was selected by EDN as Innovation of the Year, while LT's Chief Technical Officer Bob Dobkin was named Innovator of the Year. I guess you should be very careful about what you declare obsolete. *aX*

TABLE 1A

T-reg parts list main board

Resistors (1% 0.5W)	1 R1	1.3k	
	1 R2	13.3	
	1 R14	820	
	1 R15	1k	
	1 R3	250k	see text for actual value
	1 R4	330	
	1 R5	560k	
	1 R6	3.6k	
	1 R7	470k	
	1 R8	1M	
	2 R9, R13	120k	
	1 R10	12	see text for actual value
	1 R11	2.2k	
	1 R12	10k	
Capacitors	1 C1	1µ/560V	
	1 C3	22µ	
	2 C4, C5	4.7µF/250V	
	1 C6	1000µF/16V	
	2 C10, C11	47µF/500V	
ICs, diodes, transistors			
	1 U1	LMC555	
	1 U2	LM334	
	1 U3	C106M1G	Triac, 600V 2.5A
	1 U4	TLP3052	Optocoupler
	1 Q1	BC556	
	1 Q2	2SA1208	
	1 Q3	2SC2910	
	1 D1	Z100	zener diode 100V/5W
	1 D2	Z15	zener diode 15V 0.5W
	1 D8	1N4148	
	4 D3, 4, 6, 7	UF4007	fast/soft recovery rectifier 1000V 1A
	3 D5, 9, 10	LED	1 each green, red, yellow LED 0.2″ pitch
	1 BR1	DF005M	bridge rectifier 50V/1A
Miscellaneous	1 FU1	1A	fuse, round, plug-in, 2.5A slow
	1 fuse socket		2-pin 0.2″ pitch
	1 IC socket	DIL-06	
	1 IC socket	DIL-08	
	3 J2-J4	2-pin header strip, 0.1″	
	2 J5, J6	4-pin header strip, 0.1″	
	3 J7, J11, J13	two-way screw PCB terminal block , 0.2″	
	1 J12	TBLOCK-M3	
	4 standoffs	M3 * 12mm	
	8 machine screw	M3 * 6mm	

TABLE 1B

T-reg 6528 plug-in parts list

Resistors	2 R1, R2	1k 1% 0.5W
	2 R6, R10	12 1% 2W
tube	1 U1	6528 or 6080
miscellaneous	3 J2-J4	two-way socket strip, 0.1″
	2 J5, J6	four-way socket strip, 0.1″
	1 tube socket	PCB octal socket

TABLE 1C

T-reg EL84 plug-in parts list

Resistors	1 R1	1k	1% 0.5W
tube	1 U1	EL84 or EL86	
miscellaneous	3 J2-J4	two-way socket strip, 0.1″	
	2 J5, J6	four-way socket strip, 0.1″	
	1 tube socket	PCB noval socket	

TABLE 1D

DN2540 plug-in parts list

resistors	2 R1, R2	100Ω	1% 0.5W
	2 R6, R10	5.6	1% 2W
transistors	2 Q1, Q2	DN2540	
diodes	4 D1-D4	Z15	zener diode 15V 0.5W
	1 D5	Z100	zener diode 100V 5W
miscellanous	3 J2-J4	two-way socket strip, 0.1″	
	2 heatsinks	4° C/W	Wakefield 637-20ABPE

Building a Speaker Testing Turntable

A simple DIY turntable can make speaker testing easy and accurate.

PHOTO 1: Lazy-Susan bearings.

Testing speakers requires taking measurements both on-axis and off-axis. You can do so by moving the measurement mike, but the job is greatly simplified and accuracy is ensured with a testing turntable.

The heart of the turntable is the lazy-Susan bearing, which is available in various sizes both round and square framed (**Photo 1**). You can order them online or find them in most local hardware stores. A size between 6 and 10″ will do for all but the very largest pro-sound cabinets.

Along with the bearing, you'll need a 2 × 4′ piece of ½″ to ¾″ plywood. Cut the plywood into two 2 × 2′ pieces. Leave one piece square for the base, and cut the other into a 24″ circle for the top, using a jigsaw or a router and circle gutting jig. Drill a 1/8″ hole through the center of both pieces. Draw a line on the square piece from the center hole to one corner.

This is the zero degree reference.

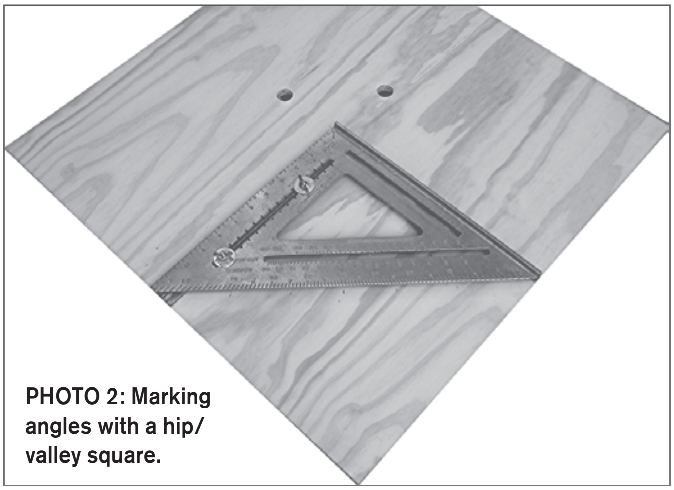

PHOTO 2: Marking angles with a hip/valley square.

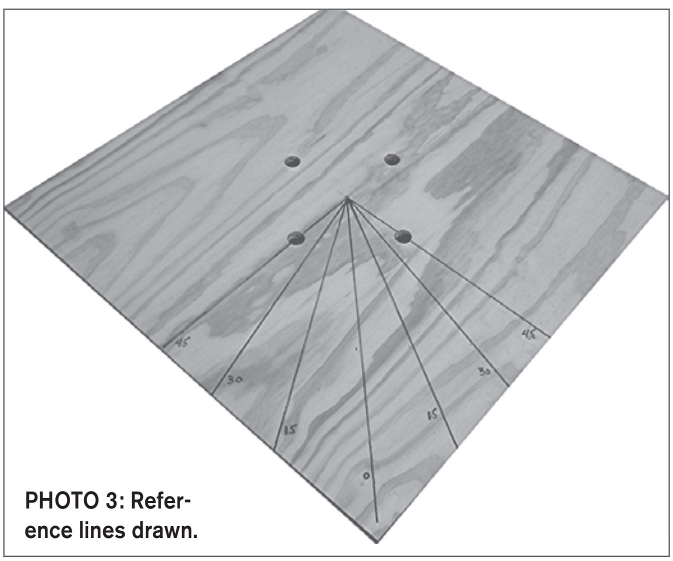

PHOTO 3: Reference lines drawn.

