CORRECTION

Within the September 2007 *audioXpress Letters* discussion on pp. 58-60, there are a couple of problems, specifically in the letter from Marc Whitney and my reply.

On pages 59 and 60, the graphics between Figs. 1 and 2 got swapped in this letter, while the captions are correct as printed. Thus, if the graphics data from these two figures is swapped, everything appears in proper context. Also, the fonts within the schematic figure got lost somehow, so what should be "Ω" symbols show up as "W" (proving Murphy's law once more).

Walt Jung
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VOODOO ENGINEERING

I read with interest Mr. Kwinn's suggestion for a construction article about a power regenerator ("Xpress Mail," July '07, p. 43). I felt obligated to come forward and clear up one or two minor misconceptions Mr. Kwinn (and many other people) seem to be under.

Virtually all audio equipment operates internally on DC power. The power supply takes in the AC line current and converts it into clean DC power of an appropriate voltage or voltages to power the circuitry. During this process the power supply removes all trace of the AC (and any noise present along with it). If the internal circuitry is critical about receiving a specific operating voltage, a regulated supply is used so that variations in the line voltage don't affect the supply voltages. This also removes any variations that might result if the input line voltage actually varies (possibly due to other devices on the line).

The purpose of a power conditioner (or filter) is to remove some or all of the noise on the AC line. This serves to "help" a power supply that isn't quite well enough designed to do a good job by itself. Power supplies frequently benefit from a little help of this type. It also blocks any "unreasonable" noise—such as the surge from a lightning strike or motor transient.

A power regenerator goes one step further and "makes all new line voltage" for the equipment to run on. This allows it to ensure that the voltage remains constant and that there is no noise present. This level of control should *not* be needed if the attached equipment is reasonably well designed. The sole way in which a power conditioner or regenerator can positively affect the sound of the devices connected to it is to *remove* any odd noises or "muddiness" that might occur as a result of noise leaking through their power supply and "annoying" the circuitry—or by providing stable voltage to equipment that has poor regulation and must run off particularly bad lines. It should also protect them from damage by major transients, but that will not be audible.

Any power regenerator such as the "famous" PS model, which actually produces power *other* than the 60Hz AC sine wave (50Hz in Europe) on which all equipment is intended to run, is an example of "voodoo engineering." There is no technical reason for varying the line frequency to produce any beneficial effects whatsoever on properly functioning equipment. If anything, there is the distinct possibility of causing a degradation in performance or actual damage. [If a transformer is defective, and buzzes at line frequency, it is *possible* that changing the line frequency *might* reduce this while not otherwise damaging anything. It might also reduce the interactions between hum in different components—not by eliminating the problems, but by shifting one so they no longer "line up." Both of these situations, however, presuppose one or more defective pieces of equipment to begin with.]

If there is, somewhere out there, a piece of equipment that benefits from a power regenerator, it would almost certainly benefit *more* from a careful redesign of its own power supply—which would also probably be much less costly.

The only exception to this is with *very* powerful devices such as large power amplifiers. Because of the way most power supplies draw current, they draw most current during the peaks of the sine wave. Under some circumstances this can allow the power waveform to become distorted in such a way that less peak voltage is available to the device—which can limit the maximum power output. This only occurs with devices that draw high current (power amplifiers of several hundred watts per channel), under some power line conditions, and requires a regenerator capable of supplying 20A or so to correct—which is far outside the context that was suggested.

The best way to improve (possibly) the sound of small signal equipment is to provide them with clean, well-filtered power, and to upgrade their power supplies (if they are deficient) with better regulation and high-frequency filtering. Of course, the actual circuitry might be a candidate for improvement as well.

Keith Levkoff
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SUGGESTION

How about a survey comparing coupling capacitors? I'd love to set up a group of Hi-Fi legends in the same room using a controlled setup and swap in different coupling caps from Sprague Orange Drops to "Unobtainium" Jensen Silver/Gold Foils. The survey could rate various aspects on a scale of 1-10 and add comments.

Craig Myers
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CORRECTION

A while back (in May) an error was found in the silkscreen pattern of the circuit board supplied in the kits that Old Colony Sound Lab is marketing to match my sub article ("Simple Servo Sub 100/120 Modification," Dec. '06, p. 8). I found this when a customer who couldn't get his sub working correctly sent me his sub (at my request).

The silkscreen markings for diode D1 (a 5.1V zener diode) and capacitor C1 (a tantalum capacitor) are backwards. The silkscreen shows the cathode side (with the bar) of the diode and the [+] leg of the capacitor going to ground. The bar of the diode and the [+] lead of the capacitor both need to be connected differently from the way they are marked.

