Burning Amp Number Three

by Nelson Pass

Introduction

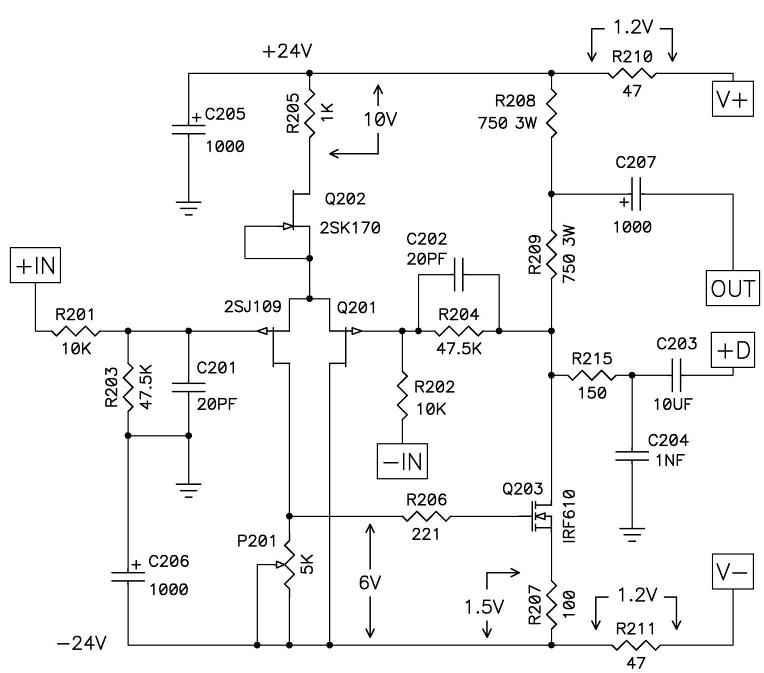
In the BA-1 and BA-2 projects we constructed two different amplifiers using very similar input and voltage gain stages (aka the “front ends”), but used them to drive two different Class A Mosfet follower output stages. Now we are going to flip this approach and begin exploring some different front end designs capable of driving these and other follower output stages.

Breaking it up this way allows some development focus and speeds things along. Those of you who have or plan to build an amp with these (or similar) follower stages should be able to follow along with minimal additional effort – you will already have the chassis, power supply, and output stage.

I intend to follow this approach in future pieces – with the power supplies being independent from the output stages being independent from the front ends. Of necessity the output stages will be follower types – this is usually where we get the best performance from simple circuits anyway, and after a while I hope we will have a small library of modular pieces that can be mixed and matched.

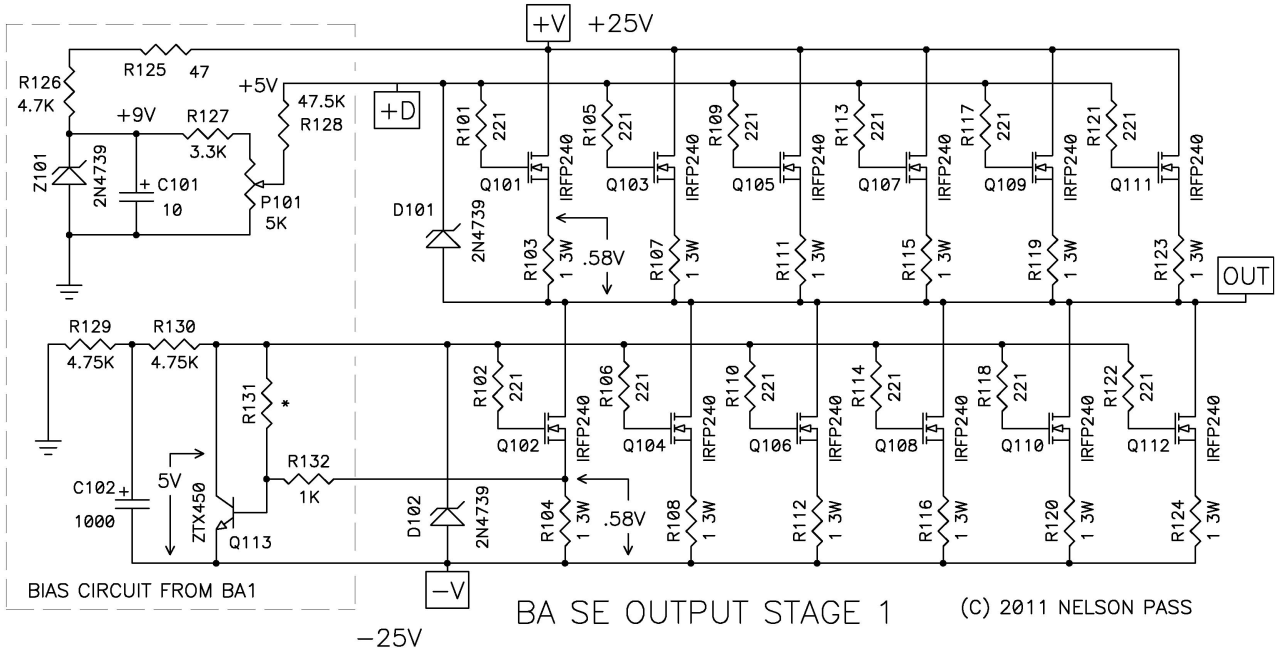
To avoid confusion, I will be presenting the schematics of the BA-1 and BA-2 in stand-alone form here, and from now on the BA series will likely refer to interchangeable subassemblies.