PHOTO 4: Turntable top in place.

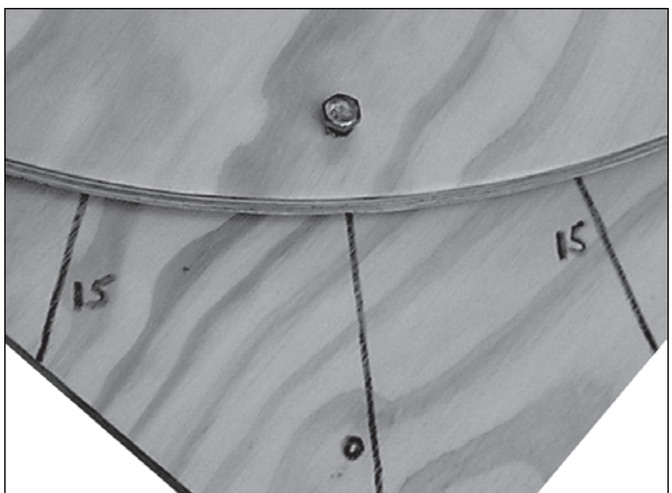

PHOTO 5: Stop bolt detail.

Using a protractor or a hip/valley square, draw additional lines radiating from the center hole at 15, 30, and 45° to either side of the reference (**Photo 2**). Draw and label the lines with a marking pen, so that they are easily visible (**Photo 3**).

Draw a line bisecting the top and then another line at a right angle to the first from the center hole to one edge. On the second line drill a ¼″ diameter hole about an inch from the edge. Install the bearing according to the instructions supplied with it. A nail inserted through the center holes in the base and top will align them for the bearing installation. Rotate the top, placing the hole over each of the reference lines on the bottom, in turn drilling through the hole into the bottom on each. A ¼″ bolt inserted through the hole will secure the top at each "stop" (**Photos 4** and **5**).

To render the top non-reflective, staple a half-moon-shaped piece of heavy felt carpet underlayment to it with its straight edge on the bisect line (**Photo 6**). To use the turntable, center the speaker atop the table, with the baffle on the bisect line (**Photo 7**). Place the meter/mike on the table reference axis. When rotated to the various angles, the center of the baffle remains at a constant distance from the meter/mike. Use carpet underlayment to cover the ground or floor between the turntable and meter/mike. For best results measure outdoors, away from any buildings or objects. *aX*

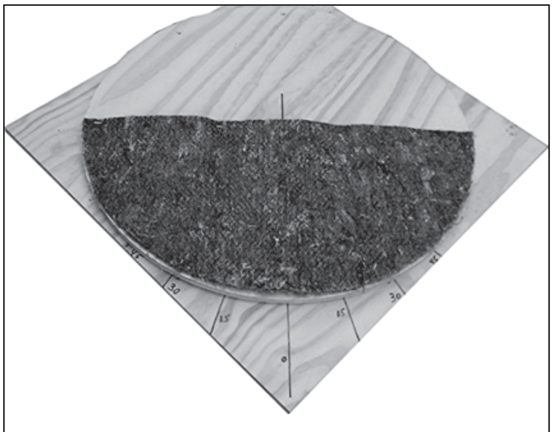

PHOTO 6: Felt carpet underlayment absorbs reflected waves.

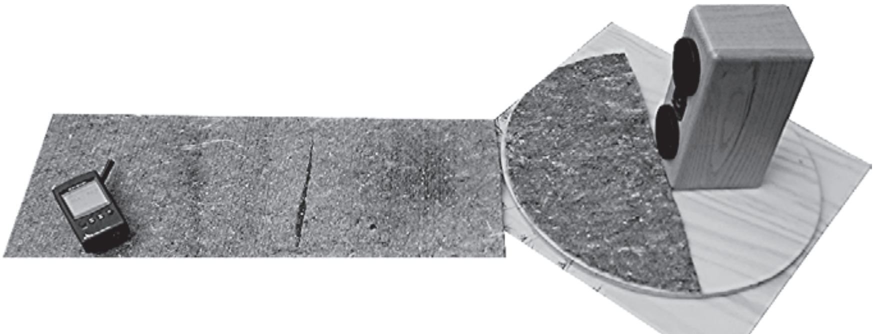

PHOTO 7: Taking a ground-plane measurement.

SchmartBoards

By Ed Simon

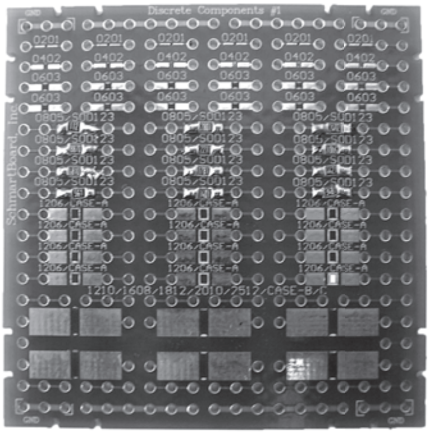

PHOTO 1: SchmartBoard.

Ed Dell asked me whether I would be interested in reviewing the new SchmartBoard prototyping system. What he didn't know is I already was unhappy with my current approach to soldering some of the new IC package formats.

I recently had a chip intended to measure the power of radio signal sources. This chip is only available in a small outline integrated circuit (SOIC) package. I first applied a very small amount of silicone rubber cement to the printed circuit (PC) card. I then jiggled the chip until it was where it needed to be. This was not as easy as it sounds because the PC traces are a bit above the board and the chip leads wanted to slip into the low spots.

Once the chip was in place, it sat alone and untouched for a day to let the cement dry. The next step was to use my built-in seven diopter magnifier (I took off my glasses) to see what was being soldered. I used the finest tip available for my Weller WLC 100 soldering station and .020″ solder so attaching the chip was doable. Adding a few chip resistors and capacitors finished the circuit. This was a large amount of effort for a very small circuit.

