Tube Designs by Fred Nachbaur

Precision Vacuum Tube Applications

The "RA-100" Integrated Hi-Fi Amplifier

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A Hi-Fidelity Vacuum Tube Amplifier

1: INTRODUCTION

There has been much press lately about the merits (and drawbacks) of the venerable vacuum tube. How much of this retro movement is based on demonstrable principles, and how much is rooted in nostalgia or subjectivity is a debate that could fill volumes.

What *is* clear is that there is considerable renewed interest in vacuum tubes, a technology that even two decades ago was considered as obsolete as spats and top hats. Now the trend is reversing, and a number of manufacturers are again supplying tube gear for audiophiles, musicians and hobbyists. In many cases, these are simply vintage circuits in new packaging. Other products offer new approaches to vacuum tube technology, adding what we've learned in the meantime to come up with some truly noteworthy designs. The project described here fits into the latter category.

Why use tubes anyway?

The central argument for the pro-tube movement is that specs can be almost meaningless, and that what counts is how it sounds to the individual listener. Highly subjective descriptions are therefore used, instead of the techno-babble we've more-or-less gotten used to in recent times. The opposite camp claims that numbers don't lie, and that you can't improve something by adding distortion of any kind.

Both camps have valid points. Most of us have heard expensive gear with spectacular specifications, but were left cold by the "too good to be true," almost clinical sound of such equipment. Similarly, most people will agree that just because something has tubes in it doesn't make it worth listening to. Turn on one of those old "All-American Five" table-top AM radios if you want a striking demonstration of just how *bad* vacuum tube equipment can sound.

Perhaps a way of reconciling the two viewpoints is to consider the distinction between *musical* equipment, and *reproduction* equipment. For musical gear, the individual frequency, waveform and phase distortions are part of what defines the sound of, say, a Fender tube amp. Just as no-one would try to define a fine Stradivarius violin with specs and distortion figures, so also it would be specious to argue that a certain tube guitar amp has over 12% THD (total harmonic distortion) at 35 watts.

Reproduction equipment, on the other hand, has always been expected to give a perfect rendition of the signal applied to it. Sounds good in principle; if such a thing existed, we should be able to exactly and perfectly reproduce everything from a grundge band to the New York Philharmonic, making the reproductions indistinguishable from the original performances.

But therein lies the rub. The overall sound of a stereo system depends so heavily on the room it's in, the speakers, volume level, personal preference, and a host of other fuzzy variables that a perfect reproduction system cannot be said to exist even in this day and age, and probably never will. Add the fact that most if not all recordings are electronically sweetened to some degree to make them "sound good" (as opposed to being an exact copy of the performance), and arguments for the clinical reproduction approach lose credibility.

Carried to extremes, an "ideal" reproduction system wouldn't even have any controls, except perhaps for selecting inputs. If you think about it, personal preference is the only reason why stereos have volume controls, equalizers, and other adjustments to let us customize the sound to suit our very individual ears and brains.

It might be best to view the reproduction gear as a continuation of the same process that started with the construction of the instruments used in the performance. What we ultimately hear is the sum total effect of everything from that original instrument design, to the way it is played by the artist, through the entire recording, mixing and distribution process, to the gear we use to play it, and how we've set its controls.

There is, therefore, a considerable difference in design approach between the "instrument" and "reproduction" categories. An earlier design ("The Real McTube") documented a vacuum tube preamplifier for use as an adjunct to electric instruments. The design approach was largely empirical, and the emphasis was on highlighting the unique distortion characteristics of the vacuum tube. This project explores the reproduction aspect; the design approach was quite mathematical and precise, and the emphasis is on controlling the characteristics of the vacuum tube. The maths used bear a striking resemblance to approaches used in designing solid-state gear; we're changing the active devices from transistors back to tubes, while retaining the same careful design techniques.

Vinyl vs. CD

Similar arguments are ongoing regarding vinyl records vs. compact discs. The CD camp points at the CD's accuracy, definition, and clarity, while vinyl lovers bemoan the CD's lack of warmth and claim that conventional records sound more "natural."

The waters are muddied by the existence of albums reissued on CD that simply *sound* bad. Part of the reason is that master tapes were often "tweaked" (especially by un-naturally boosting the highs) to compensate for the limitations of the vinyl medium; when such tapes are used to make CDs, the result is often too bright and brassy. The better re-issues will usually have a notice to the effect of "re-mastered for CD from the original recordings" as an indication that the engineers took the trouble to adjust the message to the new medium.

In any event, this project gives you the opportunity to explore these subjective and controversial debates, *letting your own ears be the judge*. You may find that there are some recordings that sounds better through the tube amp, and others that benefit from the improved definition (whatever *that* is, technically speaking) that good solid-state gear can offer.

2: DESIGN PHILOSOPHY

The design philosophy was to improve on "vintage" tube designs by incorporating refinements normally only associated with solid-state gear. Prime among these is the use of differential input stages, in which the inverting input is used strictly for feedback. Another is the use of direct coupling between stages, a technique once quite common in oscilloscopes, but rare in audio gear. The resulting circuits are thus closely related to the operational amplifier, and used in a similar fashion.

In addition to the quasi-opamp idea, some other design concepts used in the project are:

1. Use of readily available parts wherever possible. For this reason, tube types such as 6L6GC, 12AX7A and 12AT7A were employed.

2. The power amplifier was designed to be general enough to allow the use of more exotic beam pentodes (and even triodes) by suitably modifying the sockets and/or pinouts as required.

3. Direct coupling was used in crucial parts of the circuitry. More about this later.

4. Flexibility in use was a prime consideration. In addition to the "usual" amenities expected on an integrated amplifier (line-level inputs, phono inputs, tone controls, tape monitors, etc.) some additional features include: a mic/line input with gain trim for stereo instruments, etc.; a "PA mode" switch that allows bypassing the negative feedback employed in the driver/PA module for that "classic tube" sound; full, independent tone controls, including midrange; a tone bypass switch; a stereo/mono switch to allow for bridged mono operation; and an effects loop for inserting graphic equalizers, etc.

   A Word About Negative Feedback

   It is perhaps an unfortunate accident of history that the term "negative feedback" was used to describe the linearization technique of applying a portion of the output signal back to the input, in opposite phase to the applied signal. It gives the impression to audiophiles with enough knowledge to be dangerous that "negative" feedback must somehow be a "bad" thing. I've seen diatribes against the evils of negative feedback, yet in the same breath such critics extol the virtues of "ultra-linear output transformers." The joke is that such transformers (using screen-grid taps on the primary) achieve this linearization by applying local negative feedback to the screen grids!

   Perhaps a better moniker for "negative feedback" would be "dynamic compensation" or something on that order. Consider the following characteristics of well-designed negative feedback systems:

   - A: Precise control over closed-loop gain and minimization of gain drift with component aging.

   - B: Considerable reduction in distortion and noise introduced within a given gain block.

   - C: Reduction of effective series output resistance (at much less cost than using silver-wound output transformers, etc.).

   - D: Precise control and stabilization of frequency characteristics. Extension of high-frequency response where needed.

   - E: Reduces or eliminates the need for careful component matching

3: HOW IT WORKS

A block diagram of the overall amplifier is shown in **Figure 1**. (The line-art “Figures” and other schematics, etc. are available separately, and can be downloaded from my site at http://www.dogstar.dantimax.dk/tubestuf/ampindex.htm#resources . It is highly recommended that you print out these files and have the diagrams at hand as you study this document.)

The mic/line input and phono input(s) share a common high-gain, precision preamplifier utilising a differential input stage.

The Tape, CD, and AUX inputs are of sufficiently high level to be applied directly to the volume control, and thence to the line / tone control amplifier module. This module is essentially identical to the phono/mic preamp, except that its feedback network is the tone control assembly. It is therefore a true active filter, featuring three bands of equalisation (bass, midrange and treble) with approximately +/- 12 dB of adjustment.

The output of the tone control preamp is coupled, via the balance control and stereo/mono switch, to a direct-coupled driver module. This is another differential amplifier, except that in this case we also use the balanced outputs to drive the final push-pull power amplifier. Again direct coupling is used to improve performance; and again, a feedback loop is included (mainly, in this case, to reduce the effective output resistance of the final amplifier). The loop can be bypassed (with gain compensation) so you can do direct A-B comparisons between the "classic" tube sound and the new "clean" sound as implemented in this design.

Not shown in the block diagram is the rather sophisticated power supply, which provides us with the required regulated and unregulated DC voltages.