So first, some housekeeping. The official BA-1 and 2 front end now looks like this:

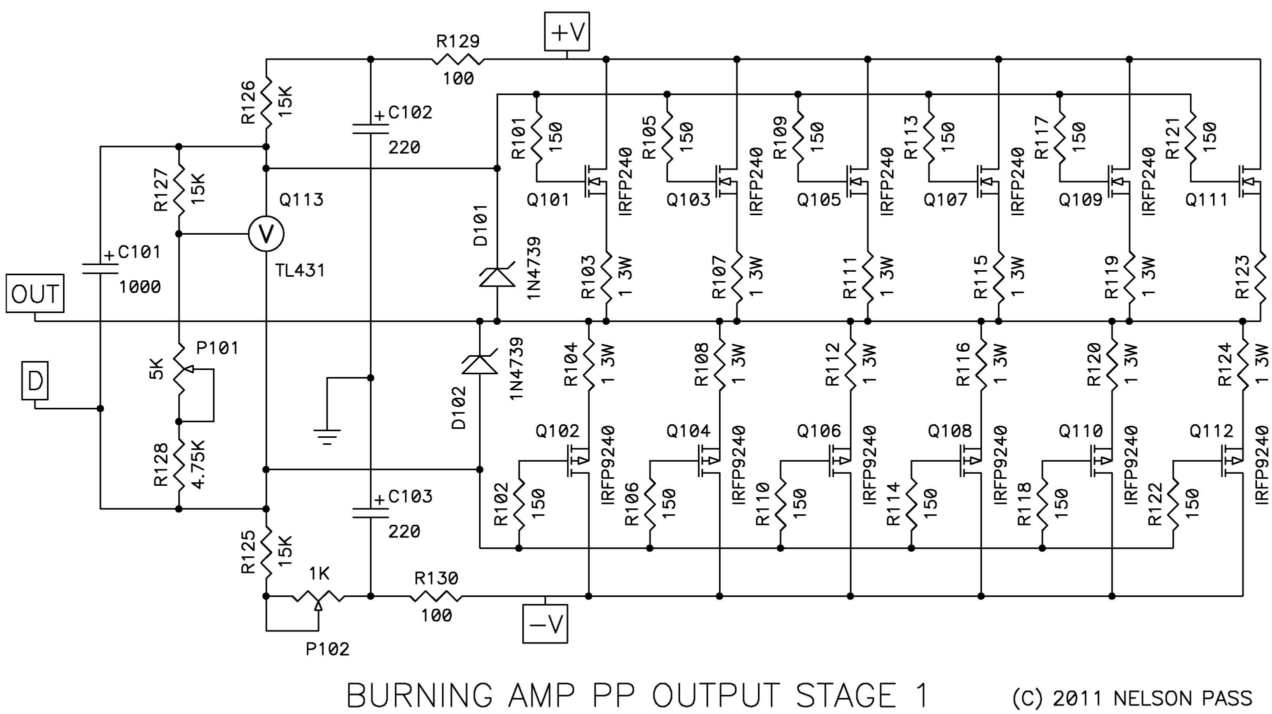


BA-1, BA-2 FRONT END CONSOLIDATED

The official BA #1 single-ended output stage now looks like this:



And the official BA #2 push-pull output stage looks like this:



Both of these circuits now include the necessary components to bias the output stages.

The BA-3

For various rational and irrational reasons, amplifiers without negative feedback loops are of interest to many DIY amplifier builders. With the BA-1 and BA-2 we already have output stages which operated without loop correction, but they both have a front end which uses a local feedback loop to improve the performance, and in this regard their performance is pretty conventional, if simpler than most.

So I think it would be fun to make the BA-3 front end also without the feedback loop and mate it to follower output stages (BA-1 and BA-2 output stages for example) to form complete amplifiers without feedback loops. Of course we have to give up the ease with which loops flatten out the response and lower the distortion. We have to make a circuit which has an intrinsically linear response - good enough for us to be happy with the quality of sound.

This begs the question, “How small does the distortion need to be to sound good?”

That's a very important question to which there is little common agreement. Some people think that distortion should be driven down into the “part per million” region (0.0001%) to be considered adequately low, and some think that it doesn't matter at all. We are going to follow the middle path, where distortion does matter, but it is not required to be infinitesimal.

There is some good agreement that any distortion that does occur should be of “low order”, that is to say second and third harmonic character with very little of any higher harmonics. Is second harmonic preferred over third? Street wisdom says that you would prefer second, but in practice there is a sizable portion of the audiophile population spending money on amplifiers with a third harmonic character, and apparently they like it. There are audiophiles with single-ended tube (SET) amplifiers who think they are listening to second harmonic, but depending on the load-line specifics of the design, they often are not.

I recall Jean Hiraga wrote that the best sound was a modest mix of second and third harmonic with the second being dominant. Individual preferences vary, but I think there is considerable truth to that. This sort of thing is often achieved in only at one region of output level in amplifiers, and I have examples where they seem to deliver their best sound at only one level.