A BETTER WAY

SchmartBoard has come up with a series of small 2″ square boards on which the solder traces are the low part of the PC card! The solder mask on the side of the traces is thicker than the copper pads—no need to apply glue, jiggle, and wait. The method involves dropping the chip in, and gravity pulls it home. Then they suggest you tape one half down while you solder the other half. There also is a plastic gripper so that you can link multiple boards together to form bigger circuits.

If you do not hold the other side down when you install the chip it is quite possible that as the solder melts to the first lead, the chip will tilt up into the air on the unattached leads.

SOLDERING TIPS

The first issue that may slow many users is that they ask you to first flux the connection before soldering. Liquid non acid flux is the right stuff to use, but most folks probably do not own any. Liquid flux is normally only used with a soldering pot, so I decided to try the system without using the flux. To make it typical, I used a Radio Shack modular fixed temperature 25W soldering iron with a 1/8″ chisel (rectangular) tip.

The important issue when soldering pieces this small is to have a steady hand. The first rule here is to be sure your arm is resting on a solid surface. If you keep your arm in the air and try to keep it steady, you may have too much motion to get a good solder joint.

The Schmart folks recommend you just use the solder already on the board without adding any. They also suggest you use a fine tip to heat only one lead at a time to prevent solder bridges. I made sure my tip was cleaned and then tinned before starting.

For a first try at soldering, I used the side of the tip which was narrow enough to contact only one lead at a time. This kind of worked. Using the wide part of the tip allowed me to solder three leads at a time. After finishing, the chip looked happy. Wondering how to test this, I tried to see whether the chip would move. It fell right off! You could see where the solder on the PC card was melted but did not attach to the chip.

Lesson learned: follow the directions! I applied flux to the SchmartBoard and moved the chip to some new traces and tried again, this time using just a clean tip with no extra tinning. As I applied the soldering iron, smoke arose from each joint as the flux burned off and cleaned the metal. This time the joints were perfect! The board had beautiful clean solder flows at each lead.

For audio use you probably do not use SOICs very often, but I had wanted to try some of the new chip resistors with astoundingly low temperature coefficients. The price of the best .02% ±5 PPM resistors was more than $10 each so this was a good reason to experiment with the 0.5% 25 PPM ones.

The SchmartBoard Discrete components #1 seemed to be just right for this test. Placing the resistors on the board yielded the first surprise. This board did not have wells for the components. I used a pair of mild steel tweezers to pick up and place the chips; the plastic ones seemed to be hiding.

Turns out that was a good thing. After I placed the chips, they wanted to stick to the tweezers. The tweezers were magnetized and the chips contained iron.

I had a nice lunch a year ago at the San Francisco AES show with a fellow who designs high-quality microphone preamps. Mentioning to him that I was using Brand H resistors for a project, he replied he had found they were not as good as they used to be because they now stuck to magnets when they previously did not.

His experience and mine is that many components these days use iron or steel in their construction. These parts tend not to work as well for audio. This problem first popped up with film capacitors.

Even though the chip resistors stuck to the tweezers, I decided to go ahead with the experiment. With skill and luck (a small piece of tape helped), I attached the components to the board.

When you buy custom PC boards over the Internet, they charge by the setup and then the area. It is just plain cheaper to buy the entire panel than just one card. Saving up designs and building an order board often leaves small areas that are left over. You can use these to make prototype cards.

COMPARISON

Using a leftover prototype card designed to hold the 0805 size chip resistors gave me a chance to compare the SchmartBoard to a standard card.

The prototype card was tinned with a lead-free solder. Placing the resistors was sometimes harder because the surface was not as smooth as the SchmartBoard. The custom card had more space, which made up for the rougher surface. The SchmartBoard had a bigger pad size. The end result is that I did not like using either board. Leaded components are still my choice when available. This may change as I become more experienced or SchmartBoard recesses the chip slots.

The SchmartBoard has a ground plane on the backside of the components card. My custom one did not. The SchmartBoard is lead-free, holds more parts, and is off the shelf. In addition, there is a family of boards that you can combine for a larger project.

Dennis Colin could certainly use this system instead of just tacking parts on a too small scrap of PC laminate. If you need to use one of the newer small ICs, the SchmartBoards are a wonderful choice. Be sure to buy some liquid flux when you buy your cards. I hope in the future they extend the technology to include the discrete components.

Manufacturer's response:
We are happy that Mr. Simon had an easy time (when using our instructions), hand soldering an SOIC. This is the experience that most people will have. When soldering small discrete surface mount ICs as in the article, SchmartBoard pads are recessed, but to a smaller degree than the SchmartBoards for the SOICs. Some customers have used plastic tweezers when soldering very small discrete components to avoid the hindrance of dealing with magnetized tweezers. We do have videos of soldering the different types of components on our website www.schmartboard.com/index.asp?page=resources_howto. Readers can also go to our website and request a demo board and chip that will allow them to try the technology before purchasing: www.schmartboard.com/index.asp?page=schmartland_samples

Neal Greenberg
VP Sales & Marketing
Schmartboard, Inc.

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2. Heath "W-5M" power amp. New coupling caps and driver stage plate load resistors, matched to <1%, two extra 5R4GYB NOS rectifiers. Works nicely and includes Heath book.

Both: Best offer
rbgrid@concentric.net
518-966-4090

Wanted

Hi Fi and Stereo equipment, tubes and transistor, kits and factory built (Heathkit, Eico, Scott, Dynaco, Hafler). Turntables, records, reel-to-reels, tapes. I am a collector, not for resale. Please e-mail me at njvet01@yahoo.com

Two Amperex AD12251W8 12" woofers.
Tom Tutay
850-244-3041

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Ad Index

The **Restoration Preamp**

By Don Walizer

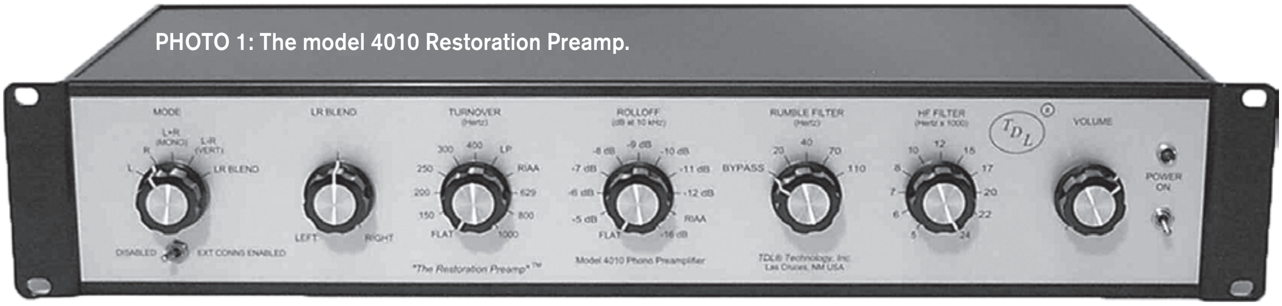


PHOTO 1: The model 4010 Restoration Preamp.

