Back

GREAT SCOTT!

Charles Hansen's extensive refurbishment and modification of his Scott 222C (May–Sept. '06) is a wonder and a great credit to him. The finished job looks just gorgeous and I bet the sound is vastly improved.

However, the task looks most daunting. Can he tell us about how many hours he put into this work? I have a 222C, but I'm not sure I could last the distance!

Bart Shepherd
Epping, NSW, Australia

Charles Hansen responds:

Thanks for the compliment, Mr. Shepherd. I have no idea of the man-hours I put in. I started in November 2004 and finished in October 2005, but it was only a part-time effort. Much of that time was spent waiting for parts, manuals, investigating design changes, computer simulations, and so on. If you want to do the minimum amount of work and just restore the 222C to original working condition, I suggest the following:

- Check all the solder joints and touch up those that need it.
- Replace all the ceramic-tube paper American Ceracaps with modern polypropylene film caps.
- Replace the selenium bias supply rectifier with a silicon rectifier (mandatory!).
- Replace the aluminum can capacitors with modern versions made by CE manufacturing.
- Replace any obviously overheated resistors and wax-coated capacitors.

PUSH-PULL AMP

I greatly enjoyed the tube construction article in the August issue by Claudio Rosada ("Tuono: PP PL504 Amplifier," p. 18). Although this is a carefully designed, meticulously tested project, I did find one error.

Mr. Rosada's ideal output transformer would load each tube with 1200Ω, making the total plate-to-plate load 4800Ω, not 2400Ω, as stated. The reason for this is that transformers follow a square-law relationship for impedance. A push-pull transformer has twice the number of turns as a single-ended one, so the plate-to-plate impedance is 2 squared, or four times the impedance of half of the winding.

In other words, each tube sees one-quarter of the plate-to-plate load impedance. Mr. Rosada's actual transformer (2750Ω) will present a load to each tube of 687Ω. This is lower than the target value of 1200Ω, but is not too serious, as the ideal load should be about twice the plate resistance which, for a PL504, is about 350Ω. The actual measurements seem to vindicate Mr. Rosada's design.

Also, the PL504 is somewhat hard to find in this country. The electrically similar PL500/27GB5 is inexpensive and readily available, but I don't have detailed specifications. Does Mr. Rosada have any experience using this tube?

Bob McIntyre
musitronix@yahoo.com

Claudio Rosada responds:

In order to answer Mr. McIntyre, I must briefly review the theory of push-pull amplifiers.

First of all, you must consider the total dynamic load applied to each tube in the push configuration; in fact, from the perspective of one primary winding, there are two "secondary" windings. The first one is the real secondary winding, and the second one is the other half of the primary.

So the dynamic load seen by one tube will be the parallel of the secondary load transferred to the primary and the dynamic load connected to the second primary winding. The final result will depend on the load connected to the other primary winding.

You can consider two specific types of this load (other possibilities are less interesting):

1) The load is another tube with the same bias current, driven by the same input signal in opposite phase.

2) The load is a DC current source and its value equals the bias current of the tube.

The first case corresponds to a push-pull stage polarized in class A. It means the bias current is sufficiently high that each tube never reaches the cutoff condition when the maximum signal is applied to the grid. In this case the signal path connects the two tubes "in series." Together they see a total load corresponding to the plate-to-plate load, so each tube will see just one-half of this load.

The second case corresponds to another two possible applications of a push-pull transformer:

a) A single-ended stage is driving one primary winding, while a current source (e.g., a pentode) is connected to the other primary winding in order to perfectly balance the DC current. With this configuration you can use a push-pull transformer in a single-ended application.

b) Two tubes are connected in a push-pull stage polarized in class AB1. For low signals the stage works as previously stated, but when the input signal exceeds a certain limit, one of the two grids is so negative that its tube goes in cutoff and becomes dynamically an infinite load. The tube bias point will define the signal level at which this condition will occur, and from now on the dynamic load drops to one-quarter of the plate-to-plate load.

Tuono is polarized with high current in class AB1, so it works in pure class A for a big part of the total signal dynamics. For simplicity in my article I was referring only to this condition.