Figures 2 through **5** in the chapters that follow are a series of schematic diagrams of the Hi-Fi Tube Stereo Amplifier. Since both channels are identical, only one is shown. Please note that part numbers refer to the various sections as follows: (We'll be assuming the left channel in this discussion.)

1-99	**Common parts (power supply, etc.) Also tube designations.**
100-199	**Left phono/mic preamp**
200-299	**Right phono/mic preamp**
300-399	**Left tone/line preamp**
400-499	**Right tone/line preamp**
500-599	**Left driver/PA**
600-699	**Right driver/PA**
700-899	**Power meters**

3-A) PREAMPLIFIER STAGES: OVERVIEW

Considerable effort was taken to design a preamp that is as clean as possible to satisfy discerning audiophiles, while maintaining a relatively low parts count to satisfy limited budgets. It should be noted at this point that, while the discussion is in the context of the full integrated amplifier, there is no reason why you couldn't build standalone preamps for specific applications using this design.

The phono/mic preamps and the tone control/line preamps use a universal preamp module with high gain (about 60 dB), and differential input stages that allow for differential input (as for balanced mic preamps) as well as either inverting or non-inverting single-ended applications. Frequency response shaping and gain setting are accomplished using the appropriate feedback networks.

The schematic diagram of the phono/mic preamp is shown in **Figure 2**, and the tone/line preamp section is detailed in **Figure 3**. Note that these are identical except for the feedback elements. The phono preamp equalisation is accomplished using a small "daughter" card directly on the main board; the mic/line equalisation is accomplished using point-to-point wiring to the gain trim controls; and the tone control equalisation is done on a separate card which include the tone control pots.

The main circuit board was designed to be virtually universal in applicability. Small outrigger "feedback cards" are used to customize the response to almost anything you might need. **Figure 6** plots gain vs. feedback ratio for the non-inverting configuration as used in the phono/mic preamp. There is an obvious linear relationship, except at high gain. (The "porch" at the low end is due to the "+1" when using the non-inverting mode; see below for further details. In the inverting mode, as used in the tone preamp section, the relationship remains linear at low gain settings.)

As detailed here, the phono/mic preamp is selectable (using a jumper or switch) between magnetic phono (RIAA equalization, high gain), ceramic phono (flat response, low gain). An external switch or switching mic input jack selects between the phono and mic/line (flat response, variable gain) modes. See **Figure 7**, which graphs open-loop response and the two closed-loop phono curves.

There is no reason why you couldn't adapt this universal preamplifier module for any other gain/equalization combinations, simply by modifying the feedback networks. (**Figure 8** gives a few ideas, covered in more detail in Other Preamp Applications.)

Phono mode

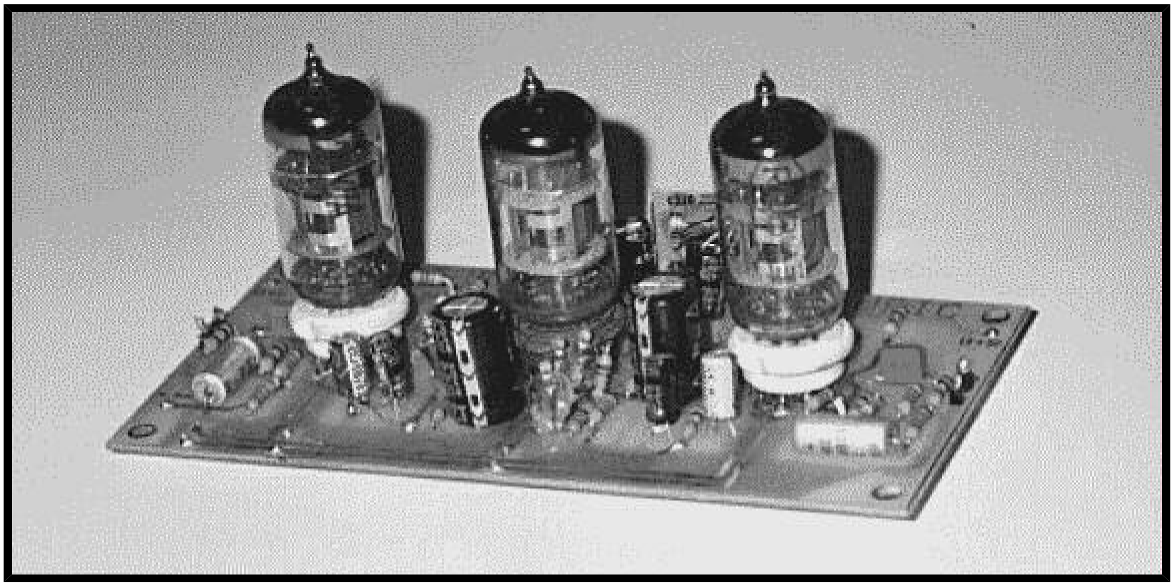
In magnetic phono mode, the RIAA curve is achieved by using negative feedback to decrease the gain at higher frequencies, according to the RIAA specification. The result is that the amplifier runs almost open-loop (about 60 dB gain) at 40 Hz, and rolls off smoothly to a gain of less than 20 dB at 20 kHz. This means that the "tube sound" caused by even-order distortion will be most pronounced in the bass region, giving it that warmth that is so highly prized by vacuum tube aficionados. At higher frequencies, this distortion is increasingly cancelled out by the negative feedback that sets the gain, keeping the mids from sounding "brassy," and the highs from sounding "splashy" (two common complaints about open-loop tube amplifiers).

In the optional ceramic cartridge mode, response is flat within 0.5 dB over the audio range, and voltage gain is set at 5 (about 14 dB). At this relatively low gain, the circuit is very clean and distortion-free. However, discerning ears may hear a subtle quality of warmth not present in solid-state gear.

The tube types and their operating points were carefully chosen to minimize power supply requirements, and to maximize tube life. Both channels together (four preamps) draw only about 15 mA from a 430 volt supply.

Mic/Line mode

The mic/line mode is similar to the ceramic phono mode, in that it is set for essentially flat response over the audible frequency spectrum. The exception is that a separate "Gain Trim" pot is provided for each channel. Minimum gain is approximately 1.5 (about 4 dB), handy for line-level devices such as some older CD players that may not quite have enough drive for the standard line-level inputs. Maximum gain is over 200 (46 dB) which is plenty for virtually any microphone or direct instrument input. It should be noted that you should use only as much gain as is necessary, since frequency response and distortion figures degrade at the highest gain setting.



A photograph of the Universal Preamp Module, configured for phono and tone amp use

3-B) PRE-AMP DIFFERENTIAL INPUT STAGES

Each of the modules (Phono/mic, Tone/line, and Driver/PA) uses a differential input stage, giving us both an inverting and a non-inverting input and affording additional useful and beneficial features. The following discussion relates directly to the Phono/mic preamps (refer to **Figure 2**), but applies by extension to the other modules also.

The appropriate input signal is coupled from the phono or mic input jacks, through switch SW2, to the grid of the first section of V1 (a 12AX7A vacuum tube) via coupling capacitor C101. Since this grid is at a substantial DC voltage above ground, resistor R101 is used to insure that the input is always at DC ground potential. Note that, as far as the input AC signa is concerned, this 100K resistor is effectively in parallel with R102 (also 100K), giving us the 50K input impedance required for most magnetic phono cartridges.

The two triode sections of V1 are used as a differential amplifier. Note that the cathodes are tied together, and go to ground via a relatively large shared cathode resistance consisting of R104 and R105. This acts as a rudimentary constant-current source. If the current in either triode increases, a comparable amount of current will be robbed from the other triode. Therefore, if the voltage on the signal input increases (goes more positive), plate current increases and the plate voltage of that triode will drop, while the plate voltage of the other triode increases by about the same amount. However, if the voltage on the feedback input increases, causing an increase in current in that triode, the voltage on the plate of the first triode section will increase and that of the other triode will decrease.

A portion of the cathode resistor voltage drop is sampled by R104, low-pass filtered by R103 and C102, and provides the grid bias for the first section of V1 via R102. An interesting feature of this circuit is that this also sets the bias and operating points for the entire amplifier, as we'll see as we progress through the analysis.

Experienced electronicists will be quick to point out that a simple cathode resistor is not an ideal current source. As a result, the "common-mode rejection ratio" (that is, the gain matching of the two inputs) of this circuit is quite low. (A great improvement results by using a pentode as a current source in the differential amplifier cathode circuit, a trick that's used in the driver stage where common-mode reduction is more critical because we also require balanced outputs.)