Hats off to TDL Technology for developing the 4010 audio restoration preamp. For the past 15 years I have been searching in vain for a modern replacement for the venerable OWL Audio Restoration Module that would incorporate the OWL's functionality with updated electronics and a more refined filter design. During that time I looked at the half dozen or so available alternatives that have come and gone, most of which were grossly overpriced, and none of which offered the intelligent design choices of the OWL for people genuinely interested in doing audio restoration. I was really excited when I heard about the 4010 because it seemed to incorporate everything that I had been hoping for in an audio restoration preamp. Having owned another TDL Technology preamp, I was pretty sure that the 4010 would have good sonic qualities; but would it really live up to my expectations?

PERKY PLATTERS

I have been using my 4010 for two months now and can say that it is everything that I ever wanted in an audio restoration preamp. The 4010 makes it easy to restore even the most challenging recordings. I can quickly check the left and right groove walls for excessive distortion and blend them for best overall result. The L-R (vertical) setting lets me check for low-frequency noise and set the rumble filter correctly. (Turntable rumble isn't an issue with my KAB Broadcast Standard turntable, but it is amazing how much low-frequency noise is present on some recordings as a result of the cutting, mastering, and pressing operations.)

The most important equalization setting is finding the correct turnover. This is what really brings a recording to life, and the 4010 excels in its ability to bring the sonic field into correct focus. For example, I recently made a CD of jazz recordings from the mid-30s to the mid-40s. Two of the tracks from a 1937 Coleman Hawkins recording just didn't sound right in comparison with the other tracks I'd restored. They were sort of dull and lifeless. I'd used a 300Hz turnover without experimenting because it was the "correct" setting for that label and year.

After wondering why these tracks just sounded OK while the other tracks sounded great, I checked the record label—it had been recorded in England! Well, that probably meant that I should have set my turnover to a lower setting. When I played the record back using a 250Hz turnover, it jumped to life. Everything sounded right, and did it swing!

Who would ever think that going from a turnover of 300Hz to 250Hz would make such a dramatic difference? Because I never owned a restoration preamp like the 4010, I was not able to make a comparison like this before. In fact, as far as I know—and I've done a lot of research on this—you can't buy another preamp that will give you this capability. I don't want to downplay the importance of accurate rolloff equalization and the usefulness of a properly designed lowpass filter. The 4010 excels in these areas as well and, again, offers more settings than any other product you can buy.

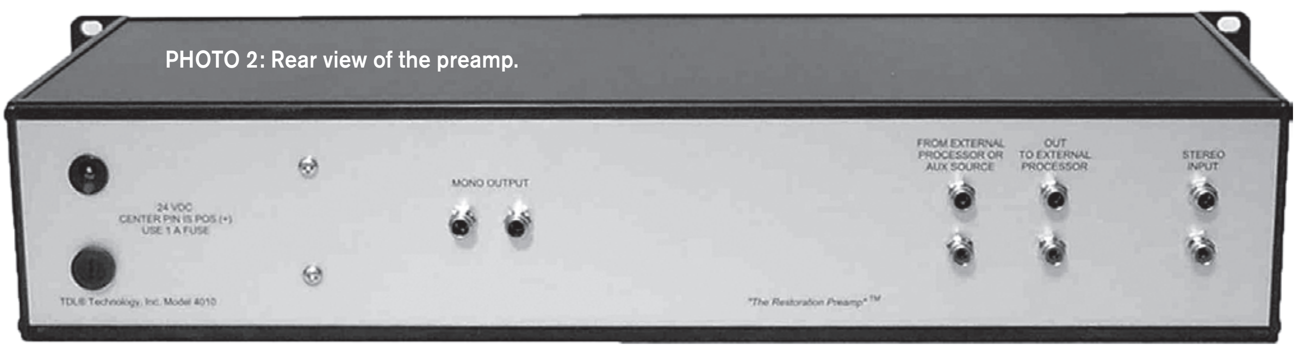


PHOTO 2: Rear view of the preamp.

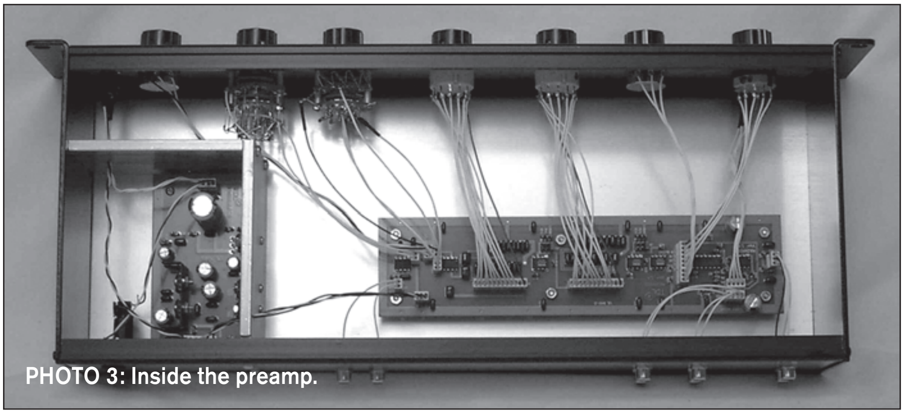


PHOTO 3: Inside the preamp.

Other things I like about the 4010 are characteristics that it shares with other products from TDL Technology. First, like me, the company is fanatical about noise. Consequently, the 4010 is dead quiet. If there is hum, buzzing, RFI, or EMI in your system, look elsewhere for its cause.