As long as the tube stays in the A region, the effective load is one-half of the plate-to-plate load, while when one tube reaches the cutoff, the load effectively drops to one-quarter of the plate-to-plate load.

In **Figs. 1** and **2** the output signal (upper track) on 8Ω load and the cathode current (lower track) of one tube are shown with a 1kHz input signal at different levels. **Figure 1** corresponds to 1W output power, and a perfect class A is visible. In **Fig. 2**, corresponding to 6W output power, the cathode current approaches the cutoff condition and is highly distorted. In this way the class B operation is used to provide an additional power boost during high peaks, while in normal condition both tubes are working.

The push-pull amplifiers theory is completely detailed in *Radiotron Designer's*

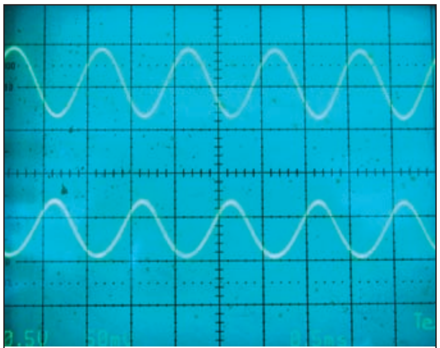

FIGURE 1: Tube measurement at 1W.

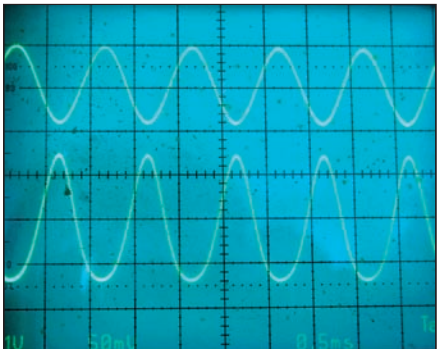

FIGURE 2: Tube measurement at 6W.

Handbook Fourth Edition, cap. 13.5, "Push-Pull Triodes Class A, AB1." PL500 data sheets are available in Frank's database at: http://frank.pocnet.net. PL504 looks to be an evolution of this tube. PL500 has a Wa power limit between 12W and 17W (the same values for PL504 are 16W and 22.5W), so you can expect a consequent reduction in output power when used in a similar circuit. I never used this tube.

TEST GEAR

The article titled Mad Katy by Dennis Colin (June '06) contains oscilloscope images showing tone burst responses. I am interested in finding out what equipment the author used to generate the tone bursts, and would appreciate finding out the manufacturer and model of the equipment.

Tom Russell
Richardson, Tex.

Dennis Colin responds:

Thank you for your interest. I used an HP8116A pulse/function generator (vintage about 1980, when HP was HP!). I've seen used ones for sale, unfortunately for over $1000. It has a gate and a trigger input, which allow clean tone bursts to be set up, that start at the sine wave zero crossing, then end after a number of cycles determined by the pulse width feeding the gate input.

The 8116A is expensive (new about $40,000!) because it covers one cycle every 1000 seconds, to 50MHz. You should be able to find a sine wave generator covering the audio range, with a gate input that's zero-crossing synchronized, for much less cost. Also, you could use an analog switch IC to pulse any sine wave source from any pulse or square wave source. To make the tone burst start and stop at the zero crossings, the pulse would need to be triggered or synchronized from the sine wave's zero points. An open-loop op amp would turn the sine wave into a square wave (or use a sine/square wave generator); the square edges could then be used to trigger the pulse generator.

I can't afford expensive equipment—I was lucky to find a sale of the 8116A, the Tek 475 scope, and more for about 2% of the original value.

TRANSFORMER MECHANICAL NOISE

While I am an avid reader of *audioXpress*, I don't do all that many construction projects. When I do undertake a project, I've found that the most difficult part to source is usually the power transformer. While it's easy enough to choose a unit that will do the job, there is no way to judge how much mechanical noise it will generate.

Many manufacturers use epoxy filling and other methods to "limit mechanical noise," but the only way to actually determine which units are noisy is to buy them and see (listen). To me, nothing ruins a nice quiet amplifier worse than a transformer that hums so loudly you have to shut it away in a closet. I would love to hear from other readers about their good (and bad) experiences with noisy transformers—especially from the vendors who advertise in *audioXpress*.