What we do instead, in the preamp modules, is "cheat the system" and carefully choose the operating points of the two triodes to be different, to help offset the error. Note that, even though there is approximately the same current flowing in the plate circuit of each triode section (about 500 microamperes), the plate voltage on the second section is considerably higher (by about 90 volts) than the plate voltage of the first section. This is because of the lower value of plate resistor R107, as compared to R106. This "trick" assumes that the two sections of the tube are reasonably well-matched, and will furthermore track as the tube ages. Tests with different tubes of varying manufacture and condition have verified that this assumption is indeed valid.

3A-2: PREAMPLIFIER OUTPUT STAGE

The output of our differential amplifier is direct-coupled to the grid of the second stage, V2, a 12AT7 dual triode. This tube was chosen over the more common (and cheaper) 12AU7 because it sports about three times the gain (transconductance), yet is capable of almost the same output drive.

R115 in series with the grid of V2 introduces a high-frequency pole with V2's grid-to-plate capacitance, effectively limiting high-frequency response. This allows us to run the preamp with relatively heavy negative feedback (low gain) without worrying about oscillations caused by phase shifts. You can think of it in op-amp terms as "internal compensation".

The difficulty with direct-coupled vacuum tubes is that the grid of the second stage sits at a substantial voltage above ground (about 200 volts in our circuit). This is why we need a relatively high B+ voltage, since we have to insure that the plate circuit of the second stage has enough headroom. So why bother at all? After all, tube amps have been made for over half a century with good old RC interstage coupling. The answer has to do with dynamic stability as the tubes age. Unlike conventional multi-stage tube amplifiers, our circuit is self-levelling because its DC biasing is in a closed feedback loop. Another reason for using direct coupling is because of the relatively large voltage swings at this point in the circuit. Capacitors (even good ones) can exhibit some non-linearity in capacitance vs. voltage -- a fact that is often overlooked in amplifier design. Direct coupling eliminates this capacitor, and as a bonus gets rid of one more pole in our transfer function that could complicate our application of negative feedback for gain and frequency response control.

In order to bias V2's grid at the proper voltage (about -1 volts with respect to cathode), we need a humungous cathode resistor R108. To avoid losing all our AC gain, this resistor is bypassed for AC by capacitor C106.

3A-3: PREAMPLIFIER DC FEEDBACK

The rest of our circuitry completes the DC feedback loop. In case you're wondering about the presence of the three NE-2 neon bulbs in series, let's analyze the circuit without them. The plate of V2 will be sitting at about 315 volts, whereas the grids of V1 are at about 27 volts, or a factor of about 12:1. We could use a 12-to-1 voltage divider, which would give us a DC closed-loop gain of about 13. That is, it would require a 13 volt change at the output to compensate for a 1 volt change at the input.

We can easily improve on this considerably. The lowly neon bulb can be viewed as the tube-technology equivalent of the zener diode. That is, the voltage drop across the bulb (typically about 65 volts) is reasonably independent of the current passing through it. The three NE-2's can therefore be viewed as a 200 volt "level shifter," allowing us to use a much lower voltage division ratio (about 4:1) in our DC feedback loop. The result is an almost 3 times improvement in DC operating point stability.

The level-shifted and divided output voltage goes through a two-stage low-pass filter consisting of R111, C105, R112 and C104. This strips AC signals from our DC feedback loop, to insure that the amplifier will still have full open-loop AC gain. The resulting DC feedback voltage is applied to the inverting input of our differential amplifier via R113. The junction of R113 and C104 also forms a convenient "AC Ground" point for our signal feedback networks.

Let's step through what happens if some change occurs in DC operating point. This change could be caused by tubes and other components aging, power supply voltage fluctuations, or input overdrive conditions. Let's assume that the change causes the output voltage to rise, as would be the case as V2's emission decreases. This would cause an increase of bias voltage (more positive) on the grid of V1B (the feedback input), causing that stage to draw more current and increase the voltage on the common cathode. V1A (our input stage) would therefore draw less current, causing the voltage on its plate (and therefore the grid of V2) to increase. This would cause an increased V2 plate current, resulting in a decrease in plate voltage, tending to buck the original change. See how the overall circuit is self-regulating?

Resistor R114 forms the plate load for V2, and C107 couples the output to the tone/line preamp module via the input and tape monitor switches. Capacitor C108 provides another pole of high- frequency attenuation (compensation) to prevent oscillation at low gain. (Before I added this additional compensation, the first prototype became a dandy 11 megahertz transmitter at gains lower than about 3!)

3A-4: PREAMPLIFIER AC FEEDBACK

So far we have a pre-amplifier with an open-loop passband gain of about 60 dB, with 3-dB corners at about 40 Hz and 2 kHz. (See **Fig. 7**.) While the low-frequency end isn't bad, the high-frequency end is pretty awful. This is partly because of the compensation intentionally introduced by R115 and C108 and partly because of tube electrode and circuit wiring capacitances. Not to worry, our bandwidth automatically increases again when we apply negative feedback, just as it does with solid-state compensated op-amps.

As hinted already, this circuit behaves very much like an operational amplifier (op-amp). But before we get on with designing feedback networks, we'll point out the ways in which it is *not* like an op-amp:

- a) Open loop gain, although quite high, cannot be considered infinite as in many solid state op- amp devices.

- b) The circuit exhibits a large output-to-input voltage offset (on the order of 290 volts). Any AC feedback elements between output and input therefore have to include DC blocking capacitors. The alternative would be level-shifters, which would in my opinion be an additional complication (and expense) without commensurate rewards in terms o performance.

- c) The DC voltage at the feedback input is non-zero (about 27 volts in practise), so again there is a need for DC blocking. The point marked "ACG" (AC Ground) is provided for convenience, acting as a virtual ground for AC, at the same DC voltage as the inverting input.

The not-quite-ideal Tube Opamp

Keeping these restrictions in mind, we can use the formula for the classic non-inverting op-amp to approximate our gain with feedback. Note that the inverting input (-IN) has a 47K resistor (R113) to "AC Ground". This is our "default" value for input resistance to the feedback input. Let's call that resistance Ri, though (as in the case of the tone control amp discussed later) it can be considerably higher, as needed. The bare-minimum feedback network would consist of just a single resistance (we'll call it Rf) in series with a DC blocking capacitor between output and -IN. The theoretical gain with feedback would then be:

Av = (Rf / Ri) + 1

For instance, let's compute our gain if we connect a "bare bones" feedback network consisting of a 430K resistor in series with a DC blocking capacitor between "OUT" and "-IN". That is, Rf / Ri = 9.15, so our gain would be 10.15, or about 20 dB.

The feedback elements do not have to be pure resistances; the above formula could be generalized to include complex impedances.

Av = (Zf / Zi) + 1

The circuit's actual performance follows this predicted formula very closely, verifying that our gain-matching shortcut described earlier works just fine. See **Figure 6** for an actual plot of the prototype. The slight curve at the low end is caused by that "+1" factor in the equation; as Rf / Ri increases, that factor becomes less significant, and the graph approaches a straight-line relationship. However, at gain settings above about 200 (46 dB), the relationship begins to fall apart as we approach the amplifier's open-loop gain. Incidentally, this gain setting is also the practical maximum as regards frequency response; the 3 dB corner at this gain will be on the order of 16 kHz.

3A-5: PREAMPLIFIER PHONO, CD and MIC/LINE FEEDBACK NETWORKS

Ceramic phono and Mic/Line

The feedback network for the Ceramic phono input is little more than our "bare bones" network described above. R118 in series with C109 forms our Zf . R119 is added as a refinement to insure that the negative end of C109 is always held at the DC potential of our feedback input, eliminating the massive pop that would otherwise result when switching modes. Note, however, that it is effectively in parallel with R113, lowering our Ri value to 38.7K. You can verify that our closed-loop gain would therefore be (150/38.7)+1, or about 5 Input resistor R123 attenuates our input signal by a factor of about 3:1, so the overall system gain is a little less than 2 (5 dB).

The final element is C110, which introduces a 3 dB corner at about 20 kHz, rolling off ultrasonics that we aren't interested in. (Without this capacitor, the gain is actually flat to well beyond 100 kilohertz! See how feedback got rid of that open-loop corner at 2 kHz?)

The mic/line feedback network is similar, except that the gain is variable over a considerable range (about 2 to 200) by the gain trim pot.