Second, build quality is second to none. High-quality components are used throughout and the 4010 has a solid feel about it. Each unit is hand assembled and undergoes a 100-hour burn-in test. The user manual contains complete block and circuit diagrams, useful information about how to use the product, and a full parts list. If you have a question, you get a prompt, courteous reply. Who says that good customer service doesn't exist anymore in the audio industry?

I sold my OWL two weeks ago. The 4010 does everything the OWL did, only better. The king is dead, long live the king.

TDL Technology, Inc., 5260 Cochise Trail, Las Cruces, NM 88012, 575-382-3173, Fax 575-382-8810, www.tdl-tech.com. aX

CONTRIBUTORS

Kit Ryan ("Modding the Music Hall CD25.2 CD Player," p. 6) has been an avid audio hobbyist for 50 years with many self-designed and hand-built audio and test equipment projects under his belt. His goal is to modify or build high quality gear without the high cost while having fun at the same time.

Ron Tipton ("DIY Microphone Calibration," p. 18) has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957 he started Testronic Development Laboratory (now TDL Technology, Inc.) to develop audio electronics. During the 1960s and 70s, TDL built active filters and pseudo-random noise generators for well known companies such as Bose Corp. and Acoustic Research. He is still the TDL president and principal designer.

Jan Didden ("T-reg: A High-Voltage Regulator for Tube Amps," p. 24) 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.

Bill Fitzmaurice ("Building a Speaker Testing Turntable," p. 30) has been a professional musician since 1966 and has been constructing instruments, amplifiers, and speakers for just as long. Vice president of DeltaSounds Loudspeakers Inc., Fitzmaurice is the author of *Speaker Builder's Loudspeakers for Musicians* and over 30 magazine articles dealing with speakers and electric instruments, as well as the action-adventure novel *Operation: Sergeant York*. Bill and his wife of over 30 years reside in Laconia, N.H.

Ed Simon (Review: SchmartBoards, p. 32) received his B.S.E.E. at Carnegie-Mellon University. He has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; MCI Center, Washington D.C.; Museum of Modern Art Restaurants, New York; The Coliseum, Nashville, Tenn.; The Forum, Los Angeles; Fisher Cats Stadium, Manchester, N.H.

Don Walizer (Review: The Restoration Preamp, p. 36) resides in Arizona.

Ian Hilton ("Keeping The Hobby Alive," p. 38) has been working in the electronics industry for 35 years. He has been involved in designing electronic equipment for use in coal mines and aircraft. He is currently employed as an analog engineer involved with optical fiber and laser technology. He has been building amplifiers on and off for donkey's years and lives with his wife and two daughters in Hampshire, England.

solid state By Ian Hilton

Keeping the **Hobby Alive**

The author presents a design for a very cheap headphone amplifier that gives surprisingly good results.

You know what it's like; you cannot play music in the lounge anymore because it interferes with "family viewing," so the Magnepan SMGas (small-ish flat panel speakers) stand mute. I suppose there's always the portable CD player, but it does tend to get in the way, doesn't it?

MP3 SCENARIO

What about one of those nifty MP3 players that my teenage daughter has had grafted onto her for several years now? They can hold a gazillion hours of music tracks and are tiny, and the quality is supposed to be quite good. After a £10 investment to purchase the player, insert the battery and load up some tracks from the PC.

Brief chat with daughter ensues: what are those bits of black foam on the end of that skinny bit of wire? Headphones? Surely not. Oh, earbuds. You put them in your ears? Well, OK. Oops, it fell out. Try again.

Ah, what if I hold my head like this? No, the other one's fallen out now. What if I clench my ears (clench my ears?) and stand like this and grit my teeth. Oh, that's frightened the cats away. Brief pause to throw earbuds out of window.

Let's get the trusty old Sennheisers out, plug in (oops, need an adaptor), and let's listen to some music. Well that sure does sound surprisingly. . . dreadful! Clearly MP3 is rubbish. And it really doesn't go very loud, does it?

JUNK BOX SOLUTION

Well, I must be able to do something about this, at least just to get the volume levels up a bit to drown out the sounds of domestic harmony. Whatever I use, it must be portable with a little bit of gain and capable of driving a simple pair of headphones. This junk box project shouldn't take too long. Let's see, an AD712 and a 9V battery will do. Need to generate a mid-point to give a ±4.5V supply. Two resistors? Hmm, not keen on that. Or, I could use an op amp to generate the mid-point. Hmm, that might drain the battery too much.

What about two resistors and a simple emitter follower to generate the mid-point? That sounds like the best approach. Here we go, all done, now plug it in for a bit more volume and. . . Wow! Where did all that music come from?

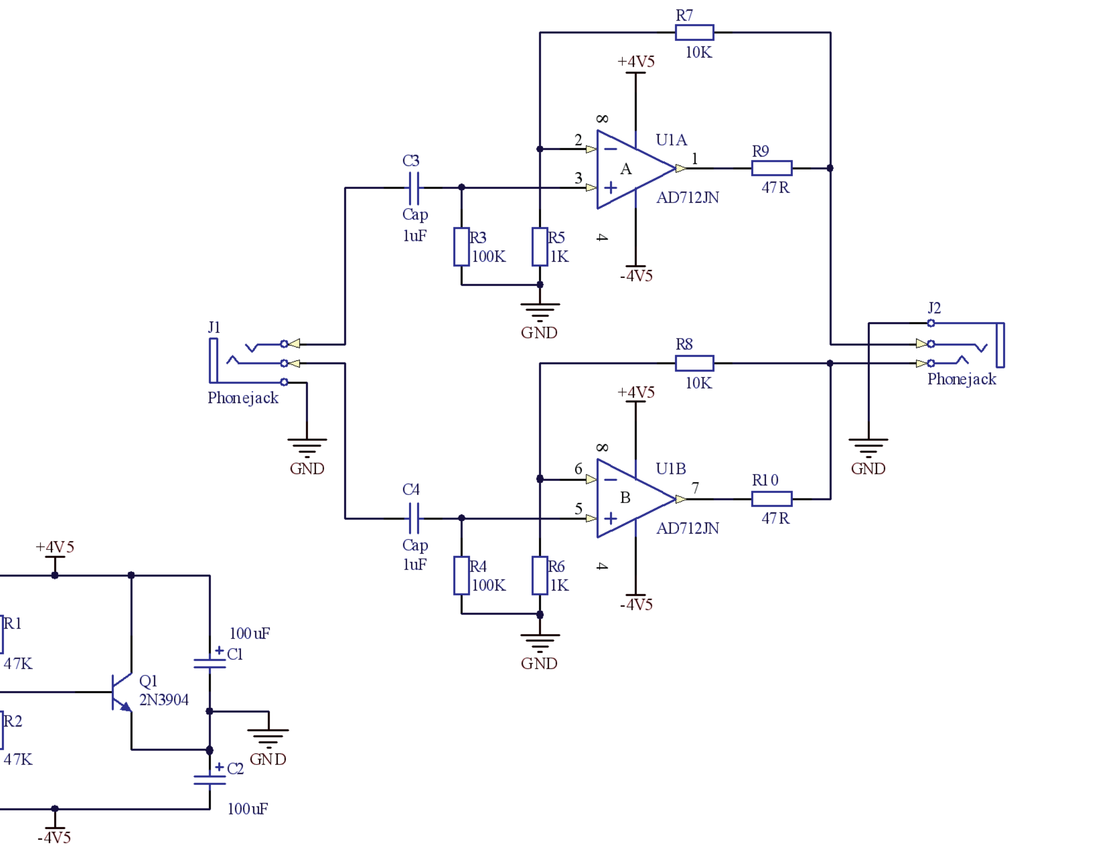
Not only is it louder, but the sound is now three-dimensional and much, much bigger. I can certainly hear details that were hidden before. Perhaps the MP3 format is better than reported. This really is a revelation. What if I had used better quality components, a nifty aluminum case, and better connectors, as well as good coupling caps?