The anode (without the bar) side of the diode and the [-] lead of the capacitor should go to ground. In other words, just turn each of these around.

Diode D2 and C2 (same types as D1 and C1) of the board are labeled correctly. Also, the schematics both in the article and as supplied with the kit are correct—the error is only in the silkscreen pattern of the kit printed circuit board. Fortunately, building to the incorrect orientation will not cause any damage to equipment or board components—just remove C1 and D1 and put them back in the opposite way and all should be well. With these parts backwards, the sub will function somewhat, though it will sound really lousy!

Bill Waslo
www.libinst.com

CHIP UPGRADES

In response to Mr. LeBeck's letter in the May '07 issue ("DVD/SACD Players," p. 58), I upgraded my Philips CD 650 and built my own headphone amp. I had some good experience with OP275 and the new National Semiconductor LM4562 (both dual) in my old Philips CD 650 player. In my WNA MKII headphone amp I use AD843 (single OP) in signal path. Another good single OP is the National Semiconductor LME49710. My equipment consists of Sennheiser HD 600 and (preferred) AKG K 701 headphones, a modified Philips CD 650 CD player, and Marantz SA 7001 KI.

I plan to get some AD828 and AD827 to compare them.

F. Stockhammer
www.stockhammer.eu/hifi/

R.K. LeBeck responds:

This interesting letter puts me in mind of my need to (finally) investigate building a real headphone amp, as well as upgrading/replacing/supplementing my headphone collection (Sennheiser HD580, Beyer DT-990 Pro). I, too, had wanted (some few years ago) to investigate the OP275, but never got around to it. The LM4562 was first mentioned to me by a Silicon Valley friend sometime late last year—its datasheet is extraordinarily impressive and I do hope to try it out. To make things clear, my experiments with the AD827 [unity gain stable] and AD828 [minimum gain stability is 2] were provoked by my frustration with the sound quality of "affordable" DVD/SACD/CD players.

Years ago, I modified a Philips DAC960 for a friend [see the 1992 POOGE articles by Jung and Galo]—the ultimate op amp upgrades were AD811 [I-V conversion] and BUF-03 for the output buffer/filter. I later applied the same devices and some of the methods to a Philips CD-80 CD player (wholly-derived [??] from a Marantz-Japan design, it seemed built like a tank). I worked on these units several times between 1991 and 1998.

Both of these units use the same Philips chipset (16-bit DAC and digital filter) as the DAC960 and the CD-80 as well as other Philips/Magnavox (typically in the US) 4×-oversampling CD players (examples: Magnavox model numbers CDB460, 560, 582 and probably others unknown to me). I was particularly impressed by the performance of the AD811 during that period, and also by the BUF-03.

In 1998, I also tried the BUF-04 in a Magnavox CD player and it then seemed like a very good alternative to the BUF-03. However, I don't believe either item is in production anymore.

In 2001, I procured an Assemblage DAC2.7 with the ordinary op amps installed because the old "Parts Connection" had a sale on many similar items back then. I soon modified the DAC2.7 [an outboard DAC] with the op amps shown in its user manual's schematic: AD811 and OPA627A. In that unit, those devices definitely improved the audible performance—later in 2003, I tweaked it a bit more, using ideas I remembered from the POOGE articles by Jung/Galo.

Thus, I have never forgotten the Jung/Galo usage of the AD811 in 1992 (it's a high-speed video op amp and is a "transimpedance" amplifier requiring some attention in its use). Here, I might also mention that a friend (in 2004) was able to get good results with an AD812 (a sort-of dual version of the AD811 with slightly diminished specs, but also a "transimpedance" amplifier). He used it as a unity-gain buffer in an old Nakamichi CD player's buffer/output-filter stage—but had to follow the "transimpedance" amplifier's (easy to do in the unity-gain configuration) requirements.

With all this history behind me [use of the AD811], I eventually thought of applying the "normal" (i.e., they are *not* "transimpedance" devices) video-oriented op amps: AD827 and AD828 (they are dual versions of the AD847 and AD848).

Finally, I first applied the AD827 (with great difficulty, because of its size as a normal dual-op amp) to the Sony DVP NS500V SACD/DVD/CD player. The results were very gratifying and SACD discs finally sounded like I'd always hoped they would.

A bit later I used a SOIC package of the AD828 in a second and identical Sony unit and it worked just as well [the Sony unit's technical documentation showed that the op amp stage had a gain of ~2, thus making the AD828 a good choice], and was much easier to install (if you forget the problems due to size and handling of the much smaller SOIC chips which were what the Sony unit's audio PC board was designed for—and is typical of almost all modern equipment).