Magnetic Phono

The magnetic phono feedback is only a little more involved. The straight thin lines in **Figure 7** show the theoretical RIAA specification (asymptotes), and the curve shows the actual response of our preamp in this mode.

Note that, in the bass region, our preamp has to run open-loop in order to get the required 60 dB gain. Our feedback network therefore doesn't even have to bother with the poles in this region. Instead, R120 was chosen to present an appropriately low value for Ri . The effect is to further "swamp" any low frequencies that manage to make it through the DC feedback filter consisting of R111, C105, R112 and C104. This pushes our low-frequency "hump" to exactly where we want it.

As frequency increases, at about 50 Hz., C111 comes into play and acts as an integrator to roll off our response. At about 530 Hz., the pole consisting of C111 and R122 tries to level the response, and at about 1600 Hz. C112 and R122 introduce the second roll-off section. The net result is a gradual rolloff over the audio range at a somewhat gentler average slope (about 4.5 dB per octave) than would be achieved with a single-pole integrator (6 dB per octave). Finally, R121 limits minimum gain to about 6 dB at ultrasonic frequencies, helping to insure stability.

3A-6: TONE CONTROL PREAMPLIFIER

The tone-control modules are virtually identical (see **Figure 3**), except for a couple minor component changes and different feedback networks. When tone bypass is selected, the voltage gain is set at 10 (20 dB) over the audio spectrum. In tone control mode, the gain will be approximately the same with all three tone pots in the centre of their rotation. Varying the pots will give up to about 12 dB of boost or cut, via the three variable T-networks. Note that the tone- control mode is basically a unity-gain configuration (to allow equal boost and cut). We achieve the nominal 20 dB voltage gain (=10) with a 10:1 divider at the ouput, feeding the input side of the tone control networks.

By now it may be obvious that we are using the INVERTING mode for the tone preamps. The non- inverting input is effectively grounded for AC using a capacitor (making the schematic look strange indeed!), and both signal and feedback are applied to the inverting input. This greatly simplifies the design of the feedback networks. To ensure that our system phase is consistent from input to output, the inverting mode is also used in the driver/PA section that follows.

The feedback network for the tone control stages is an outboard card that also contains the linear 500K gain control pots. This also includes the gain bypass switch, which essentially reduces the tone control amplifier to a simple line amp with a pre-set gain of 10 (20 dB).

There is an important component marked C_{tweak} on the schematic. The purpose of this capacitor is to cancel out the effective parallel capacitance of your input cabling. The proper value is best determined experimentally. Connect a decent signal generator to the CD input, and measure the output signal at C307 with an oscilloscope or AC voltmeter. Set the scope or meter for a full-scale reading at 1000 Hz, then increase the input frequency until the output reaches 71% of full-scale. Note the frequency in kilohertz (we'll call it F_0); the desired capacitance in picofarads can be found using the formula:

$$159000/(F_0 * 47)$$

In the prototype, the F_0 frequency was about 7.5 kHz, and the simple addition of a 470 pF capacitor levelled it out to well beyond 30 kHz. This tweak capacitor is best installed directly on the tone control card.

Again, we have the op-amp design model of the preamplifier to thank for this simple fix to a problem that plagues many a tube design, often without any simple recourse.

Also on the topic of frequency response: the low-frequency response of this stage is a bit of overkill, with a 3 dB corner well below the audible range. This can actually be a drawback, since subsonics such as turntable rumble can result in power wastage, reducing the power available for audible frequencies. A simple fix is to put a 0.1 uF capacitor into the "FX loop" instead of the usual jumper cables. This rolls off the subsonics below about 30 Hertz (dependent somewhat on volume control setting; the rolloff frequency is somewhat higher at high volume control settings, acting almost like a simple "loudness contour").

3B: OTHER APPLICATIONS FOR THE UNIVERSAL PREAMP MODULE

Figure 8 shows a few other options for the basic preamp circuit. This was originally intended as an Appendix, but it's sufficiently important to be included in this discussion of the "Quasi-opamp Tube Preamp." You could design any number of stand-alone applications for this device, the ones provided here are just a few examples of the possibilities.

You might have an old reel-to-reel in a closet somewhere. If not, keep an eye out at the second- hand stores; these can often be picked up for a song. The biggest drawback with many of these is the early transistor playback preamps, especially the ones that use germanium transistors. Replace the playback amp with the variation shown in **Fig. 8a**. Note that for proper equalization at both speeds (3-3/4 and 7-1/2 ips) you'll need to switch between two different values for C_{NAB}.

The NAB tape standard is even simpler than the RIAA phono spec, consisting essentially of a 6 dB per octave de-emphasis. Again, we use the preamp in open-loop mode at low bass frequencies, and again we use a low value of Ri to maximize our bass response.

Fig. 8b shows a preamp for high-impedance, single-ended dynamic microphones. It is similar to our Mic/Line mode, except that the fixed gain simplifies construction for standalone projects. Add an impedance-matching transformer, and the circuit is usable with low-impedance microphones.

Fig. 8c shows a application of the differential input capability of our preamp. This is for balanced, low impedance microphones. Note that this is a transformer-less circuit, and therefore offers a significant improvement over **Fig. 8b**. However, the circuit's shortcomings should be pointed out: First, as mentioned earlier, the common-mode rejection ratio of this circuit is nothing to write home about, and so it will not be suitable if your microphone is connected via a long cable run near stage lighting cables, etc. Secondly, the microphone element will be sitting at about 25 volts DC relative to system ground. A dedicated balanced mic tube preamp that avoids these shortcomings would not be difficult to design, I'd consider taking on such a project if the interest is there.

Finally, **Fig. 8d** shows a "spring reverb" driver. The transformer can be a small output transformer, such as might be salvaged from an old tube radio. A turns ratio of about 25:1 (corresponding to an impedance ratio of 5000 ohms to 8 ohms or thereabouts) would be fine. The first preamplifier (e.g. "left") amplifies your instrument's output with enough drive to run the reverb input transducer via the matching transformer. The second preamplifier (e.g. "right") amplifies the much lower voltage appearing at the reverb's output transducer. Separate gain controls allow you to mix "dry" and "reverb" signals to your heart's content. The resistor values are given as a starting point; depending on the instrument and your particular spring reverb assembly, you might have to trim values to suit. As shown, there should be enough gain to allow you to distort either signal by overdriving the preamp sections.

Incidentally, because of the unique DC feedback system, the clipping characteristics of this circuit are quite a bit different than what you might be used to, even compared to other tube circuits. The author has not experimented with the fine points of this, so the field is ripe for experimentation in this area.

3C: DRIVERS AND POWER AMPLIFIERS

The driver/PA section is shown in**Figure 4**. Starting from the output side of the circuit, there is a more-or-less conventional push-pull pair of 6L6GC's feeding a high-quality output transformer, which couples the amplifier's output signal to the speaker load. Small value cathode resistors give a very slight amount of local feedback and help to reduce cross-over distortion. Mainly, however, these give a convenient test-point for measuring quiescent plate current of each tube. Divide the voltage across the resistor (in millivolts) by ten to obtain quiescent current in milliamps (this will typically be 40-50 mA, corresponding to a grid voltage of -72 volts with most tubes).

At the grids are silicon diode pairs which absorb most of the grid current, should grid voltage exceed about +1.2 volts. These are for protection during power-up, should the user forget to turn off the "standby" switch before enabling the main power switch. (Note: there is a "hack" available in the "Tweaks and Hacks" section that eliminates this problem.) These diodes also effectively prevent unwanted class AB2 operation which could easily damage your tubes at high signal levels with low grid bias.

The screen grids are tied to taps on the output transformer, supplied specifically for this purpose. Such transformers are sold as "ultra-linear" transformers, but it should be pointed out that it's not the transformer that is particularly linearized by this process, rather it's the local negative feedback to the screen grids that causes the substantial improvement in linearity (and thereby reduction of THD). This design also allows us to use substantially higher plate voltages without damage, permitting higher output power and better overall linearity.

The control grids are direct-coupled to the plates of a differential input pair, using a 12AT7A this time because of its higher drive capability. Unlike the comparable circuit in the preamp stages, however, this circuit is exactly balanced by using identical operating points in the two triode sections of the tube. Also unlike the preamp, it is fairly important that the two sections of the dual triode be reasonably well-matched. If you find that you have to turn the "balance" control (discussed later) too far from centre to get equal grid voltages on the two output tubes, try a different 12AT7A.