Keith Levkoff
kLevkoff@panix.com

CORIAN SPEAKER?

I would like to congratulate Peter Smith on a well-executed design and clear, thorough write-up ("NeoNoir: A Ported Two-Way," 9/06 *aX*). Not having enough time to work on my own designs, I enjoy living vicariously through the adventures of others; I found myself agreeing with just about every one of Peter's design decisions.

The Corian baffle was a tease though. Ever since remodeling my kitchen with the stuff, I have wanted to build a Corian speaker cabinet. Being the rock equivalent of MDF, it seemed that it should be denser and more passive than wood products.

Could Peter give us some details on any audible improvements with the Corian baffle? What tools were used—are carbide router bits OK for Corian or do you need special cutters? How thick was the Corian lamination? Any special finishing for the rounded edges?

iain@impaudio.biz

Peter J. Smith responds:

Thank you for the kind words on the article. Unfortunately, I did not do the millwork on the Corian baffle. A speaker builder friend kindly offered to cut the baffles for me out of sink cutouts he had lying around. I do know there are a number of special blades available for Corian and people have mentioned slow cutting speeds. I also know there is finish work starting with 220 grit wet paper and progressing to rubbing compound.

I did some basic drilling into my baffles and found the Corian brittle with a tendency to chip around the entry and exit points of the drill. Placing masking tape over the holes and drilling through the tape helped. I also struggled with scratches whenever hard tools got near the Corian. I would head to the Internet for specific guides on working with Corian.

As for sound and stiffness, the NeoNoir has the Corian glued with Liquid Nails (flooring) to a ¾″ piece of MDF. This makes for a stiff composite baffle, which always helps sound quality. I would not build an entire speaker cabinet out of Corian, because I would not trust the glue to hold, plus Corian is only ½″ thick. The harder and thinner material may actually be more excitable than regular MDF. So, the bottom line is that Corian was a nice experiment, but I am moving back to wood.

PREAMP ERRORS

The preamp construction article by Cornelius Morton in the September issue of *audioXpress* has four errors that I have found. There may be more, but I am not experienced enough to have seen them. The Figure 3: Power Supply Schematic, and the Figure 5: Front and Rear Panel Layout, are switched from their accompanying diagrams. On page 6, under Line Stage: "The output of V2A is coupled through C7,..." should be C8. On page 7, center column: "A full-wave bridge rectifier, D1, rectifies the 6.3V AC, R1 limits..." should be R5.

I am also interested in power-supply changes the author might suggest were I to build this as a high-level only preamp, eliminating the phono stage tube requirements.

Steve Perlmutter
stevis@chilitech.net

Cornelius Morton responds:

Thanks for your interest and inputs. Yes, the captions for *Figs. 3* and *5* are reversed. I noticed the same; links between the images and caption are correct.

C7 should be C8 as you indicate. The bridge diode limiting resistor should be R5, not R1, which was properly identified earlier. Concerning power-supply changes required to run the line stages only, they are relatively simple.

B+. Delete C11 and R21 for both channels. The voltage will increase a couple of volts but this will not be a problem.

Filaments. You may need to adjust R5 but the testing procedure covers this. I would expect that R5 will be 0.47Ω.

The line stages on this preamp are easy to listen to. Distortion is low and increases linearly with level as noted in the article. The distortion curves taken for different frequencies are virtually identical. There are no bumps or discontinuities in these curves that would tend to grab the ear. I have recently run a frequency analyzer on this. The output at 0.7V RMS indicated 0.15% second harmonic with a third harmonic visible at 30dB below the second. Nothing else was visible.

TUBE PREAMP TIPS

As an avid reader and author dating back to *Glass Audio* days, I read Cornelius Morton's article (*aX* 9/06) with interest. It is a very well-written and thorough article, even detailing the construction details.

Because I have built and tested many of these valve phono preamps, I have a couple tips that will even further improve the preamplifier design:

1. Operate the first stage with a grounded cathode. Check out the curves (http://tdsl.duncanamps.com/pdf/vm322.pdf), note the improved linearity when operating at zero volts! Operation with the grounded cathode has two advantages; first it is simpler, eliminating the (cough, cough) electrolytic capacitor in the signal path, and second, the gm is maximized while the plate resistance and the resulting gain is maximized. The result is better sound and lower noise, and you save money to boot!