Going all the way back to Threshold, my favorite amplifiers have tended to have a second harmonic character at low levels, gradually transitioning to a mix of second and third, and dominantly third as the output approaches the power limit. Later, we will examine a knob on the BA-3 which allows you to play with this mix.

I don't spend a lot of time getting excited about this sort of thing – I think it's possible to appreciate both types. The point is that low order distortion can be tolerated better, and a design without a feedback loop is going to want to use Class A operation to not only lower the absolute amount of distortion but also to concentrate that distortion into the second and third harmonic.

Nothing unusual about that – front end circuits are routinely Class A. The difference here will be an emphasis on high bias current, and on a balance between degeneration and loading of each gain element.

Degeneration and Getting Loaded

Degeneration - reducing and linearizing the gain of a transistor or tube - is easy to accomplish. Most of the time it simply means that the Source/Emitter/Cathode of the gain device has some resistance placed in series with it. This has always been a popular technique, even before Harold Black invented the negative feedback loop, and remains so today among amplifier designers trying to avoid the dreaded *Transient Intermodulation Distortion* (TIM) boogeyman, a feedback loop artifact.

You should understand that degeneration is regarded by many as a form of negative feedback, and there is considerable argument as to what constitutes the difference between this and feedback loops. However you want to characterize it, it remains local and not a loop.

Loading the output of a gain element is less popular among amplifier designers, at least among those whose diagrams I peer at in the midnight hours. It is not popular largely because it reduces the gain without any other apparent benefit – *just throwing away perfectly good gain that could otherwise be used for feedback*. It conjures up the image of throwing away perfectly good food, and your mother would not approve.

On the surface it makes logical sense, but of course it assumes that you are using a feedback loop. If you are not planning on doing that, then you might consider if there is some other overlooked benefit.

Certainly loading the Drain/Collector/Plate with some resistance lowers the gain, as does degeneration, but unlike degeneration, which raises the output impedance, loading the output lowers the output impedance. That is one potential benefit, since we often a reasonably low output impedance to drive a circuit that might follow.

I think though, there is another. We have a saying in my house (lifted from Cesar Milan) that "A tired puppy is a happy puppy." It is my experience that sometimes you get subjectively better sound from a gain device which is operated at a significant portion of its capacity – not allowed to loaf around with an easy job. We may find that it sounds better if we make it do some *work*. There's nothing magic about this. Making a gain device traveling a load line that represents real work gives it some character. Your mother would probably agree.

I know, I know - this makes the technical cognoscenti roll their eyes and/or snicker. But consider that there are quite a few audiophiles who do not like the sound of loading an input differential pair of transistors with a constant current source or current mirror. They like the sound of a resistor instead, even though it doesn't measure as well. Maybe they like the lower feedback figure that this creates - or maybe they like the sound of their input transistors doing a little real work.

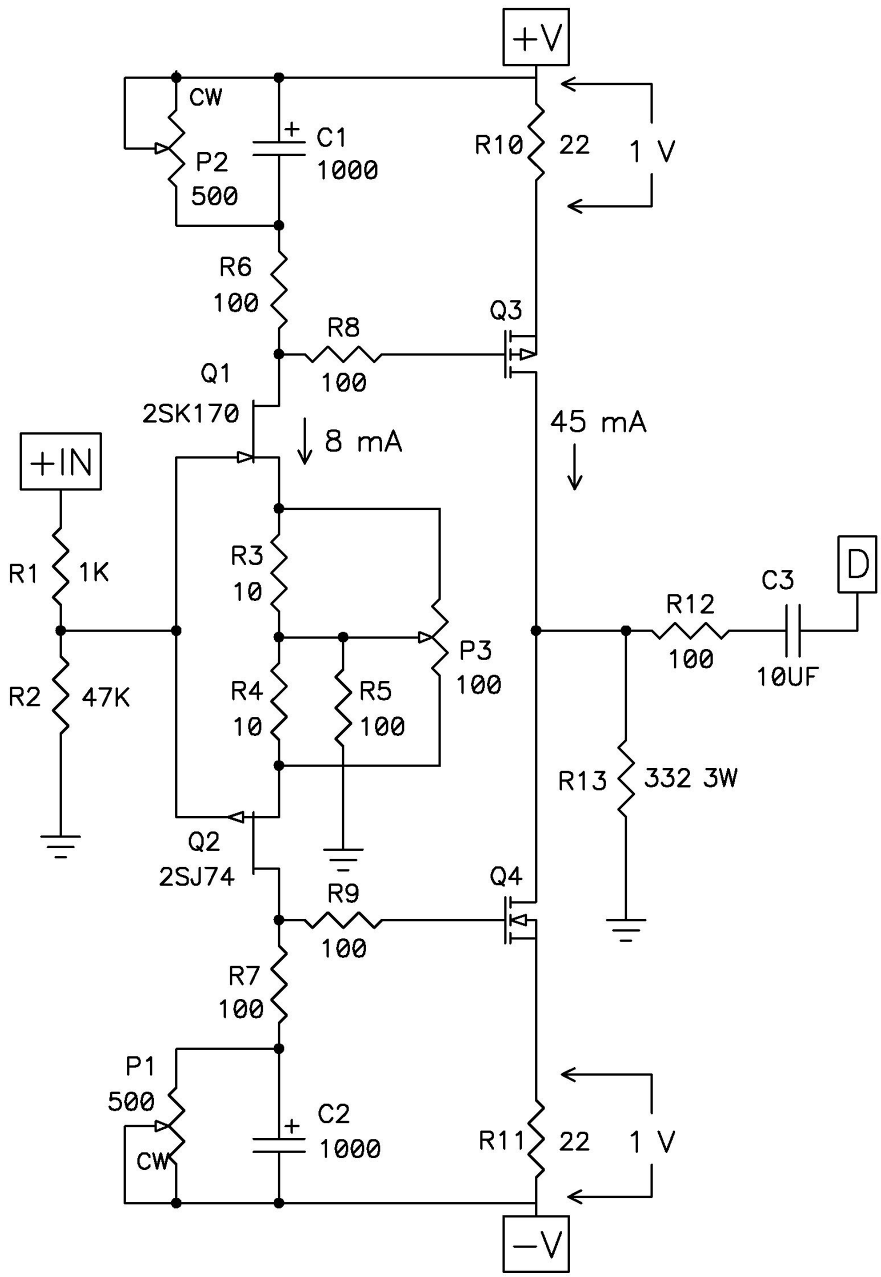
I'm just sayin'...