Also unlike the preamp circuit, the common cathodes are not current-driven via a large resistor. Instead, a small pentode is used as a current source.

This pentode, a type 6CB6A, was chosen strictly for its DC characteristics. Any other sharp-cutoff pentode with similar characteristics will work, with suitable modification to the cathode resistor values. Its only function is to provide a reasonably constant current to the two differential amplifier tubes, thus assuring a high common-mode rejection ratio. This also increases immunity to fluctuations in the negative supply voltage, which is why we can get away with a simple zener diode reference voltage (though see the "Tweaks and Hacks" section for more about this).

Note that the cathode of V8 is sitting almost at -590 volts, and the cathodes of V7 will typically ride at about -300 volts. This would present an unacceptable heater-to-cathode voltage on both tubes, if the same (ground referenced) heater supply were used as supplies the rest of the circuit. This is the reason for using a separate heater supply. The 5V winding on the power transformer was nominally intended for the heater of the 5U4 (or similar) rectifier tube, and is therefore well- isolated from ground. A resistive voltage divider in the power supply biases this heater voltage at approximately the midpoint between the cathode voltages of V7 and V8. Even at that, we're pushing the limits of the recommended heater-to-cathode voltage; however, I have not observed any ill effects because of this, using tubes of varying manufacturer and condition.

The screen of V6 is set at +120 volts (relative to cathode) with a zener-diode regulator, to help insure the tube's constant-current characteristic. The control grid is set at a variable slight negative voltage relative to cathode, which gives control over its current output, thereby shifting the operating point of the differential pair and providing our PA bias adjustment. (Looks like the long way around to set grid bias, but it's actually very simple, reliable, and in a funny way, elegant.)

Offset between the differential stages is set with a trimpot in the cathode of the non-inverting half of the differential pair. This allows us to trim out differences in tubes, and assure exactly the same quiescent current flows in the plate circuit of both output tubes. Again, if you have to turn this balance pot too far off centre, use a 12AT7A with better matching between sections.

Note that PA tubes' bias can either be set to a given voltage at the grid of each output tube (typically -72 volts), or it can be set by adjusting for equal voltage on the cathodes (representing equal current draw in each section) - typically 40-50 mA. The latter method is probably the better, unless the tubes are reasonably well-matched. Incidentally, this provides an easy way to check the match between two tubes, at least as far as emission (and to a lesser degree, transconductance) are concerned. Set the grids to be equal, and the cathode voltages should be equal also (within reasonable limits).

A "PA mode" switch allows us to switch in or out a negative feedback network around the whole circuit. Since the circuit includes the driver tubes, PA tubes, and output transformer, any non- linearity in any of these components is greatly reduced. The main advantage, however, it reduction in equivalent output resistance, which reduces the depression in midrange response on most dynamic speaker systems. Effective series resistance (open-circuit voltage divided by short- circuit current) of the prototype was reduced from about 18 ohm to approx. 8 ohms -- ideal for 8- ohm speaker systems. (It should be noted that this does not affect the "impedance match" as applied to maximum output power capability.)

This feedback network can be bypassed for that "classic" tube sound, should you desire to do A-B comparisons. You might actually prefer the "classic" sound, especially on older rock `n' roll material. To allow us to set the same gain in "classic" mode as in "clean" (feedback) mode, a gain trim control is provided for the "classic" mode.

The open-loop voltage gain of the driver/PA circuit is approximately 25 (at the 8 ohm output tap). We use feedback to reduce this to a gain of about 4, so it is reasonable to expect about a 5:1 improvement in noise and distortion specs. Given that the open-loop distortion of a pair of 6L6GC's in AB1 mode is rarely better than 1%, we can reasonably expect to lower this to about 0.2% by using this "dynamic compensation" system. Another way of stating this is that the total level of distortion will be reduced by well over 10 dB... nothing to sneeze at!

100 ohm Resistor R515 (615) is shown at the output of the PA; its sole function is to improve your odds of not blowing the finals and/or arcing the output transformers should you open-circuit the speaker terminals when operating at high volume. If you're careful about this, you may omit this resistor, since it does rob some output power and can get quite warm during high-power operation. In general, never never open-circuit the outputs of any tube amplifier. Always use a dummy load (i.e. 8 ohm, 50 watt resistor) during testing.

3D: POWER SUPPLIES

3D-1: Power Supply for full stereo Integrated Amplifier

Both channels share a single power supply (see **Figure 5**). Transformer T1 provides 500-0-500 VAC for the plate supplies, 6.3 volts for most of the heaters, and 5 volts for the driver heaters. (A second 6.3V winding would have been preferred, but as designed the driver works fine at a 5 volt heater supply. More about that in the "Tweaks and Hacks" section.)

The "raw" B+ is filtered using a capacitor-input CLC pi filter, with values chosen for the best tradeoff between regulation, output voltage and ripple. A reference voltage of about 230V is supplied by two 0B2 gas-tubes in series, and is used to derive the regulated 230V line for the driver stage (via a transistor regulator in the emitter-follower configuration). These gas tubes could be replaced by series zener diode strings, but isn't recommended because of the tendency of high-voltage zeners to drift with temperature. We need a stable reference voltage here, to prevent PA grid bias from drifting. There's no need be concerned with the "noisy" aspect of the gas tubes, since the differential pairs in the PA drivers will tend to cancel out such noise.

An only slightly more involved regulator circuit supplies the 430 volt B+ for the four preamp sections. Each of these are thouroughly decoupled, first on the main power supply board and then some more on the two preamp boards. You may be asking, "Why such a fancy regulator, if you're just following it with CRC filters anyway?" If so, you have good design instincts. The preamps, by virtue of the self-levelling DC feedback design, are very forgiving about supply voltage fluctuations, and will indeed work fine anywhere between about 400 and 450 volts. A suitable series resistor would therefore be all it takes to derive their supply voltage from the 600V raw B+ line.

However, there is a potential problem during warm-up, before the preamp stages draw any plate current. Unless you appropriately increase the voltage rating of all the electrolytic capacitors on the preamp board to accomodate up to almost 700V during warm-up, you could cause serious trouble. The inclusion of a regulator consisting of two inexpensive transistors and three resistors is a much smaller price to pay.

The driver stage is direct-coupled to the PA grids, imposing special requirements. Since PA grid bias and balance are set by varying the driver currents, the well-regulated positive supply (230V) is a must. In addition, a reasonably well-regulated negative supply is needed. The reference for this is supplied by a string of zener diodes, to reduce size and cost. (We could just as well use a string of gas tubes, as we did for the positive supply. However, such a large voltage drop would have been bulky, expensive, and wasteful of power, since gas tubes typically need to draw at least five milliamps for good regulation.) Another transistor voltage-follower provides the regulated -550V rail for the drivers.

The simple negative supply regulator is actually one of the few areas of the design that I'm not completely happy with. High-voltage zener diodes tend to be quite sensitive to changes in temperature, meaning that the regulated voltage can easily drift over a range of ten volts or more. The result is a wandering grid bias; not fatal to the design by any means, but not ideal either. However, it does work adequately. You might ask why we don't use a similar approach to that of the positive regulator, where a lower voltage reference is divided up. The answer is that it is difficult (if not impossible) to find PNP transistors with the requisite 500V+ breakdown voltage.

I therefore left the circuit as-is for the published design, and am including a fix in the "Tweaks and Hacks" section. If you're a closet purist, you'll be a lot happier with the modified version.

The filament supplies for the preamps are rectified and filtered, giving reasonably clean 6.3V DC to substantially reduce hum and noise induced from the filaments to the control grids of these tubes. They actually consist of two separate half-wave supplies, one for each channel; one supply positive, and the other is negative (relative to ground). The reasons for this unusual approach are: A) Since our power meters require a ground-referenced supply, one side of the filament transformer must be grounded. This eliminates as an option the common solution of having a tapped "hum balance" pot between floating filament lines. B) This circuit is essentially a voltage doubler, allowing us to use a low-current 12-volt cooling fan such as is used for computers, etc. (an absolute necessity if enclosing the amplifier in a cabinet).

Note that the power on/off switch is a double-pole unit, with the second pole used to switch in a 3.3K "bleeder resistor" to the main positive supply. Don't omit this if you build the amplifier. It serves a dual purpose: 1) to help protect you from shocks from charged capacitors when working on the unit, and 2) to insure that the positive supply drops before the negative supply does, quickly driving the output tubes into cutoff on power-down and eliminating the possibility of nasty turn-off transients.