Finally, I'd recently heard of a low-cost universal disc player, the Oppo DV970HD DVD-V/DVD-A/SACD/CD/and so on disc player, which had begun getting attention in the magazines I normally read every month. Wanting to extend the experiment beyond the Sony players, I opened up the OPPO unit and saw that it used the same op amps as the older Sony unit [the ubiquitous 4558 dual op amp]. Fortunately, use of a test disc (CBS CD-1) and an oscilloscope made it easy enough to find the op amp I was most concerned with (the two front stereo channels) and thus replacement of the SOIC 4558 with an SOIC AD828 (sourced from Digi-Key) was reasonably easy enough to do.

Subsequent tweaking: replacement of the electrolytic coupling capacitors in the front stereo channels—in the Sony *and* the OPPO units—with the economical Black Gate; 16V rated substitutes (source-able from Michael Percy Audio and other suppliers) improved the sound quality even further (all of these new capacitors were bypassed with small—because easy to use in these modern units using so many relatively tiny electronic parts—Panasonic 50V rated polypropylene units from Digi-Key).

TUBE BOARDS

The article titled "Three New Prototyping Boards" (June '07, p. 38) was great to help DIY builders know what's available. My **Photos 1-3** show the JCO tube

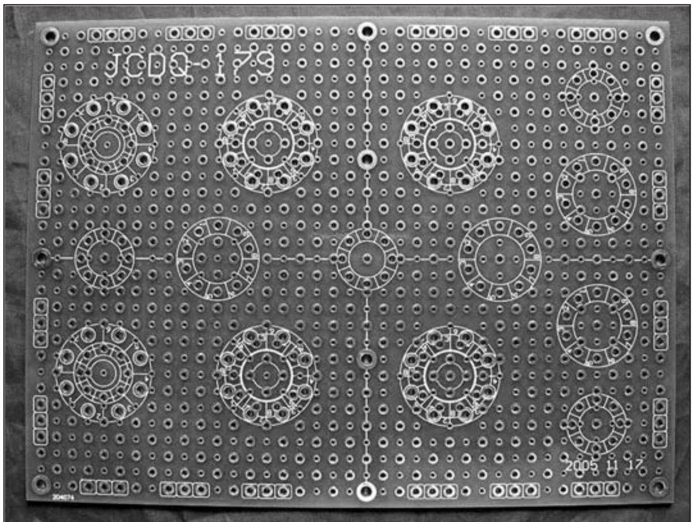

PHOTO 1: JCO tube prototype board.

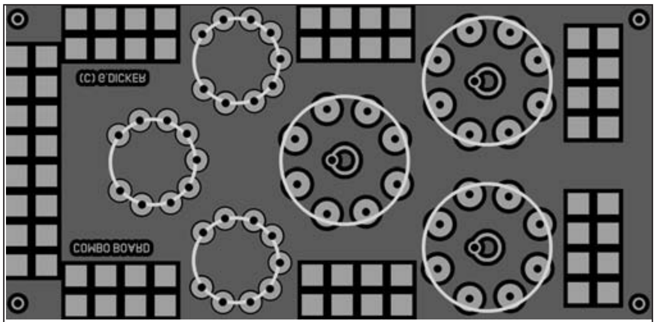

PHOTO 2: Author's protoype board.

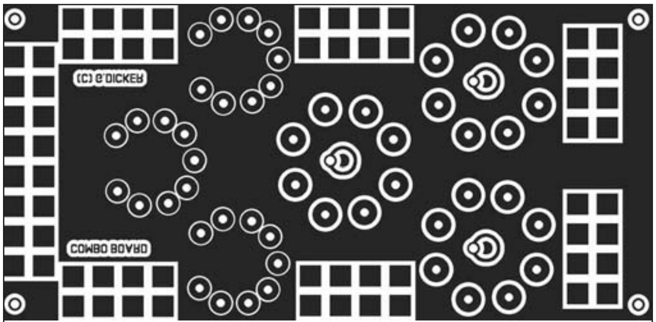

PHOTO 3: PCB pattern.

proto board available on eBay for around $25US posted, and my own proto board available for $15US including worldwide postage. The PCB pattern is also included for those who may wish to make their own.

Graham Dicker
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IC AMP UPDATE

In reference to Jack Walton's "High End 120W MOSFET IC Driven Amp," *aX* 7/07:

1. Unless the amplifier is powered from a 240V line, it is against NEC to switch both input lines. The grounded neutral on a 120V line may not be switched (or fused). UL will not list a device with a switched neutral.
2. How does the power supply provide ±56V from a 44VCT winding on T1? 88VCT would work.
3. The Fig. 2 schematic shows device U1 as a loudspeaker. The parts list identifies the LM4702 as U1. Where does the lead connected to R13, R3, L1 actually go?
4. Starting component ID with "1" on each schematic leads to confusion. I suggest that you sequentially assign each schematic a number. Component ID then starts with C11, C12; C21, C22, and so on. This also ties components shown on the silkscreen to the proper schematic diagram.