Anyway, either answer suits our purpose here, so for this circuit we are going to depend on a balance between degeneration and loading to get the measurements and sound that we want.

The Meat

You probably skipped over the previous section to get to this part.

I think you'll like this front end circuit. In fact, some of you already like this circuit – it's a relative of the F5 amplifier, scaled down and absent the feedback loop.



BA3 FRONT END

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This is familiar enough. Q1 and Q2 are JFETs which self-bias into resistors R3 and R4 at currents around 8 mA. R1 is chosen to avoid oscillatory interaction with whatever source impedance you might have, and R2 provides a DC reference to ground in the event that the source does not, and also establishes the nominal input impedance.

Q1 and Q2 are largely degenerated by R5, setting the amount of AC current which flows through them for a given input voltage. The voltage gain of this initial stage is approximately the value of Drain load resistors R6 plus R7 divided by R5. In this case we have roughly unity gain – the Jfets are used as unity gain DC level shifters to Q3 and Q4.

Coming off the Drain of Q1 is the loading network of R6, C1 and P1, and there is a comparable network of R7, C2, and P2 attached to the Drain of Q2. R6 clearly sets the AC load for Q1, but the DC requirements to bias up Mosfet Q3 are higher than that, so P1 in parallel with C1 provides a higher resistance value below about 0.5 Hz, and gives the approximately 3 volt DC drop required to bias the Mosfets.

P1 and P2 are adjusted so as to set the DC bias of Q3 and Q4. You will want to set them at zero when you first fire up the circuit, and increase their resistance to achieve the correct bias voltages across R8 and R9 (about 1 volt) while also keeping the output DC offset voltage at a minimum. This circuit is capacitively coupled at the output, but low offset measured at the Drains of Q3 and Q4 will maximize your output voltage swing.

The voltage appearing at the Gate of Q3 is amplified by something less than the ratio of R10 divided by R8, and with the same happening at Q4 and considering the transconductance of the Mosfets, comes out at about 15. Both of them added make a system voltage gain of about 30X, or 30 dB.

R8 and R9 help set the voltage gain, and they also help stabilize the bias of Q3 and Q4, else it would tend to drift upwards as the parts warm up. The bias current here is about 50 mA, and it will deliver peaks of approximately 100 mA. Q3 and Q4 require heat sinks.

Of course you can bias this circuit higher if you wish – 100 mA bias is perfectly OK as long as you properly heat sink Q3 and Q4, and if you are crazy (like me) you can experiment with higher bias, remembering that the parts are rated at 25 watts, and that it costs you voltage losses across R8 and R9. If you want to play with even higher bias, you can consider lowering the values of R8 and R9 and also R13, all in proportion.

The supply voltage is only critical with respect to the voltage rating of the input JFETs, which are nominally 25 volts. In actual testing, they break down around 40 volts. I wouldn't worry about running them as high as 30V. Hot-rodding this circuit would likely involve cascoding the input Jfets to allow higher voltages.

You can also vary the bias current of Q1 and Q2. You measure the current by reading the voltage across R3 or R4. With 10 ohm, the voltage wants to be 0.08 volts for 8 mA of current. You can play with this, but keep in mind that the dissipation of this part is rated at 400 mW, and for this circuit 10 mA will represent around 200 mW. These JFETs were chosen at Idss figures of 10 mA (Idss is the current that flows when the Gate and Source are at the same voltage). If you get lower values, you may want to consider higher values for P1 and P2, say 1 Kohm. They are best matched, but P3 can be adjusted to compensate for some mismatch.

P3 and the Second Phase

Potentiometer P3 is provided for those of you who have distortion analyzers or want to play with the mix of second vs third harmonic. As with the F5 amplifier, you will find that you can iteratively trim P1 through P3 for minimum distortion. I do so at approximately the 1 watt output level (2.8 V). You should start out with the setting at the mid-point. The minimum distortion point will generally be where the plus and minus halves of the amplifier balance to such a degree as to null out the second harmonic. By introducing an imbalance you can variably re-introduce the second harmonic, and depending on which way you turn the pot, you can choose the absolute phase of the second harmonic.