The "Standby" switch is also highly recommended, even if you install the "hack" to eliminate its need in actual operation. The reason is that it allows you to safely set final grid bias without any possibility of damage to the pricey 6L6GC's and associated circuitry.

3D-2: A Simple Power Supply for Standalone Preamps

If you're just building a dual stand-alone preamplifier (as for, say, a stereo magnetic phono preamp), your power supply requirements will of course be greatly reduced**Figure 5A** shows a simple supply using readily-available 12 volt, 1/2 ampere transformers (units capable of 1 ampere preferred for continuous operation). The 12 volt output from the first transformer is used to power the filaments, and is also fed to the second transformer connected as a step-up device to provide about 110 VAC. This supplies a voltage tripler circuit that provides the 400+ volts required for the preamplifier circuitry.

Filaments are again powered from a DC supply. The four diodes D4-D7 form a full-wave bridge rectifier, and CRC filters provide a sufficiently clean DC voltage to the filaments to satisfy even "sticklers".

3E: POWER METERS

After all that, the peak-output power meters are almost trivial. Aside from the rectifier diodes and regulator transistors in the power supply, it's our sole concession to solid-state technology. The meter is a basic application of the popular LM3915N LED bar-graph display driver, which shows output power (assuming an 8-ohm speaker system) in 3 dB steps from 200 mW to 100 watts. Note that this represents Peak power, the equivalent RMS power will be one-half of the peak value (for sine wave input). If the last light (representing 100 watts peak) lights too frequently, this is your cue that you're pushing the amplifier to its limits, and you'll start hearing signal degradation due to clipping.

The power requirements for the LED's and display driver are derived from a simple half-wave rectifier and filter, sourced from the 6.3 volt filament winding of our power transformer. Power requirements are very modest (under 20 milliamps under normal conditions) so this little frill doesn't cost us much in terms of resources.

Figure 9 shows the schematic of this simple yet impressive feature. (Can you imagine trying to implement something like this using tubes? The mind boggles...) Note that R702/802 is shown for an 8-ohm load. For 4 ohms, reduce to 160K; for 16 ohms, increase to 330K.

4: CONSTRUCTION HINTS

The prototype of the RA-100 was constructed on an aluminum chassis with bottom plate, and an aluminum front panel (both fromHammond). The chassis size is 17" x 10" x 2" (432mm x 254mm x 51mm) and features spot-welded corners for strength, and flanges on the underside for mounting a cover plate.

The front panel is designed to fit into a standard 19" rack, and measures 19" x 7" (483mm x 178mm). The mic/line trim pots and input jacks, Power, Standby and PA mode switches, and indicator lamps pass through both the front panel and the front edge of the chassis, securing the panel to the chassis. Strips of 1/16" steel bracing about 1«" wide and 12" long are used to provide additional structural support; without them the chassis is a bit too flimsy to support all that "iron" (transformers and filter choke).

All other controls and indicators are mounted on the upper portion of the front panel. The tone control pots are mounted to their own little PC boards, as are the LEDs for the power meters. The input and output jacks, speaker outputs, fuses and line cord are mounted on the rear apron of the chassis.

If you wish to duplicate the layout of the prototype, download the fileartwork.zip, (be sure to read "artwork.txt" included in the zipfile) which includes blueprint templates for the chassis, front panel, and rear panel. Print them out and tape to the chassis (and front panel), then use a centre-punch to locate the required holes. The front panel and rear apron layouts can be photocopied onto self-stick mylar film, which can be stuck to the panel and apron before assembly to make workable text legend "silk screens".

It is highly recommended that the pre-amplifier modules, driver modules, tone-control cards and power meters be wired on PC boards, as per the artwork provided in the artwork.zip file. Optionally, you can wire the bulk of the power supply on PC boards also. For the prototype, I got as far as making boards for the main positive supply, but had enough of PCB etching and built the remaining sections either on perfboard (positive and negative regulators) or simply siliconed the capacitors to the chassis and hardwired the remaining components to the chassis (filament supplies).

Note that the preamplifier modules and driver modules are designed for double-sided boards This can be tricky, and again is recommended only for experienced PC prototypers.

Note also: There is a polarized component shown on either side of the centre tube (12AT7) which is not used in normal applications; this was left as an option should someone wish to use the preamp in a "cathodyne" configuration. This, and its associated output connection pads are ignored in this design.

If you are interested in commercial production of this design, contact me for Gerber files of the PC boards which can be used for bulk PC board production.

Wiring tips

- Be extremely careful. This cannot be overemphasised. The voltages and currents in this project can easily kill you. Be constantly aware of this fact, and the worst you can do is make a bunch of bad smoke. Burned out components are replaceable; you're not.

- When assembling the preamplifier boards, make them mirror-images of each other; i.e arrange them so that the phono/mic preamp section is closest to the corner of the chassis (and therefore furthest from the power transformer) for both channels. If you use a different layout, it may be preferable to put both (left and right channel) phono/mic preamps on one card, and both tone control preamps on the other card.

- You may prefer, as I do, to have the power meters and tone controls symmetrical. This will mean that, for the right-channel versions, you will have to mount the pc board flipped (wiring side up), with the components surface-mounted on the wiring side. In the case of the tone controls, you will also have to reverse the components for the BASS and TREBLE sections in order to match the front-panel legends.

- Wire up the power supply, output transformers, and 6L6GC sockets first. I used an 18-position barrier strip to make things easier, with connections as follows:

  - 1 (chassis), 2 and 3 = strapped together to provide the system ground point.
  - 4 = 6.3V heater supply
  - 5, 6 = 5V driver heaters supply
  - 7 = B- (-590V) from negative regulator
  - 8, 9, 10, 11 = right channel speaker windings, wired in parallel (8=10, 9=11)
  - 12, 13, 14, 15 = left channel speaker windings, wired in parallel (12=14. 14=15)
  - 16 = regulated B+ (230V)

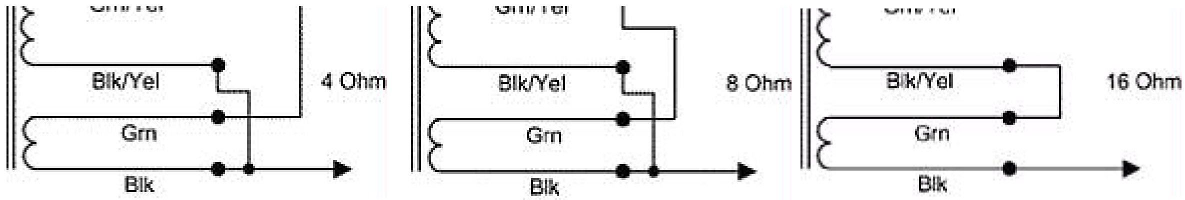
  - If you use a longer barrier strip, some of the other key lines may be similarly tied off.

Output Transformer Connections

Connect the primary wires of the transformer as follows:

Red: -------------	Raw B+
Solid brown: -----	plate of V9 (V13) = pin 3
Brown w/ trace: --	screen of V9 (V13) = pin 4
Solid blue: -------	plate of V10 (V14) = pin 3
Blue w/ trace: ----	screen of V0 (V14) = pin 4

Connect the secondary wires of the transformer as appropriate for your speaker impedance. Because of the voltage feedback method, this design does a bit better with higher impedances; for this reason the 8- or 16-ohm connections are recommended.



4 ohm — Change R517/617 to 820K; Change C503/603 to 6.8 pF

8 ohm

16 ohm — Change R517/617 to 1.6M; Change C503/603 to 3.3 pF

- Run all power wiring as twisted pairs. This includes AC input, filament supplies, B+ and B- lines, speaker lines, and the wires to the power meters. Ground returns should be grounded at the central grounding point only. Resist the temptation to use the chassis as a ground return. The only components that can be chassis-grounded are the CD, AUX, and TAPE IN/OUT jacks on the rear apron.

- Run all audio lines using shielded cable. Connect the shield to ground at one end only; do not use the shield as a ground return. In most cases, it is advisable to ground the cable at the "source" end; this will aid troubleshooting down the road. For the rear apron jacks, ground the cables at the jack end (even for the TAPE OUT jacks). Cut away the shield at the other end, and shrink-wrap or tape the cut to prevent the little "fuzzies" from shorting against anything.