Charles Stillhard
Bisbee, Ariz.

Jack Walton responds:

1. As a ham radio operator for many years, I posed this concern to the technical experts at the American Radio Relay League in Newington, Conn. Reader Stillhard is correct. In 120V AC lines ("Hot Neutral Ground") the ground should not be fused or switched, which can pose a shock hazard. In 220V AC lines ("Hot Hot Neutral") both lines may be switched.
2. Each side of the center-tapped transformer I used delivers approximately 44V AC. After rectifying and filtering, and given the losses in the surge protectors and diodes, this averages out to somewhat less than the theoretical value you would expect.
3. I am sorry for the confusion with respect to the connection of the driver board and the output board. The U1 and U5 designations are the same device. While a loudspeaker schematic symbol is shown, in actuality this is just to illustrate where the speaker should be hooked up. The node that joins the "load sharing" resistors on the output board (R3-6) is the same one that conjoins L1, R1, R3, R13.
4. I lifted a page from the work of Erno Borbely ("New Power Amp Modules, Parts I and II," *Audio Amateur*, 3 and 4/93) and decided that a more flexible layout would use separate driver and output boards, allowing for bipolar transistors or MOSFETs in the final stage. A board for the output stage is not really necessary; just keep in mind, however, that the gate stopper resistors should be as close to the MOSFET gate as possible.
5. I apologize for the component ID numbering, but my CAD program automatically numbers components.

It seems there is an error in Fig. 2 of Jack Walton's article. I believe that C13 (the 470nF negative power supply bypass cap) should be connected to the side of R16 nearest the LM4702.

I don't build many projects, but I still find construction and design articles interesting and informative. I was wondering when people would start using the LM4702.

Keith Levkoff
kLevkoff@panix.com

Jack Walton responds:

As reader Levkoff correctly points out, capacitor C13 is connected directly to Vee pin of the LM4702. The connections are correct on the printed circuit board, incorrect on the schematic.

The Gerber Files for the driver board have been posted to the *aX* website (www.audioXpress.com) along with images of the top and bottom copper layers and silkscreen. These files and images are for Version 1.4 boards. The easiest way to get started is to use the images (which were produced at 100%) and print onto 3M transparency film.

Subsequent to publishing the article, I had some observations for improving the performance beyond the statistics published at the end of the article:

1. Before installing the RC network consisting of R2/C14 (330Ω, 10pF), check to see whether the amplifier actually oscillates. If there is no problem with oscillation, you can leave out these components and the THD% improves by a hair.
2. On the output stage schematic I showed 330pF capacitors (C1, C3) connected from gate to source of the output MOSFETs. If you eliminate these capacitors, the slew rate of the amplifier will decrease, but stability will be enhanced. Your mileage may vary depending upon the load characteristics and the precise configuration of the output stage.

In the amplifier I used for my day-to-day listening, C1/C3 are in place, but the R2/C14 RC network is disconnected.

After the article was published, Troy Huebner of National Semiconductor wrote an application note which is required reading: "*LM4702 Driving a MOSFET Output Stage*," AN-1645. In particular, Huebner demonstrates an output biasing scheme for the Renesas Lateral MOSFETs which simply consists of a 1K potentiometer across the source and sink pins of the LM4702 and a 47K resistor from sink to ground. He also uses 10pF capacitors (which I show as 30pF silver micas, C4 and C11 on the output stage) which "hot up" the slew rate of the AN-1645 amplifier. I have used this simpler bias scheme which works well.

Huebner further describes the method for explicitly determining the value of gate-stopper resistors. Use of other output devices, such as the Vishay IRFP240/9240, Magnatec BUZ901/906, and Toshiba 2SK1530/2SJ201 MOSFETs, are described in this application note. National promises an application note that will discuss the optimization of the LM4702 compensation capacitors. The LM4702 is soon to have several brothers and sisters in a large family of high-end driver devices.

Version 1.6 driver boards, the 2SK1058 and 2SJ162 lateral MOSFETs, and some of the ancillary bits and pieces are available on my webstore, www.tech-diy.com/LM4702_Kits.htm. I did not want to flog the product on *audioXpress* (and the PCBs were not yet available at press time). I promise to keep interested readers up to date on developments via my website. The amplifiers based upon the LM4702 are very listenable, and I am gratified by the enthusiastic response to the article.