Figure 1 is the schematic for the minimalist—almost zero cost headphone amp. I claim no originality for it, because after I had built it I found loads of references to similar designs on the Internet. Maybe the emitter-follower biasing is a bit unusual, but it works fine. I did not include a volume control, because the unit works OK with the volume control on my MP3 player. I also use it to drive an FM transmitter from the MP3 player in the car. *aX*

TABLE 1

U1	AD712
R1, R2	47k
R3, R4	100k
R5, R6	1k
R7, R8	10k
R9, R10	47R
C1, C2	100μF 25V electrolytic
C3, C4	1μF film cap
Q1	2N3904
J1, J2	3.5mm jack socket
S1	SPDT
B1	9V battery
	battery clip
	case
All resistors 0.25W	

FIGURE 1: Schematic.



HIGH PERFORMANCE CURRENT REGULATORS REVISITED

A Special Note: This test data is still very preliminary, and subject to changes. It is being shared for purposes of commentary and critique.

Several readers wrote in with comments and questions related to the use of the LM317 and MOSFET current sources within parts one and two of "Sources 101"[1, 2]. These experiences (not all of which were good) led to the following discussion of two "all-weather" current sources/sinks. These current sources are suitable for virtually any type of audio application—that is, either power related or within the signal path, at currents from just a few to 100mA.

TWO "ALL-WEATHER" CURRENT SOURCES/SINKS USING MOSFET CASCODES

The key development that came about deals with the improved MOSFET cascode current sources of **Fig. 1**, both using the DN2540 depletion mode MOSFETs. There are really a couple of parts to this story-within-a-story. One is that it was in part stimulated by some e-mail exchanges, the first of which used a MOSFET current source in a power amp. More importantly however, some later e-mail discussions revealed disappointing performance with the original *Fig. 13C*[2], *when this circuit was used as an active load for a triode preamp stage*. This was traced to an excessive white noise output, which in turn was due to the use of the LM317 within Fig. 13C. In this circuit the LM317 establishes the basic current operation of the circuit, insofar as noise properties.

Although my own original use of the Fig. 13C circuit was within a power supply regulator, I made a definite mistake in not establishing a caveat on noise vis-à-vis the LM317, *if used within signal path circuits*. Its noise level is such that while it might be OK for use as a push-pull power stage cathode bias (i.e., high level), it definitely isn't suited for lower level stages. In fact, the feedback I got was this: The Fig. 13C circuit was indeed OK in the high level tube applications, but not in preamps—*due to a high residual hiss level*. I apologize for not anticipating this type of use, and regret any inconvenience caused.

Fortunately, there is a simple and very attractive alternate solution—just use one of the two following advanced DN2540 cascode circuits. And, the second part of the story comes from the stellar performance achieved with these cascodes. It took some special means to see just how good they really are!

The schematic of the new circuits is shown in **Fig. 1**, which consists of a conventional cascode (left), plus what I think is a very useful wrinkle, a bias-multiplying cascode (right). The left circuit isn't really new, because it is a relatively straightforward application of FET device cascoding, and this setup is similar to what my correspondents used. As discussed[1], it is usually desirable in this type of circuit to have the Vgs (off) of the upper device (M2) be specified as substantially higher than the lower part (M1). But, if two identical parts are used just as shown, the D-S voltage seen by M1 becomes the Vgs of M2—not always best for highest rejection. Nevertheless, with a simple circuit such as this, and no other high Vgs (off) family parts to use, you use what's available, the same part for M1 and M2. And, it still works, quite well, in fact.

The slightly more complex circuit to the right adds a second pair of resistors, which provide a DC gain for the gate bias of M2, as:

$$V_{DS}(M1) = V_{GS}(M2) * (1 + R3/R4)$$

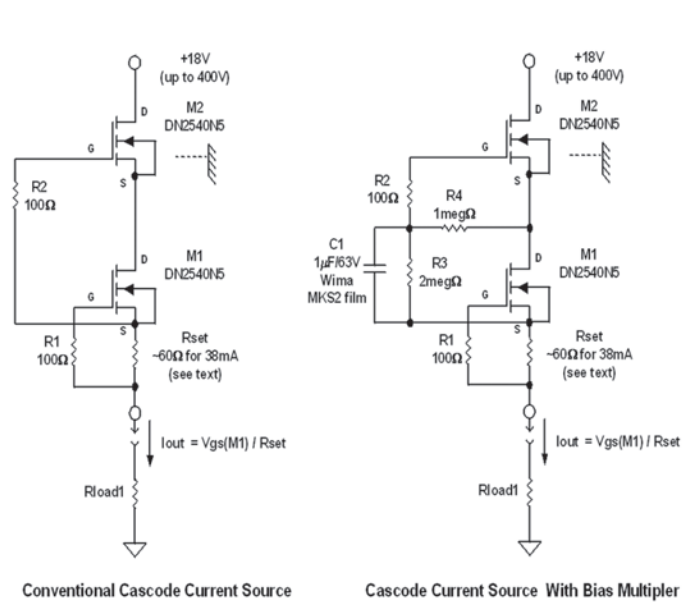


FIGURE 1: DN2540 Cascode MOSFET Current Sources.