- The phono and mic inputs require special handling. The input jacks should be isolated from ground using insulated mounting hardware. This is one of the few cases where the shield of the cable is connected at both ends. Connect to the jack's ground return at the jack end, and to a common tie point with the grounds going to the DPDT switching mic jack (or dedicated 3-pole switch if you want to implement the Ceramic phono option also). The output of your switching circuit should be connected to the input of the preamp using shielded cable, again with the switch end grounded to the common tie point, and the preamp end grounded to the ground pad nearest the input terminal.

- Try to insure that the lines to the TAPE, CD and AUX inputs are all approximately the same length, so that your Ctweak frequency-response correction will equally affect all inputs.

- The lines to the tone card should be kept as short as possible, preferably using low-capacitance cable.

- The lines to the input jacks on the rear apron may optionally be two-conductor shielded cable, one for each channel. This is what I used on the prototype. However, if you want to improve on the published performance spec for channel separation, run separate single-core cables instead. Use single cables on the phono inputs and runs to the tone cards, regardless.

- Don't be careless with any lines that carry supply voltages. Make sure that all connections are well-insulated. Don't let any uninsulated high-voltage point come any closer to any other point (including, of course, the chassis) by less than about 1/4" (6 mm) worst-case. Be especially careful with the lines going to pins 4 (screen) and pins 5 (control grid) of the 6L6GC's. Arc-over between these pins can be fatal to your output tubes.

Other tips

- Don't skimp on capacitor voltage ratings. Capacitors in the power supply circuit are the readily available 450V and 350V units, connected in series to double the voltage handling capability; the price for this facility is that 2 meg equalising resistors are a must -- don't omit them. Same goes for the equalising resistors across the series-connected rectifier diodes. For the main positive capacitor bank, use capacitors designed for high current capability. Use only new capacitors of the same brand and manufacture in any given bank. After a few minutes of operation, (and periodically during construction and testing) check the voltage drops across each series pair, if there is an imbalance of more than, say 10%, replace the pair.

- If you'll be running the amplifier open-frame, no additional ventilation should be required. However, if you enclose it in a cabinet, forced air cooling (using a fan) is an absolute necessity. This thing runs hot! If installing it in a rack, make it the top-most unit to prevent cooking other modules in the rack. The top cover of the rack should be well vented; otherwise, fans will be required.

- Keep this project well out of reach of small children and pets at all times. I keep coming back to the safety aspect, not because I'm paranoid, but because there is real danger in carelessness.

- The power transformer can be smelly for awhile after building the amplifier, as the varnish used in manufacture cures. After about a hundred hours of operation, (burn-in) the smell should pretty much dissipate. It's a good idea to tighten the screws at the corners of each "iron" after the burn- in period to prevent the possibility of buzzing. (A small amount of transformer buzz on the power transformer is normal, due to the capacitor-input filter.)

Adjustment

Assuming that everything goes well, there is not much to adjust. Turn the unit on (in Standby mode) and let it warm up for at least a minute. Check the main supply voltages:

- +600 V raw positive: will measure closer to 700 volts under no load (standby mode)
- +430V positive regulated
- +215V positive regulated
- -700V negative unregulated
- -590V negative regulated

Measure the grid voltages of the 6L6GC's, one channel at a time. A convenient point to measure this is at the anodes of the protection diodes on the driver card, the end sitting right near the centre of the card. Adjust the BIAS and BALANCE controls to give -72 volts at each grid. The BALANCE control will be quite symmetrical, a fact that you can use to get the voltages to converge quite rapidly.

Only after you have gotten the grid bias "in the ballpark" should you even consider applying B+ to the 6L6GC's by flipping the Standby switch. Also, get into the habit of always having a load on the speaker outputs, even if you are not applying any signal. It just takes one bad transient to ruin your tubes or transformer (not to mention your day)

Cathode current vs. Grid voltage

If your tubes are reasonably well-matched, the cathode current on each tube of each push-pull pair should be within about 20%. If not, your tubes aren't very well matched. This is not necessarily fatal, you can trim the BIAS and BALANCE pots to get the cathode currents approximately equal (in the range of 40-50 milliamps) by measuring the voltage across the 10- ohm cathode resistors. (50 milliamps will give a reading of 0.500 volts).

In any case, only the most discerning ears will hear any difference in reproduction quality, especially at lower volumes, and especially in "clean" mode because of the self-correcting nature of negative feedback.

At the recommended setting (40-50 mA cathode current, or -72 volts grid bias), the amplifier will actually be operating in Class A mode (both tubes conducting at all times) at power outputs under a few watts. Beyond that it will go into Class AB_1 mode, and the tubes take turns going into cut- off beyond a certain signal level. The grid bias essentially controls the point at which this "crossover" takes place. Decreasing the grid bias raises this point, resulting in less distortion. I don't recommend going any lower than -60 volts. At about -55 volts the plates of the tubes start to glow red. Not a good thing. Even at -60 volts, the power transformer will get too hot to touch after a couple hours.

At the other extreme, say around -85 volts, the tubes are sitting closer to cutoff and the amplifier gets a lot more efficient (less heat) but crossover distortion becomes quite noticable, even on a scope.

Considerable experimentation has shown the recommended -72V grid bias to be the best compromise between the efficiency and distortion factors. Still, there is some leeway here. If you want absolutely the best sound quality you can get, and don't run the amp for more than an hour or two at a time, set bias for about -60 volts. On the other hand, if you want to run the amplifier continuously, a bias of -80 is recommended.

A Word About Fuses

I'm sure you know what's coming here, but I just have to say it. Don't replace the fuses with ones of larger ratings, unless you absolutely positively know what you're doing. The main AC fuse should be no larger than 4 ampere, slow blow (MDL). In fact, 3 ampere fuses are recommended. Go to 4 amperes only if you're using a lower grid bias and find that 3 ampere fuses are regularly blowing. Even then, don't be upset with me if you ruin your power transformer. If 3 amp fuses blow at the recommended grid bias, even at extended high volume settings, something is wrong and you should see that it gets corrected instead of just putting a "crowbar" into the fuse-holder. Fuses are designed to protect your equipment, and you.

The "raw B+" fuse should under no circumstance be rated for greater than 0.6 ampere (600 ma.), fast blow (AGC). Be sure you use a fuse holder that is capable of withstanding the over 600 volt potential to chassis ground. (It should also be noted that the AGC series is not UL or CSA approved for use over 240 volts AC. The rationale is that at higher voltages, arcing could possibly occur across the burned ends of the fuse wire. However, my thought on this is that any protection is better than no protection at all.)

If you need to replace the B+ fuse, turn the amplifier off first.

A HI-FI VACUUM TUBE AMPLIFIER

by Fred Nachbaur, Dogstar Music ©1998 - 2002

5: PARTS LIST, RA-100 REFERENCE AMPLIFIER

rev. 05/01

(See NOTE below for some sources.)

PART	DESCRIPTION	NOTES
C ------		**CAPACITORS**
C1-4	47 uF 450V electr.	cer/mylar unl. noted
C5-8	220 uF 350V electr.	
C9	47 uF 450V electr.	
C10, 11	10 uF 450V electr.	
C12	47 uF 450V electr.	
C13, 14	10 uF 450V electr.	
C15	0.01 uF 600V	
C16, 18	4700 uF 16V	
C17, 19	2200 uF 16V	
C20, 21	47 uF 450V electr.	
C22-24	10 uF 450V electr.	
C101,201	0.1 uF 50V	
C102,202	2.2 uF 50V elect.	
C103,203	10 uF 450V elect.	
C104,204	47 uF 50V electr.	
C105,205	10 uF 50V electr.	
C106,206	20uF 250V electr.	
C107,207	1 uF 450V electr.	
C108,208	100-1000 pF (as needed)	
C109,209	1 uF 250V electr.	
C110,210	47 pF 250V	
C111,211	6800 pF 250V	
C112,212	2200 pF 250V	
C113,213	1 uF 350V electr.	
C301,401	0.1 uF 50V	
C302,402	2.2 uF 50V elect.	