METER MEASUREMENTS

I can confirm the 10dB peak of the Radio Shack digital sound level meter, in an interesting story. While working on my current speaker project (a sort of reincarnation of the JBL Paragon using a LE14A woofer and a LE85 compression driver on a home made tractrix horn), I received tremendous help from the JBL Forum (see thread: http://audioheritage.org/vbulletin/showthread.php?t=12420). I was having a problem with a large peak in the response of the LE85 at 6kHz, which turned out to be my RS meter, not the LE85. That reminded me of the previous speaker I built which *also* had a peak at 6k (measured by the RS meter).

I then produced a simple LCR trap with a rheostat for the R to adjust the peak and successfully "tamed" the 10dB peak to a reasonably flat response. The problem was that in listening to the speakers, especially on female vocals, both my wife and I preferred the speaker without the trap (with the 6k peak). At the time I found this curious, but built the final version to what our ears liked. Now with my new measuring setup (see above thread), I find the "ear tuned" speaker is very nearly flat. So much for measurement versus "golden" (or not so) ears!

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MAN AND MUSIC

After reading the review in *audioXpress* (7/07, p. 6), I ran out and bought *This Is Your Brain on Music*. When reading this book, I wished there were a companion CD with some of the sounds and songs that were referred to in the text. I just now discovered that the author has a website with these sounds: www.yourbrainonmusic.com.

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MUSICAL INFLUENCES

The review of *HiFiCritic* (July '07, p. 45) was disappointing, as was my rough impression from their website. I don't doubt the integrity of Mr. Colloms and their other reviewers, especially in light of their no ad policy. However, I question their ability to distinguish physically caused sonic differences from mindset influences.

Case in point: the "colouration" from the Quad speaker's illuminated logo. Peter Walker, if he can divert himself from enjoying the Great Concert Hall in the sky, will laugh his head off!

Circa 1957, *Life* magazine published an article on "synaesthesia," in which some people could supposedly "hear colors," "smell sounds," and so on—an "intersensory crosstalk" phenomenon. The lighted Quad logo certainly has *visual* coloration (unless it's pure white, of course). But a sonic influence?

Now, over decades of observing my mind and the tricks it can play (which the great writer Jiddu Krishnamurti said can be great fun), I've become well aware of the influence on musical perception of many factors, both mental (beliefs, and so on) and physical (my health, comfort, room lighting, and so on). Regarding the latter, it's no secret that dimming or darkening the ambient light enhances musical pleasure; concert halls and seductive lovers do this! The ears have less competition from the eyes.

I suggest that *HiFiCritic*'s reviewers

try evaluating the Quad light's "influence" when blindfolded (a new raison d'etre for "blind testing"). Then, if they hear no effect of the light, they could show some semblance of conducting scientific research by investigating the effect of visual distractions on musical perception!

Dennis Colin
Gilmanton, I.W., N.H.

MORE ON LM12 AMP

Thank you for printing Bill Christie's article describing the LM12 amp. I usually build tube amps, but the LM12 looks like an interesting first solid-state project. I have a couple of questions:

1. I haven't found a source for the special Vishay/Sprague 22,000µF caps. Can you point me to where I might order some?
2. Mouser has a MJ15015G and a NTE388 cross-referenced to a MJ15015. Would these parts be OK for Q1?
3. I wasn't going to bother building the clipping indicator circuit. I'm assuming it's not really needed. Am I wrong?
4. I found an AVEL toroid (made in Conn., where I live) that has a 40V secondary rated at 1000VA. I know with tubes this would be close enough. Except for a possible size difference, would this work in your LM12 circuit? Or would I need to change some circuit values? If so, I'll just get the exact part you recommend.
5. After I've built this amp, do you have recommendations for the first power-up, such as using a Variac, or key voltage checks? I was going to use dummy 8Ω loads, but do you have any other thoughts?

Thanks again for a very interesting article. And thanks in advance for your time.

Tom Ottman
Berlin, Conn.

Bill Christie responds:

Thank you for your interest in the article. I hope these responses will answer your questions satisfactorily.

1. I apologize for a typo in the article regarding the part number for the Vishay/Sprague filter capacitors. The part number should read 36DY223F0075CB2A, which is available from Mouser Electronics.
2. The cross-referenced transistors should work fine.
3. The clipping indicator is entirely optional. Sometimes I like bells and whistles.
4. The transformer I specified has two 38V windings forming a 76V center-tapped secondary. If the AVEL toroid is 80V center-tapped or has two 40V windings, it should work fine.
5. It's always good to use a Variac for initial testing and possibly a lower power supply fuse.