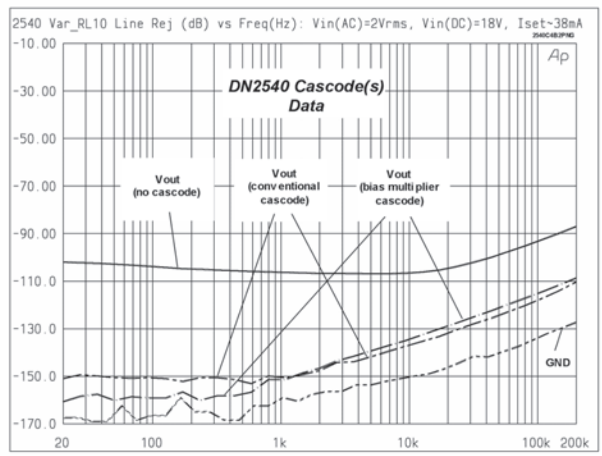


FIGURE 2: The performance of *Fig. 1* DN2540 MOSFET current sources for both conventional and bias-multiplied modes is excellent. With a new higher resolution setup, rejection of the conventional cascode is ≅ 150dB, while that of the bias-multiplier cascode is ≅ 160dB at low frequencies. For reference, a re-tested plot of no-cascode DN2540 operation shows performance comparable to original data (Fig. 12B from Sources 101 Pt. 2).

So, if the DC bias of M2 is, for example, 2V, the resistor ratio shown increases it to 4× this, or 8V. This multiplied bias then becomes the V_{DS} of M1. This higher bias should enhance the M1 rejection characteristics for input changes, as it moves M1 V_{DS} further into its pentode region. The capacitor C1 bypasses R3, for lowest overall noise.

This approach allows noiseless biasing of this cascode, i.e., selective biasing without the use of Vs dividers, or batteries. When I first tested the bias-multiplying cascode, I couldn't see any improvement vis-à-vis the conventional type (left). Both circuit forms showed performance virtually indistinguishable from the setup noise floor.

But, I had also made some further setup improvements, which have a direct bearing on measurable results for these circuits. Here's what I did to change the test setup from that previously described[1].

1. Raised the test circuit drive to the DUT to Vin(AC) 2V RMS, which adds 6dB of dynamic range vis-à-vis the original 1V RMS setup.
2. Changed test load resistor Rload1 to 10Ω, a step which adds 20dB more sensitivity to the Vout error voltage. To be valid, this step assumes that all Zs being measured are >> 10Ω, which they are in all cases thus far (in this DN2540 case, it turns out that the measured Z is on the order of 100MΩ or more).
3. Created new Audio Precision software to do a rescaling of the output data files, so that all measured data points in a given sweep are reduced by 20dB (the ratio of the Rload1 change, or 10:1). This effectively brings the setup calibration back to a 1Ω load reference. It also offsets the displayed dynamic range 20dB, downward. The total *display* dynamic range shown is still 160dB. It effectively adds 20dB more to the lower end of the measuring dynamic range, as the noise is effectively reduced relative to the Vout signal. In these plots, the 0dB reference trace isn't shown, but it is/was calibrated prior to a test run. The working test range was also calibration-checked at low levels, with a 10MΩ resistor, resulting in a nice flat line at –139.9dB (just about as it should be, with a measured -140dB being ideal).

The resulting test plots of **Fig. 2** are for the Supertex DN2540N5 TO-220 part(s) operated at 38mA, a current level that corresponds roughly to similar data[2]. It is easy to see the additional dynamic range here, which extends to –170dB. The plot which is repeated for the non-cascode 2540 operation is almost identical to the original 2540 plot at a 30mA level (Fig. 12B[2]).

A focus of this particular experiment was to discern whether the bias-multiplying cascode version (**Fig. 1**, right) really did make a difference in terms of rejection, vis-à-vis a more conventional version (**Fig. 1**, left), where the upper FET's gate is tied directly to the lower FET's source. Lo and behold from **Fig. 2**, yes, it does seem to make a difference! The rejection is better with the extra DC bias, just as theory says it should. The amount is about 10dB, not an overwhelming amount, but also clearly an improvement. This difference might change to even more between the two operative modes, were M1/M2 to see additional drain voltage. Here each sees at the most about 9V, which isn't very high for a 400V part. Interestingly, I couldn't see this differentiation at all, with the older setup and Rload1 = 1Ω. With the load of 10Ω, it is easy to see.

SELF-GENERATED NOISE

All of the aforementioned tests are relevant to synchronous noise of a current source circuit, which is noise that is related to AC components of the unregulated power source. But, another important noise type is *self-generated noise,* which is the form generated by a device (or circuit) with a steady DC bias. It is this type of noise that led to the relatively poor performance of the LM317 cascode circuit in the preamp stage. Generally this type of noise—white noise—is relatively constant over a wide range of frequencies, and typically manifests itself (audibly) as hiss.

In a test series planned for a future *audioXpress* article, I tested several different current source circuits for self-generated noise, including the LM317 and DN2540, with a test DC operating current of 10mA. I measured the noise components of the output current with a 100Ω resistor, an appropriate scaling amplifier, a 10kHz low-pass filter, and an RMS responding voltmeter.

To summarize in a few words, the DN2540 at 10mA came in more than 50 times lower in noise vis-à-vis the LM317, for similar conditions. Expressed in terms of noise density in pA/√Hz, the LM317 was over 1000 pA/√Hz, with the DN2540 less than 14pA/√Hz. Note: *These figures couldn't be called exact, in particular for the DN2540, where the noise is on a par with the input amplifier of the test setup. However, the main point here is simply that the DN2540 is many times lower in noise than the LM317.*

CONCLUSION

So, these two MOSFET cascodes can be termed "all-weather" current sources, because they can operate as either current sources or current sinks. They are also all-weather from the standpoint of a tremendous current range capability, from just a few mA up to 100mA or more. Plus, of course, there is the wide voltage range, up to 400V. Suitable for a wide range of solid-state or tube uses, in power or signal circuits, the **Fig. 1** configurations are the most versatile of any discussed. I recommend them without any major reservations. Although the TO-220 package of the DN2540N5 could be a downside for low power uses, there is a handy alternate, the TO-92 DN2540N3 (same internal chip).