2. Add a bleeder resistor to the output. 330-470K is fine.

3. Add local bypass of film capacitors at the plate resistors.

4. For the second stage it is better to direct-couple to the first stage. It eliminates parts in the signal path to improve the sound. From the curves you see that the first stage is going to run at about 35V. If you wish to run about 10mA in the second stage, the Rk is 3.3k (getting rid of C4 and R7). Then you set your first zero for the bass rolloff with C5 at 30Hz = 10-15µF. Use a tantalum or low ESR electrolytic

for this; 50-100V rating. Again bypass with a 1μF film cap.

David Wolze
dwolze@comcast.net

Cornelius Morton responds:

Thank you for your interest and comments. I have a few comments regarding your suggestions which I will try to keep in the order they occur.

Grounded cathode operation, zero bias. This would raise the possibility of a bit of grid current on heavy passages, which would be mitigated by the relatively low DC resistance of a moving magnet cartridge. The operating point of 35V DC Ep and 10mA Ip reflects a gm of 11500 S with an equivalent noise resistance of 222Ω, Rn = 2.5/gm, and a best case noise floor of -85.3dB referenced to a 5mV input signal. For a gm of about 9900 S the equivalent noise resistance is about 254Ω, resulting in a best-case noise floor of –84.5dB referenced to a 5mV input signal. The measured noise was better than –80dB with the phono input open and measured at the preamplifier output; shorting the input might gain a dB. Actually, I thought that getting that close was not too bad. The best noise floor would be –88.5dB, achieved by paralleling the two sections of the 6922 at an Ep of 35V DC and a total Ip of 20mA, though for safety I would still like cathode bias.

Eliminating the electrolytic. Current electrolytic design maintains capacitance well into the megahertz region and shows low hysterisis losses when the polarizing voltage is large compared to the signal voltage as is the case here. With the low signal levels involved, I believe that the differences in distortion would be very small and offset by the self-bias correction effects for tube aging.

The bleeder is there in the form of the 250k phono level control.

You could add local bypass caps, but again the current crop of electrolytics exhibit low levels of inductance and maintain capacitance over a wide range of frequencies. Looking at the first phono stage, the plate load R2 and C11 form a voltage divider from the plate to ground for the AC signal. R2 has a value of 8200Ω, while the impedance of C11 is only several ohms at best and usually much lower. For a plate signal of 1V AC the resulting voltage across C11, at an Xc of 2Ω, would be about 0.25mV. If the impedance of C11 increased by 100% that voltage would only increase to 0.5mV. In either case it is good filtering.

DC coupling between the two phono stages. I considered and then discarded this for several reasons. To maintain plate voltage for the second stage, the source voltage would be increased by the plate voltage of the first stage—35V DC for your suggestion and 60V DC for the design. This would complicate the power supply and make it nearly impossible to provide sufficient positive bias to the filaments to prevent filament to cathode conduction without exceeding filament to cathode voltage specs or really complicating the filament circuits. Your desire to replace C4 and R7 with a tantalum C5 is not good. Tantalum caps are about the most nonlinear caps made by man.

All resistors in this design are 1% metal film and all non-electrolytic caps are 5% film except the caps in the RIAA compensation circuits which are 3%. The values and accuracies of the RIAA circuit components are critical in maintaining a ±0.25dB accuracy in the RIAA compensation. A good reason to keep C4 and R7.

One way of slightly decreasing distortion and decreasing cost would be replacing all of the line-level controls with 58k resistors and the phono level with a 470k resistor. Then move the volume control to the input of the first line stage with it being driven with the selector switch output. This would require volume adjustment between sources which was not my original intent.

Thanks again for your interest.

BASS RESPONSE EQUALIZER

I enjoyed, and concurred with most of, George Danavaras' article on low-frequency electronic equalization of rooms (Aug. '06, p. 36). A couple of comments:

The equalizer seems to assume a monaural signal source. This source might be a simple passive mix of right and left line-level channels, or maybe is already a dedicated subwoofer output from a preamp or receiver/amp. In any case, there should be an awareness of the significant loading of some sources by the rather low impedances at the input of the equalizer (R1, R2, C1, C2) which for many line-level sources would attenuate the high frequencies in the source signal—desirable for the sub channel but not for the signals potentially used for the full-range ones. If the input is a dedicated LF channel, then you're likely alright—but then the high-frequency signals are probably not present to begin with.