I just read Bill Christie's article "Cascoding National's LM12 for 140W." I am in Greece and am unable to read the whole article about this amp; I can only see the pdf files existing in your site.

From what I've seen, this design is much different than the original posted by National Semiconductor. Can you tell me why? Also, why does your design sound better? The recommended rail voltage in the National book is 100+100V. How much is yours?

A note about proper thermal design in the National book recommends keeping the tip 31-32 drivers outside heatsink. Maybe they don't require cooling, but is there a chance that they need thermal compensation?

Finally, the original article mentions a swing of 90+90V and output of ±10A. Do you think something like that is possible?

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Bill Christie responds:

The circuit that I have adapted for my design is taken from National Semiconductor's application note AN446 and is shown in Figure 11 of that publication. The circuit uses a higher voltage power supply, so I needed to adjust some of the component values in the cascode circuit. The only other change was to lower the gain for an inverting input buffer with higher than unity gain. I added the input buffer primarily to raise the input impedance. National's circuit has an input impedance of 1kΩ, which would be too low for many preamplifiers.

I redrew the schematic to make it easier for me to read, but is circuitwise identical to that in Figure 11 except for the changes in the component values just mentioned. The power supply that I used is ±56V. This gave me all the power I wanted. The National circuit shows a ±100V supply, which theoretically could output 900W of power. I assume it will work as advertised and don't see any reason why it wouldn't sound as good as mine provided the preamplifier can handle the low input impedance.

The TIP31 and TIP32 transistors are used with MJ15015 and MJ15016 in darlington configurations to increase the current gain. They dissipate very little power and do not require mounting on the heatsinks.

I very much enjoyed Bill Christie's article on the LM12 ("Cascoding National's LM12 for 140W," 8/07, p. 38) and would like to build several of these amps. I have not been able to find the artwork you mentioned as being available on **audioXpress.com**. Since you obviously have built a number of these and have the artwork, you also probably have a supplier to manufacture them. Why not offer them to other *audioXpress* readers?

I see several projects a year that require circuit boards, but it's difficult to find someone to make them. The setup costs for the first set can be prohibitive for small boards.

Bruce Brown
TuninFork@aol.com

Bill Christie responds:
I etched the circuit boards myself. I'm forwarding a 600 dpi file of the artwork to you and to *audioXpress* for inclusion on their website (www.audioXpress.com). I printed it mirror image on transparency film with an inkjet printer with the ink volume set to heavy and quality set to best. PC board material with a photosensitive coating is available from Parts Express. A 6″ × 9″ board will handle two channels.

BIAS SCHEMES

After reading Tim Smith's article on building a pentode amp ("Pentode Petite," *audioXpress* 6/05), I put it on my list of projects to build. When I came across a set of power and output transformers from a Harman Kardon A 300, I decided it was time to build the Pentode Petite.

The A 300 used 7355 output tubes and a voltage doubler's HV power supply, but they looked like they would work perfectly. The power supply I built used the voltage doublers connected to Smith's design for the rest of the HV section. It put out 335V. The filament supply paralleled all the tubes to use the 6.3V secondary of the power transformer. I built the bias supply as presented in the article with a separate transformer. It produced 24V unloaded, so everything seemed to be fine.

The sound of the amp was very nice, with no hum or oscillation, but I had trouble keeping the output tubes balanced, and the power and output transformers became warm after 3-4 hours of use, even at moderate levels. The bias voltage at the grid of the EL34s was –18 to -19V and the cathode current was 105-125mA (not even close to the 60-80mA specified in the article). I even tried several sets of EL34s and still couldn't solve the issue. The tubes ran very hot, but no glowing on the plates. I even tried using different secondary taps to change the plate load on the output tubes, all to no avail.

My friend Larry builds many tube amps and never uses fixed bias. He prefers the automatic bias setups for simplification and less maintenance, so I decided to change the pentode to an auto bias system.

The mods to do this are as follows:

1. Remove the complete bias system from the amp (T3, D2, D3, C12, C13, C14, R8, R9, LED 1, C15-18, PA1, and PA2).
2. Remove R12 and R13 from the cathodes of each of the output tubes.
3. The ends of R18 and R19, which used to go to the bias pots, now need to be grounded.
4. From the cathode (pin 8) of each EL34, connect a 520Ω 5W resistor to ground and parallel it with a 200-250mF 100V electrolytic cap (+ to pin 8).

You now have an amp with auto bias. So how did it turn out? The amp sounds every bit as good as it did before, but even after 8 hours of listening the power transformer just becomes warm to the touch. The output tubes run much cooler and the measured voltage at each cathode is within 1-2V of each other. I suspect the power output is lower than with fixed bias, but for most situations, this really isn't a problem.