I knew you would like a choice ;)

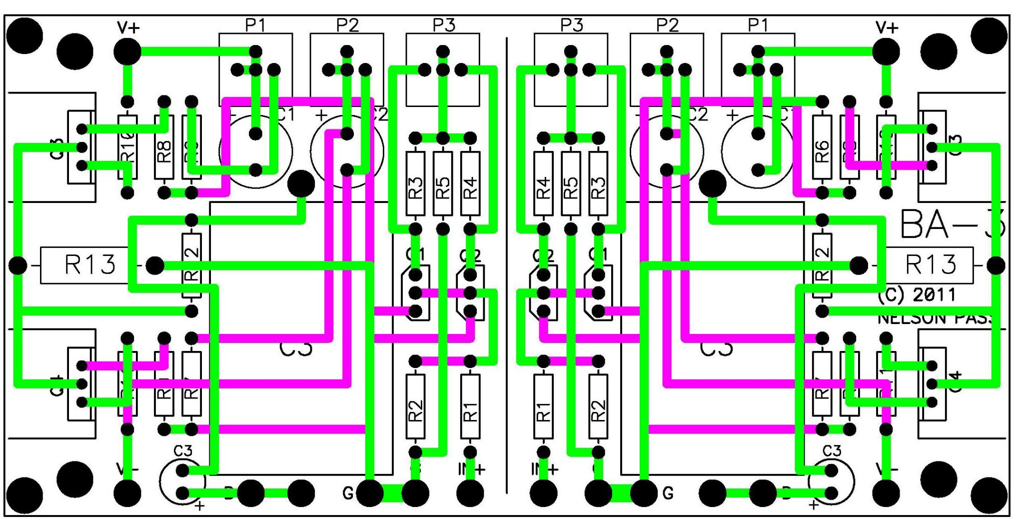
There is flexibility in choices of parts and you can swap these components for others, keeping in mind that you often will have to adjust resistor values to get similar bias figures. I have mentioned different Idss values for the input JFETs, but you may also find that your Q3 and Q4 have Vgs figures different than mine. If so, you may again find that P1 and P2 should be increased.

It usually is not actually necessary to run out and buy a new potentiometer - if the maximum value of the pot doesn't get you want you want, then simply add a resistor (having the same value as the pot) in series with the pot and you can increase it from there which will involve cutting traces on the circuit board.

CONSTRUCTION

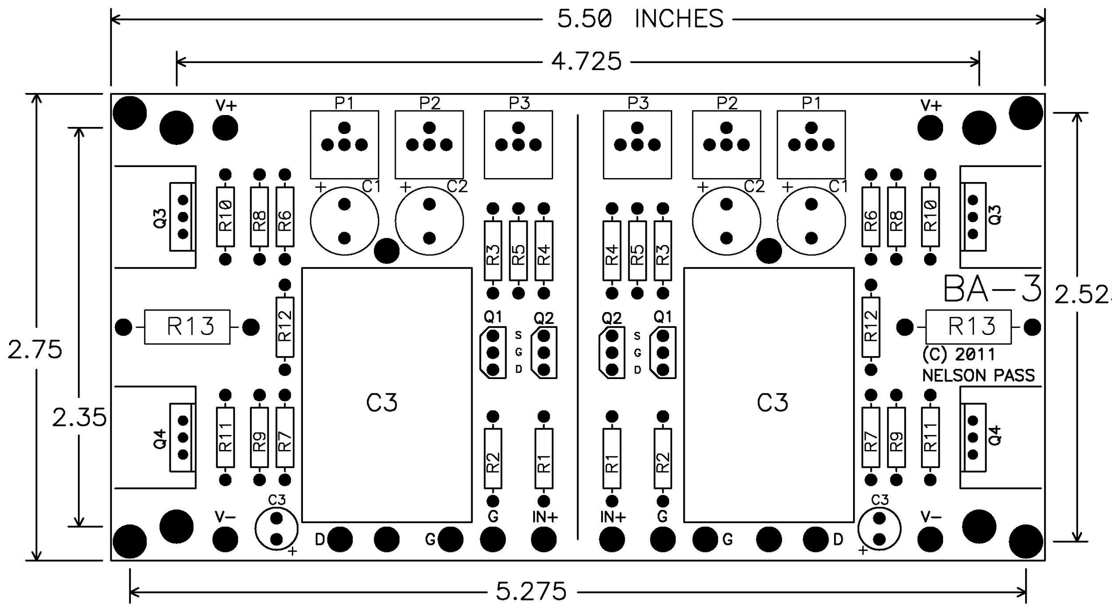
DIYAudio.com will be offering the authorized printed circuit board for sale. Anyone is welcome to cobble together their own for non-commercial purposes, that is to say not for sale.