He states that the low-frequency cutoff due to C3 is "about 1Hz." I calculate about 18Hz (-3dB) due to the loading by R3 and the attenuator network in series to common, a mite high for a subwoofer. If you moved C3 inboard a bit and placed it between the attenuator taps and the 160k, R40, the -3dB point would instead be at about 3Hz.

When I read "self-filter" and "self equalizer," I thought for a moment I'd missed some eponymous trick circuit of Doug Self's—but inspection of the schematic revealed that it should read "shelf equalizer."

The phase change circuit can indeed be useful, but you should understand that, for that simple circuit, the phase shift is frequency-dependent—so the description given of phase versus potentiometer setting is correct at only a particular frequency. As well, for sufficient generality, I'd recommend the inclusion somewhere in the chain of a polarity inversion capability—although, if you have ready access to them, flipping the wires at the sub driver itself could do the job.

Or, for that matter, if you are not concerned with absolute polarity, you could flip relative polarity by reversing the connections at the full-range speakers.

The one-op-amp synthetic inductors are tried-and-true topologies, although they do tend to be a bit noisy as the R4/R3 ratio becomes large—typically this will not be a problem for low-frequency processing because the noise bandwidth is so small. Among other options, circuits based on GICs (General Immitance Converters) use two op amps per inductor but if done right have better noise performance.

Also, when concatenating sections of such *dip* circuits, with a lot more work you can often eliminate the buffers separating the sections, reducing the number of op amps in the signal chain.

All in all, the proof is in the listening, and supported by the helpful measurements. Thanks for all of the hard work, George!

Brad Wood
Canoga Park, Calif.

George Danavaras responds:

I would like to thank Brad Wood for the interest in my article and the kind words.

Concerning his comments: It is true that I use the equalizer for a dedicated subwoofer, which is an open baffle design as described in the *audioXpress* article ("Double Dipole Subwoofer," November 2004). This is why the −3dB frequency for the low-pass filter which is realized with R1, R2, C1, and C2 is about 10kHz (supposing that the output impedance of the previous stage is about 100Ω). The low-pass filter at the input is also used to reject any possible EMI signals due to the long cable used to connect the equalizer to the system. When the equalizer is going to be used for the full audio spectrum, then the C1 and C2 should be decreased to about 470pF or 1nF.

Your calculation for the low-frequency cutoff due to the high-pass filter forming with the capacitor C3 and the attenuator network is correct. Initially I used a value of 4.7μF for C3 which gives a cutoff frequency close to 1Hz. I used the value of 330nF when I installed the open baffle subwoofer in order to eliminate the "hard drive" at the very low frequencies.

The correct expression is "shelf-equalizer." I am sorry for the confusion.

Although I included the phase change circuit in the design of the equalizer, later I found out that the use of such a circuit at least in my listening room is limited. So usually I bypass this circuit. Maybe it is more useful in other cases.

Finally, I don't measure any noise at the output of the equalizer that is due to the inductor realized with the op amp.

HEAD AMP DESIGN

I read with much interest the article by Paul Rossiter ("A Head Amp for Very Low Impedance Moving Coil Cartridges" *audioXpress* 9/06, p. 22) about the design of a high-performance head amp for moving coil cartridges. I have a few questions regarding the design and implementation.

1. I have not been able to find a source for the 2SC3329 and 2SA1316 transistors. Can you suggest a source for them?

2. Mr. Rossiter performed some performance analysis of the amplifier using SPICE. I have been unable to find SPICE models for these devices (also for the MPSA12 and MPSA62 Darlington transistors). Is there a way to obtain these models? I have access to SPICE, and if I have the models I can analyze the effects of some changes I have considered.

3. One design change that I would like to model (and on which I would solicit Mr. Rossiter's opinion) is to replace the packaged Darlington transistors with a pair of complementary Darlington transistors (also known as a Sziklai pair) constructed from pairs of the 2SA1513 and 2SC3329 transistors. This complementary Darlington pair is constructed from an NPN and a PNP transistor as opposed to two devices of the same type.