I love building and modifying vintage audio equipment and encourage everyone who reads *audioXpress* to consider submitting articles. For me this is the best thing about the magazine. I very much liked Tim's article and think it is a superb amp with lots of neat and innovative features. My modifications are in no way due to a problem with his design, but an improvement that worked for me.

Bruce Brown
Valley Springs, S.D.

Tim Smith responds:
I'm sorry that Mr. Brown had difficulty with the bias supply with his version of the Pentode Petite amplifier. However, it seems that he has implemented a very useful cathode bias scheme that I would recommend as an alternative biasing scheme for anyone who wishes to build this amplifier. I have always used fixed bias because it essentially enables more power output and better balancing of the output tubes. However, in retrospect, I think that a cathode bias scheme may be a better choice for a novice project such as this one.

He noted that the voltage applied to the E34L grids was in the −18V range. This is insufficient and reveals that there may be a problem in the bias supply as he built it. The bias supply is based on a small Radio Shack 12.6VCT transformer which feeds a voltage doubler supply. Output from this supply, measured at the negative end of C14, should be about −34V with 120V supply.

R8, R10, P1, P1, and R11 form a series resistive chain which places the wipers of P1 and P2 at about -22 to −23V when both pots are centered (perhaps Mr. Brown inadvertently used higher values for R8 and/or R10 than indicated on the schematic). This voltage, with about 325V on the E34L screens, provides about 70mA idle current for each tube. If less idle current is desired, you can replace R11 with a 3.9 or 4.3k .5W resistor. However, this modification will slightly lessen the amplifier's gain and increase its distortion.

I'm glad to find out that a number of people have decided to build this amp or a version of it. Mine is now with a friend who uses it a lot and is happy with it. For those who are interested, I am currently working on an enhanced design that uses a custom power transformer which will let the amplifier use the full 30W capability of the output transformers. This power transformer, wound by Heyboer, probably costs less than the combined cost of the three power transformers in the original Pentode Petite design.

OP-AMP STAGES

Mr. Colin's article ("Optimum Stages for Minimal Distortion," Aug. '07, p. 50) appeared just as I was pondering whether or not an op amp gain stage ahead of an LM3886 (or other IC amplifier) might be a good idea. The application note for the LM3886 and similar devices shows sample circuits typically with a gain of 20, which might be a little less than desired for some applications. I had feared that adding another stage perhaps with a gain of 5 would degrade the sound more than simply changing the NFB resistor values to achieve a desired gain of 25.

Although Mr. Colin's focus was phono

amps (and probably equally applicable to line-level preamps), I believe he addressed my concern.

There may be an issue with which [the op amp or the LM3886] reaches clipping first, but that could be worked out. I would be interested to hear the author's thoughts on the matter.

I would also have liked Mr. Colin to include noise into the equation, because I can conceive of that being as much or even more of an audible concern than distortion. In any case, the article is much appreciated.

Marc Whitney
Phoenix, Ariz.

Dennis Colin responds:

Thanks for your interest. The LM3886 is a 68W audio power amp, and as such, its distortion is likely higher than a good line-level op amp such as the OPA 134PA (or dual unit OPA 2144pA). Consider that power amps generally have an optimum amount of NFB (too little might not attenuate low-order distortion enough, while too much can produce excessive high-order distortion, particularly with reactive load phase shifts).

Therefore, it's probably best to use the manufacturer's recommended amount of NFB—e.g., the closed-loop gain of 20 (26dB) that you stated for the LM3886—and then drive it with a good op amp whose gain you can set as desired. But if you need a gain of 25, which is only a 1.9dB increase over the gain of 20, the LM3886 shouldn't be compromised noticeably, if at all.

68W output power is 23.3V RMS into 8Ω; peak voltage is 33.0V. With an LM3886 gain of 20, then, its full-power input voltage requirement is 1.65V peak, 1.17V RMS. Now, while CD players tend to provide 2.8V peaks with "hot" recordings (2.83V peak or 2.0V RMS is the 0dB full-scale Red Book standard), many sources might not provide the 1.65V peaks needed for full power output.