ACKNOWLEDGMENTS

Henk Huitema, John Tracy and Ben Veltmaat provided useful feedback on their experiences using these current sources, within original correspondences taking place in Sept 2007. My thanks to all of them. In their final setup, they all reported successful use of the **Fig. 1** (left) cascode. This cascode type has previously been discussed[5].

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THE IMPASSE PREAMP

I really enjoyed reading "The ImPasse Preamplifier" (*aX* 2/09) by Stuart Yaniger. I particularly got a kick out of the section headings alluding to lyrics from one of my favorite rock songs, and to presidential campaign slogans as well. But I do have a few questions and comments.

The article lauds the low input capacitance that the medium mu 6SN7 affords a gain of 20 circuit. It cites 80pF with a 6SN7, and a whopping 300pF from the high mu D3a. That may be so, but I'd like to point out that the high mu 12AX7, with a 1.7pF grid-plate capacitance, would do better with this than either of the other triodes in a gain of 20 circuit. Its input capacitance would be only 34pF. If you don't like the 12AX7, there are many other high mu tubes with low grid-plate capacitance—the 6SL7 and 12AT7 come to mind.

The article also points out quite correctly that the CCS in the plate circuit helps to produce a gain stage with a terrific PSRR. The problem is that the phase inverter output stage has a lousy plate PSRR; it passes on just about the full noise of the supply. And this noise is strongly attenuated at the cathode, so there is no common mode cancellation by the differential F4 power amps. With the given inverter output stage, the best PSRR of the overall circuit would be obtained if the gain stage cut the power supply noise about in half.

Given a 350V DC supply, a gain stage of 20 with a plate voltage of 120V DC and a PSRR of 0.5 can be implemented simply with a 12AX7. Use a cathode resistor of 5k11, a plate resistor of 237K, and add a 392K resistor between the plate and ground. Tap off a grid bias about 1k5 from the cathode stage's grid. Advantages of this approach are lower input capacitance, a simpler circuit, and that only 0.22W are dissipated in the top plate resistor as opposed to 1.5W with the 6SN7.

While it is true that the AC voltages at these points would be identical, the impedances would certainly not be. The triode's cathode impedance is about 1/gm, while its plate impedance is much, much higher. (You might partially compensate for this by adding resistance in series with the article's C4, the cathode coupling capacitor.) But if you're going to match the inverter stage's cathode and plate resistors to 0.1%, you'll want to do the same with the input resistances of the F4s you're driving. Alternatively, you could compensate for your F4s mismatch in the inverter stage's resistors.

I have always thought that this tube should be considered as a possible audio standard. Its transconductance, mu and maximum cathode-heater voltage are higher than those of the 6DJ8, and it doesn't have the 6DJ8's unusual construction that is sometimes blamed for a strident sound. Is the 6KN8 still commercially available?

Christopher Paul
cpaul@suffolk.lib.ny.us

Stuart Yaniger responds:

I thank Mr. Paul for his interesting letter and the very cordial and positive correspondence that it triggered.

His comments regarding system power supply rejection are very well taken and I agree with them. As a practical matter, with the regulator shown, system signal-to-noise is better than -100dB with respect to 20V, so the point that I made in the article and his clarification are more academic questions. But indeed, it does show the need for a reasonably quiet supply or some means of canceling supply noise if this is not the case.

Indeed, it is possible to reduce the gain, and hence the Miller effect, of a high mu triode by loading down its plate sufficiently and adding some degeneration. Unfortunately, the side effect is decreased linearity—a triode is most linear when the load line is horizontal (constant-current), and decreasing the effective AC plate load swings the load line more vertical. In the case of the 12AX7, because the plate resistance of the tube is so high, the situation is exacerbated—with a CCS plate load, the tube is superbly linear. With a load resistor only two or three times the plate resistance, the modulation of plate resistance with current swing is rather severe. In a quick test of Mr. Paul's proposed circuit, linearity was degraded by an order of magnitude. I should note that the CCS shown in the article only uses about $4 worth of parts.

The question of source impedances of a split load is a vexing one which generates much heat, and it is important to note my stated restriction that plate and cathode be loaded equally. Much of the issue depends on what you use as the definition of "source impedance." If it is defined as the equivalent Thevenin impedance derived from voltage change when a test load is applied to the output, indeed both the plate and the cathode show the same source impedance as long as the restriction is honored. Identical test loads must be applied to plate and cathode. If this is the case, it may easily be experimentally verified that both sources are identical and are approximately equal to the reciprocal of the tube's transconductance.

Adding a series resistor to the cathode leg will ensure that, with equal test loads, the source impedances will be unbalanced and it will break the symmetry—the frequency response with identical reactive loads (e.g., cable capacitances) will now *not* be equal on both legs. A good mathematical derivation of this slightly surprising result may be found in Morgan Jones's *Building Valve Amplifiers,* 3rd edition, on pages 405-406 (available from Old Colony Sound Lab, www.audioXpress.com). However, Mr. Paul has given me the opportunity to issue a warning that I neglected to make in the article—the cabling from each phase to the power amplifier must be the same! If the cable carrying one phase is longer than the other, the reactances will be different and the frequency response and balance will suffer disproportionately.

On the matter of resistor matching, as long as the input resistors of the F4 are more than a decade larger than the plate and cathode resistors of the output phase splitter of the Im-Passe, their tolerances are a decade less critical. Though it certainly won't hurt to match them more tightly, standard 1% values will work fine.

The 6KN8 is a very interesting 6DJ8 variant. Its gain is higher, but its linearity is somewhat lower when used as a voltage amplifier. This, of course, balances out when used with 100% degeneration, as is the case here.

Again, I thank Mr. Paul for the most interesting discussion and for the excellent and interesting contributions that he has made over the years. I was delighted and honored that he would read my article and take the time to analyze and comment on it. *aX*

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