Here is the trace diagram of this board:



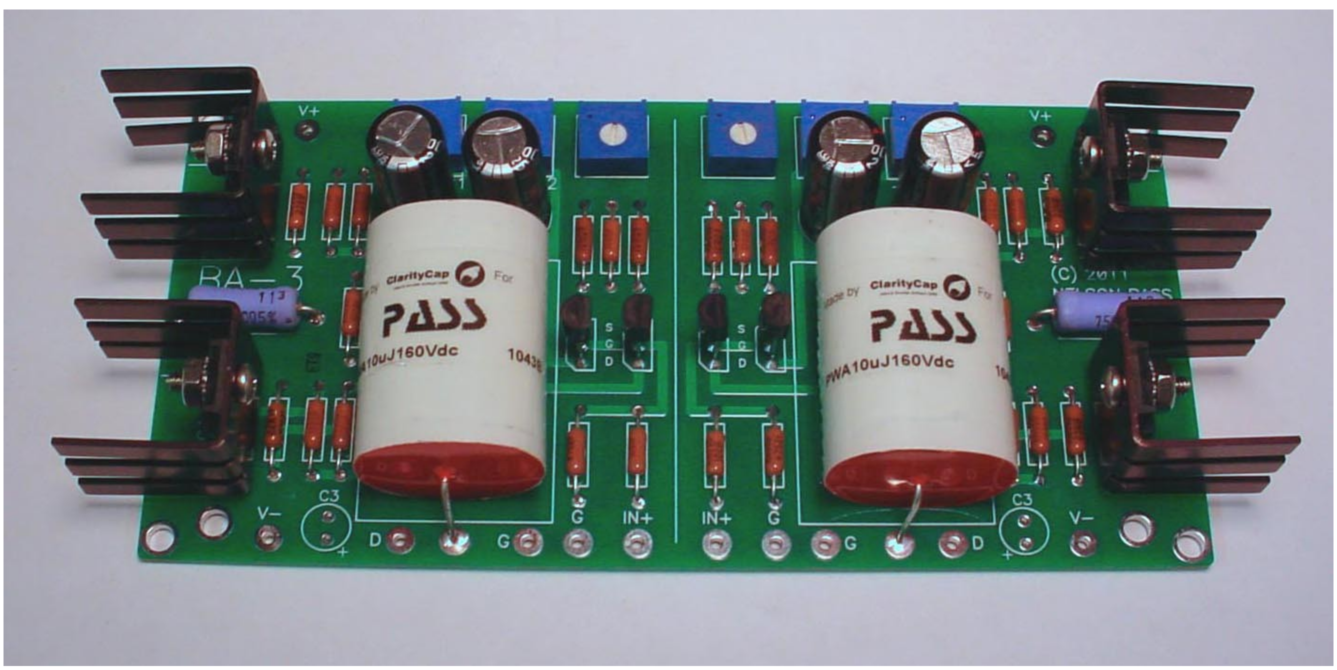
BA-3 PCB TRACES (C) 2011 NELSON PASS

Here is the component placement:



BA-3 COMPONENT PLACEMENT

Here is a nice photo of the board which I assembled all by myself:



You may note a few items. I have made an allowance for two different C3's on each channel, one the polypropylene capacitors in white and another an electrolytic which you don't see stuffed. You may use either or both.

Questions always arise about the quality requirements of various capacitors and resistors and such. My attitude here is that you should feel free to use what you like. Certainly the objective (measured) performance will not depend much on the quality of the passive components. On the other hand, here is a circuit that deserves nice parts.

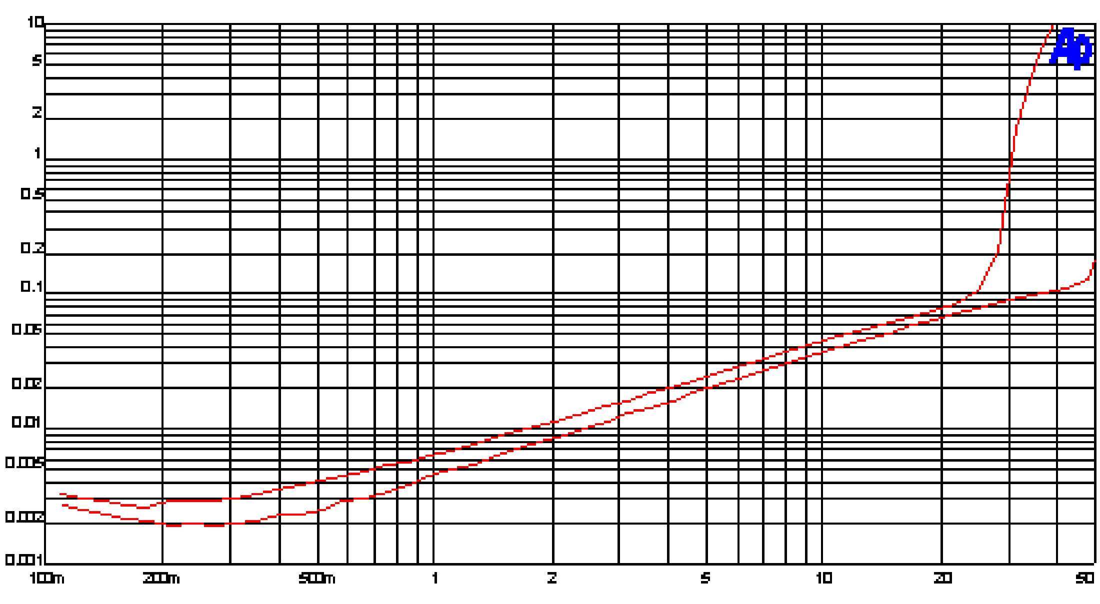
Where you will see some dependence for certain is on the input and output transistors. Toshiba is the supplier of record for this, with the 2SJ74 and 2SK170 input Jfets and the 2SJ313 and 2SK2013 output Mosfets. The 2SJ74 and 2SK170 have become harder to get, but it's worth the effort and price. LSK170's are available, and one of these days I hope that Linear Systems will also get their P channel complement to market. Others will work well enough, usually with more noise and the need to adjust the values of the Source resistors.