So it appears that an op amp preceding the LM3886 would be desirable. The OPA2134PA (dual for stereo) has excellent performance: 0.00008% THD at unity gain, or 0.0008% with a gain of 10 (20dB); 8nV/√Hz voltage noise, negligible current noise (3 femtoamps/√Hz), 8MHz GBW, 20V/µS slew rate, and it costs only $2.63 from Digi-Key. Using this op amp offers additional benefits:

1. Non-inverting overall polarity can be restored if the power stage is (or performs better when configured as) inverting.
2. The power stage is driven from a very low impedance.
3. Ultrasonic and/or infrasonic rolloffs can be easily implemented.
4. Overall distortion can be lower because the power stage's NFB can be set to the optimum level. Note: I found the OPA2134PA's input and output sonically indistinguishable in an "intrinsic fidelity" test (see "Intrinsic Fidelity Testing," June '07).
5. Noise will likely be lower. In a two-stage amp, if both stages have the same net equivalent input noise voltage, and the input stage gain is 3.16 (10dB), the overall input-referred noise is only 0.41dB higher than a single-stage amp. *However*, here, the OPA2134PA (or OPA134PA) noise is probably *much* lower than that of the power stage. As an example, suppose the power stage's input-referred noise is 10dB higher than that of the op amp, and the op amp gain is set to 3 (9.5dB). Then the overall two-stage input-referred noise is 6.8dB *lower* than that of the power stage used alone.

This assumes that the op amp's feedback resistor is low enough to not contribute much added noise. With a 1k value (4.06nV/√Hz noise at +25° C), the OPA2134PA noise will be degraded by 1.0dB. With this, a non-inverting configuration, and a 499Ω resistor from inverting input to ground (for a gain of 3), the wideband unweighted noise will be about 103dB below full output signal.

These low resistor values require a non-inverting op amp configuration to allow a high input impedance (a 100k resistor from the non-inverting signal input to ground is good). Then, if the power stage is inverting, you can simply reverse the speaker leads to maintain correct overall polarity (absolute polarity can sometimes be audible because of the ear's asymmetric nonlinearity at high SPL).

I recommend an OPA2134PA input stage with a gain of about three, followed by a passive volume control, which feeds the LM3886. The low-z op amp output can easily drive a tapped volume pot with a loudness-compensation circuit (such as I used in "A High-Quality Solid-State Headphone Amp," Nov. '05). The result should be a very nice and inexpensive 68W per channel amp. *aX*

CONTRIBUTORS

Edward T. Dell, Jr. ("Paper vs. Screens," p. 6) is owner and publisher of Audio Amateur, Inc., publishers of *audioXpress*, *Voice Coil*, and *Multi Media Manufacturer* magazines.

Iain McNeill ("A Mains Analyzer–How Clean Is Your Juice? Part 1," p. 8), an avid reader since the days of *Speaker Builder*, resides in Aptos, Calif.

Dennis Colin ("Frequency Delay Dispersion," p. 16, and "Intrinsic Fidelity Testing," p. 36) has demonstrated the audibility of phase distortion at Boston Audio Society, and has designed the "Omni-Focus" speaker (bipolar coincidental with phase-linear first-order crossover), ARP 2600 analog music synthesizer, 1kW biamp and PWM supply at A/D/S, and Class D amps.

Stephen Moore ("A Hybrid Valve MOSFET SE Amp, Part 2," p. 26) has extensive experience in the advanced energy field from both the science and business perspectives. He has written multiple patents relating to control theory and the energy field and is widely published on topics of lithium battery systems. He is a registered expert in the International Electrotechnique Commission (IEC) regulatory and standardization body SC/TC21A.

Roger Russell ("Sounds and Hearing, Pt. 2," p. 32) has a degree in electrical engineering from Rensselaer Polytechnic Institute. He has written several articles about loudspeaker design, testing, and other audio devices. Several patents have been awarded for some of his designs. Although retired now, he was a member of the Audio Engineering Society and the International Society for General Semantics for many years. His interests range from stereo sound to photography, new-age music, mystery clocks, and science fiction. He has a web page at *www.roger-russell.com*.

Charles Hansen ("Tube Imp Update," p. 40) holds five patents in his field of engineering and enjoys modifying guitar amplifiers and effects to reduce noise and distortion. With hundreds of articles to his name, he is a frequent collaborator to the magazines of Audio Amateur Publications.

Rickard Berglund ("AUDIO AID: Build A New Tone-Contol Ciruit," p. 42) has done research in analytical chemistry and industrial electronics design in Sweden. He is now engaged in construction and manufacturing of tube amplifiers.

The audio hobby for **Alex Arion** ("Classic Circuitry: Greek Style," p. 52) began in the magic '60s in Communist Romania in his grandmother's attic, where, together with two friends, he installed a small "illegal" laboratory. Their first attempt was a beginner's guitar amplifier and a professional guitar pickup. The tubes used included ECC40s and EF40s. The rectifier was a 5µ4, Russian equivalent. They mounted all in a hat box that they found in the attic. After finishing his studies and many years of research, Alex moved to Greece, where he continues to build different tube-based and solid-state amps.