An advantage of this configuration is a lower Vbe drop (only one base-emitter junction in the compound base-emitter connection versus two for the MPSA12 and MPSA62 transistors). This will result in better Vbe matching over temperature changes than the standard Darlington connection. This approach will require sorting and matching transistor pairs, but may result in improved performance. A concern here would be potential instability due to increased parasitic impedances caused by the compound connection of the two devices.

4. I would also like to explore the addition of a second parallel transistor to each of the input devices (thus reducing the effective value of rbb). While theoretically capable of a 3dB improvement in input noise, I believe that because the parallel transistors are unlikely to be perfectly matched, the effect will not be as great as 3dB. Mr. Rossiter discusses this as a possibility with the potentially adverse effect of higher distortion; therefore, I would be interested in analyzing it before changing the design. I also believe that the values of R5 and R6 would need to be adjusted to change the emitter currents in the parallel pairs of transistors. Again, Mr. Rossiter's opinion would be of interest.

5. One last (small) point is that the schematic of Fig. 7 in the article seems to have the transistors in the MPSA62 Darlington pair (Q4) incorrectly connected. The two base-emitter junctions should be in series, not parallel, and the collectors should be in parallel, not series, I believe. This is a small matter because the devices are packaged together in one package. It is mostly a cosmetic concern.

Wayne Torrey
waphylz@cox.net

Paul Rossiter responds:

I thank Wayne for his interest and comments on the article. I will try to address each of the points he has raised.

1. I obtained the 2SC3329 and 2SA1316 transistors from Erno Borbely (who will be very familiar to the readers of *audioXpress* for his many excellent FET-based designs). I have checked with him and he still has a number of these parts available (sales@borbelyaudio.com).

2. The SPICE package I used (Micro-Cap 8) had the models for the 2SC3329 and 2SA1316 already included in the model library, so I just used those. For the MPSA12 and MPSA 62 I used the model for the MPSA77 also from the Micro-Cap 8 library, with only the gain and polarity parameters changed to match the MPSA12/62 datasheet values.

Unfortunately, due to changed circumstances since I wrote the article, I no longer have access to the library and so cannot provide the actual model parameters. However, as I noted in the article, the (audio frequency) distortion results from the simulation were not particularly dependent upon the actual model used for the first transistors and so anything close, suitably adjusted for the rbb′ and current gain of the 2CS3329/2SA1316 (given in *Table 1* in the article), should suffice. Similar comments apply to the Darlingtons.

3. Replacing the Darlington transistors with complementary feedback pairs (Sziklai pairs) is an interesting idea because the 100% local voltage feedback is then applied around the pair rather than each transistor separately. In amplifier output stages such an arrangement offers better large signal linearity than the Darlington equivalent and also better thermal stability because the base-emitter voltage drop of the output device is then within the feedback loop and only the base-emitter voltage drop of the driver affects the quiescent current (and this can easily be compensated). In the present application thermal stability is not an issue and the small signal distortion seems to be totally insignificant (and is certainly way below the noise floor). But please don't let these comments stop you from further experimentation (and the pursuit of perfection), and I would be keen to hear the outcome. I imagine that the high-frequency domain might need some attention to ensure that there is no instability.

4. I did discuss using paralleled transistors for Q1 and Q2 with the attendant lowering in noise contribution. However, in this design the noise ultimately becomes determined by the feedback resistor (R2 in *Fig. 6*), and so the gain in overall noise performance to be obtained by paralleling the input transistors is rather limited (remember, there are already effectively two transistors in parallel in the design). As discussed in the article, you can obtain a 1.5–1.8dB reduction in noise by reducing the 10Ω feedback resistors R2/R3 to 4.7 or 3.3Ω. However, the attendant reduction in other impedances required to maintain the gain leads to an increase in distortion, and so (as is usually the case) there is a tradeoff to be made.

5. Wayne is quite correct about the error in the symbol for the MPSA62 (Q4) in *Fig. 7*. That will teach me to not simply use a library symbol that looks about right! However, as he says, it is of no dramatic consequence because the elements are (I hope!) properly connected within the package. *aX*

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