The 2SJ313 and 2SK2013 are still available, and you can use them or substitute other Mosfets, such as the Fairchild FQP3N30 and FQP3P20. The Vgs of the Toshibas is about 2 volts. Some of the alternatives will be in the region of 4 volts or so, requiring a higher value for P1 and P2, probably 1K ohm.

Matching is nice, but not essential. P1-3 will cover much of that action. Keep in mind that every time you adjust one of these pots, you will probably have to re-adjust the others, so try to do every adjustment in half-steps, a little bit at a time.

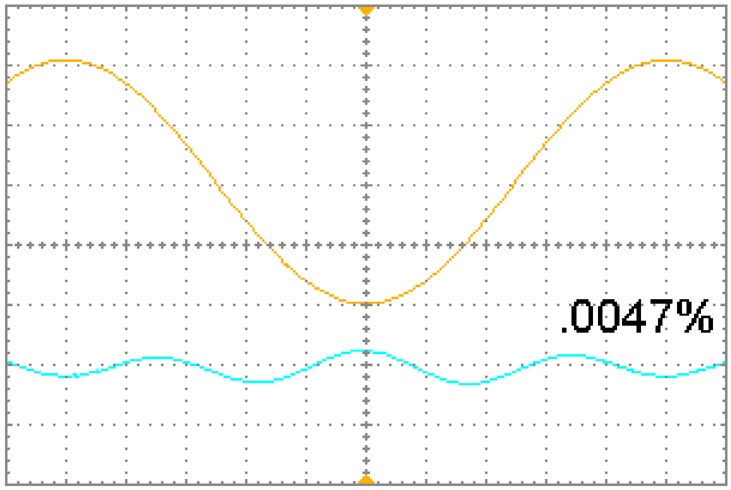
Performance

The data following are for a circuit with the parts and bias as shown and with P3 set for minimum distortion. Here's the distortion curve of the BA-3 front end vs output watts (as if it is driving a perfect follower output stage)



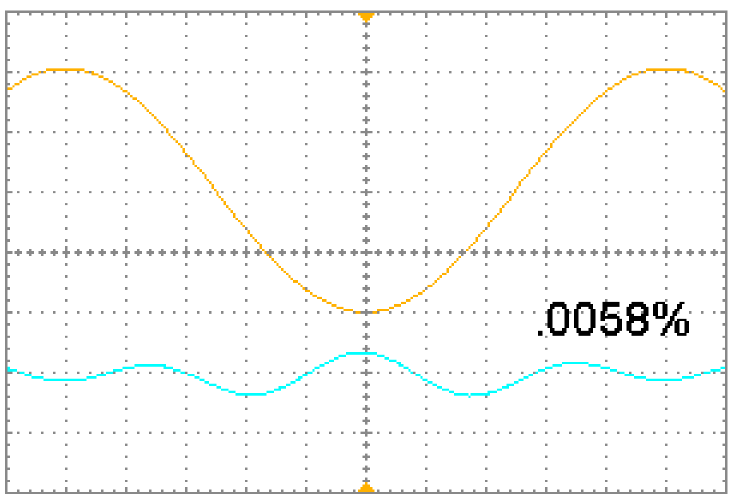
BA-3 DISTORTION VS LEVEL AT 32V AND 24V SUPPLIES

With the second harmonic largely nulled by P3, we see the following distortion waveforms for the circuit at 2.8 volts output without a load (although you will note that the circuit already has its own 332 ohm load to ground). Here it is with +/-32 volts rails:



THD @ 1W 1KHZ 32V SUPPLIES

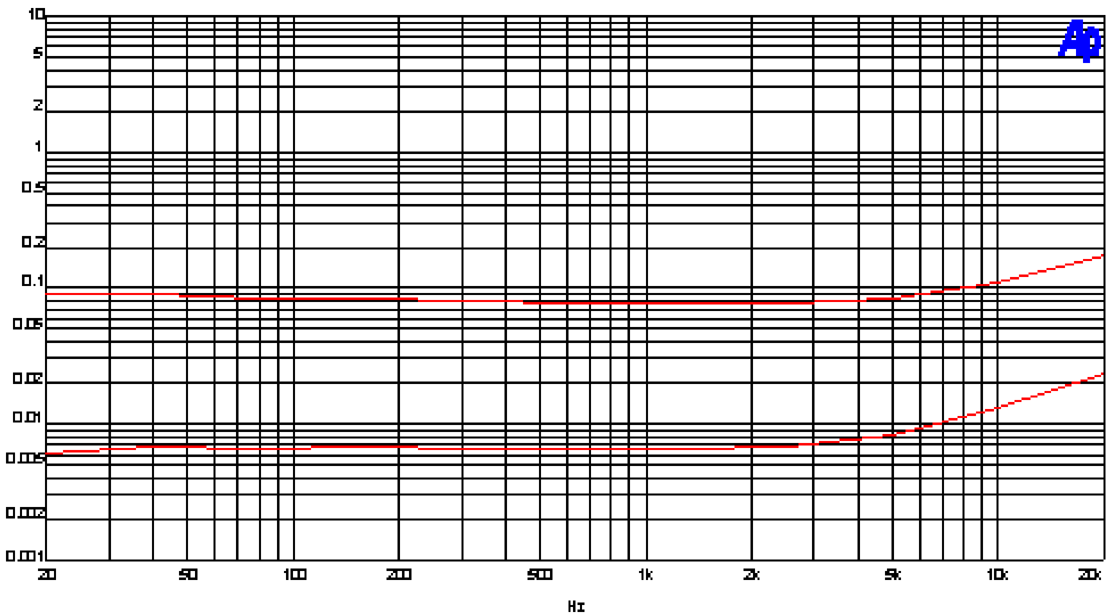
And below is the same, but with +/-24V rails:



THD @ 1W 1KHZ 24V SUPPLIES

You can clearly see that the distortion in this case is mostly third harmonic. Adjustment of P3 will bring the second harmonic out, and as you turn the wiper toward the Source of Q1 you favor the positive going waveform. When you turn the wiper toward the Source of Q2 you favor the negative going waveform. Favoring the positive going waveform is possibly preferred – with proper speaker polarity, this mimics the acoustic character of air.

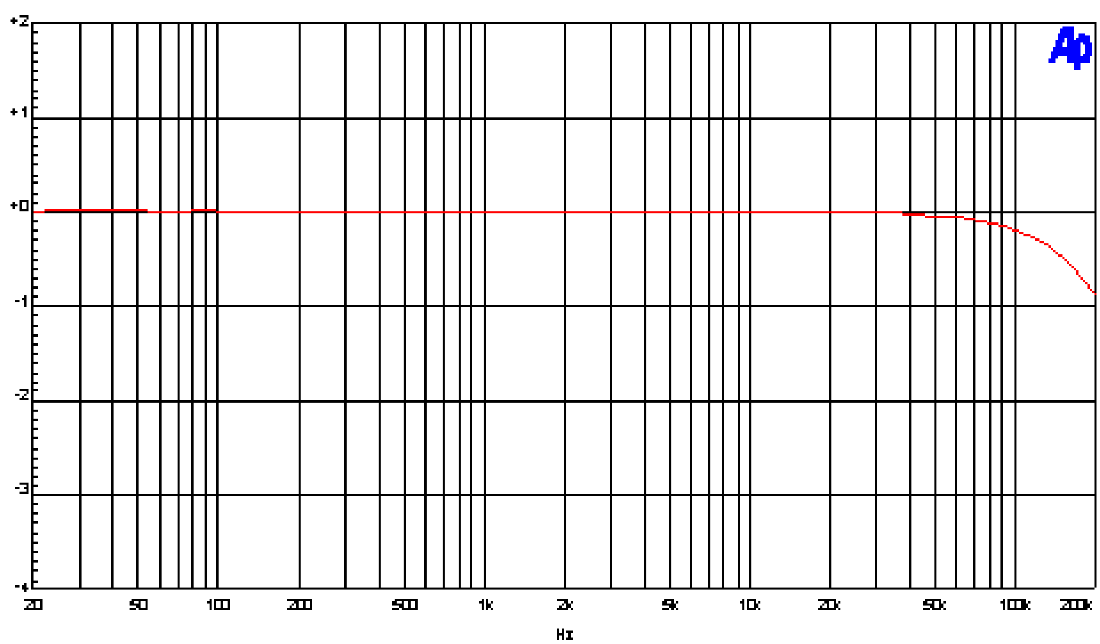
Here's the distortion vs frequency at 2.8V.



BA-3 DISTORTION VS FREQUENCY AT 1 WATT, 25 WATTS

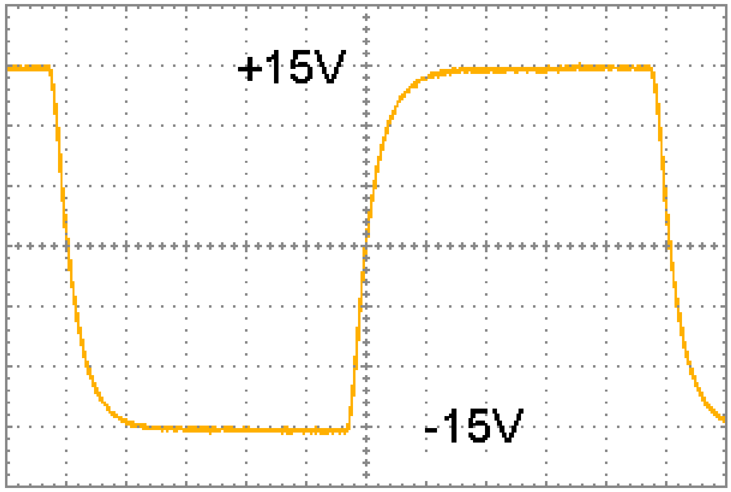
Not much to say here. This was taken with a 600 ohm source. It gets a little better with a 25 ohm source, but that's not to say it sounds better.

Here's the frequency response curve at 2.8V, also with a 600 ohm source:



BA-3 FREQUENCY RESPONSE AT 2.8V

Here's the square wave response at 15 volts output:



100 KHZ SQUARE WAVE

And that concludes the objective part of our show...

In Conclusion...

I bet you thought that getting good Class A performance without a feedback loop was exotic, difficult, and expensive.

Not if you do it yourself.