C303,403	10 uF 450V elect.	
C304,404	47 uF 50V electr.	
C305,405	10 uF 50V electr.	
C306,406	20uF 250V electr.	
C307,407	1 uF 450V electr.	
C308,408		not used
C309,409	.01 uF 100V mylar	
C310,410	1000 pF mylar	
C311,411	4700 pF mylar	
C312,412	15 pF ceramic	
C313,413	1000 pF mylar	
C314,414	4.7 uF 50V electr.	
C315,415	2.2 uF 350V electr.	
C501,601	0.05 uF 400V low leakage	
C502,602	0.05 uF 400V low leakage	
C503,603	4.7 pF (for 8 ohm)	sub 6.8 pF for 4 ohm; 3.3 pF for 16 ohm
C504,604	1 uF, 450V electr.	
C505,605	0.1 uF	
D ------		**SEMICONDUCTOR DIODES**
D1-4	1N5408	ECG/NTE 5809
D5-8	1N4007	ECG/NTE 125
D9	1N4002 or better	ECG/NTE 116
D10-13	1N4984 120V 5W zener	ECG/NTE 5158A
D14	1N4983 110V 5W zener	ECG/NTE 5157A
D15, 16	1N5408	ECG/NTE 5809
D501-504	1N4002 or better	ECG/NTE 116
D601-604	1N4002 or better	ECG/NTE 116
D505,605	1N4984 120V 5W zener	ECG/NTE 5158A
F ------		**FUSES**
F 1	3A Slo-blo (MDL) fuse and holder	Holder: Mode 55-800-0, etc.
F 2	0.5A fast-blo (AGC) fuse/ holder	Holder: Mode 55-800-0, etc.
J ------		**JACKS AND CONNECTORS**
J P1	3-wire AC plug and cord 15A	Mode 30-031-0

J1, J3	Red banana plugs (speaker)	Mode 31-462-0
J2, J4	Black banana plugs (speaker)	Mode 31-461-0
J101	RCA jack	Mode 24-181-0
J102	1/4" phone jack w/DPDT switch	auto-switch option Mode 24-697-0
J201	RCA jack	Mode 24-181-0
J202	1/4" phone jack w/DPDT switch	auto-switch option Mode 24-697-0
J301-309	RCA jack	Mode 24-181-0
J401-409	RCA jack	Mode 24-181-0
L ------	**INDUCTORS**	
L 1	3 Hy 500 mA filter choke	Hammond 193N
LED ----		**LIGHT EMITTING DIODES**
LED 1	Green T1-3/4 LED	Mode 55-555-0
LED701-703	Green T1-3/4 LEDs	for power meters
LED801-803		" Mode 55-555-0
LED704-706	Amber T1-3/4 LEDs	" Mode 55-553-0
LED804-806		"
LED707-709	Red T1-3/4 LEDs	" Mode 55-552-0
LED807-809		"
LED710,810	Green T1-3/4 LEDs, hi brightness	" Mode 55-555HB-0
NE -----		**NEON PILOT LAMPS**
NE 1	125 VAC Amber neon pilot	Mode 55-443-0
NE 2	125 VAC Red neon pilot	Mode 55-442-0
PCB ----		**PRINTED CIRCUITS**
PCB 1, 2	Preamp / Tone amp PCBs	(artwork available from website)
PCB 1A, 2A	RIAA / Ceramic EQ boards	"
PCB 3, 4	Driver PCBs	"
PCB 5. 6	Tone Control PCBs	"
PCB 7, 8	Power meter PCBs	"
Q ------		**SILICON TRANSISTORS**

Q1 - Q3	2SC4273 (NPN 700V 1A TO-220)	ECG/NTE 51
Q4	PNP 400V 1A, TO220	ECG/NTE 38
R -----		**RESISTORS**
R1	10R 5W wirewound	1/4 5% unless noted
R2-17	2M	
R18	1.8K 2W	
R19	47K 2W	
R20-23	33K 2W	
R24	680K 1/2W	
R25	100K 1W	
R26	91K 1W	
R27	100K 1W	
R28	33K 10W wirewound	
R29, 30	470R	
R31-34	3.3K	
R35	470R	
R36, 37	1R 5W wirewound	
R38, 39	330R 1W wirewound	
R40	3.3K 5W wirewound	
R101, 201	100K	
R102, 202	100K	
R103, 203	47K	
R104, 204	1.5K	
R105, 205	27K	
R106, 206	330K 1/2W	
R107, 207	180K 1/2W	
R108, 208	91K 1/2W	
R109 ,209	160K	
R110, 210	51K	
R111, 211	22K	
R112, 212	100K	
R113, 213	47K	
R114, 214	27K	
R115, 215	4.7K	
R116, 216	47K	
R117, 217	10K	

R118, 218	150K	
R119, 219	220K	
R120, 220	1K	
R121, 221	1.2K	
R122, 222	43K	
R123, 223	2K	
R124, 224	100K	
R301, 401	not used	
R302, 402	100K	
R303. 403	47K	
R304. 404	1.5K	
R305, 405	27K	
R306, 406	330K 1/2W	
R307, 407	180K 1/2W	
R308, 408	91K 1/2W	
R309, 409	160K	
R310, 410	51K	
R311, 411	22K	
R312, 412	100K	
R313, 413	470K	
R314, 414	27K	
R315, 415	4.7K	
R316, 416		not used
R317, 417	10K	
R318, 418	47K	
R319, 419	470K	
R320, 420	47K	
R321, 421	47K	
R322, 422	47K	
R323, 423	18K	
R324, 424	18K	
R325, 425	22K	
R326, 426	22K	
R327, 427	18K	
R328, 428	2K	
R329, 429	10K	
R501, 601	47K	
R502, 602	220K	

R503, 603	470K	
R504, 604	470K	
R505, 605	100K 2W	
R506, 606	100K 2W	
R507, 607	2.2K	
R508, 608	2.2K	
R509, 609	2K	
R510, 610	2K	
R511, 611	22R	
R512, 612	220K 2W	
R513, 613	10R 2W	
R514, 614	10R 2W	
R515, 615	100R 5W ww (opt)	
R516, 616	10R 2W	
R517, 617	1.2M (for 8 ohm)	sub 820K for 4 ohm; 1.6M for 16 ohm
R518, 618	47K	
RV -----		**POTENTIOMETERS**
RV 1	100K Stereo, Log taper (Vol)	available from Radio Shack
RV 2	500K Linear taper (Balance)	Mode 62-248-0
RV101, 201	100K Linear (mic gain)	Mode 62-247-0
RV301-303	500K Linear (tone controls)	Mode 62-248-0
RV401-403	500K Linear (tone controls)	Mode 62-248-0
RV501, 601	50K trimpot (classic gain)	
RV502, 602	500R trimpot (bias balance)	
RV503, 603	500R trimpot (bias current)	multiturn recommended
S ------		**SWITCHES**
S 1	DPDT 6A toggle (power)	Mode 42-323-0
S 2	SPST 3A toggle (standby)	Mode 42-310-0
S 3	2P4T rotary (source)	Mode 48-513-0
S 4	2P3T rotary (tape monitor)	Mode 48-514-0
S 5	3PDT miniature toggle (PA md)	Mode 41-251-0
S 6	DPDT miniature toggle (Mono)	Mode 41-243-0
S101, 201	DP3T rotary switch	Cer/mag/mic option - Mode 48-514-0

S301, 401	DPDT mini toggle (tone bypass)	Mode 41-243-0
T ------		**TRANSFORMERS**
T 1	Power Transformer	Hammond 282X
T501, 601	Output Transformer	Hammond 1650N
V ------		**VACUUM TUBES (VALVES)**
V1, 3, 4, 6	12AX7A	dual triode
V2, 5, 7, 11	12AT7A	dual triode
V8, 12	6CB6A	sharp cutoff pentode
V9, 10	6L6GC	matched pair preferred
V13, 14	6L6GC	matched pair preferred
V15, 16	0B2	gas regulator (or use 5W 230V zener string)
VR -----		**NEON REGULATORS** (Mode 55-120-0)
VR101-3 3x NE-2		matched with VR201-3
VR201-3 3x NE-2		matched with VR101-3
VR301-3 3x NE-2		matched with VR401-3
VR401-3 3x NE-2		matched with VR301-3
z- -----		**MECHANICAL PARTS**
z-HS1-3	TO-220 isolated heatsinks	

Chassis, 433 mm x 254 mm x 52 mm (17" x 10" x 2")
Bottom plate for chassis, 433 mm x 254 mm (17" x 10")
Front Panel 484 mm x 178 mm (19" x 7")
Re-inforcement struts to strengthen chassis/ front panel approx 310 x 25 mm
Hookup wire (600V)
Shielded cable, single and double conductor
Perf-board as required for power supply sections
Mounting hardware (bolts, nuts, washers, L brackets for mounting perfboard)
Strain relief for power cord

Knobs for potentiometers (several types available from Mode Electronics)
Terminal strips, phenolic solder type
Barrier strips (for power and speaker distribution)
Tube sockets (4 ea. octal, 2 ea. 7-pin chassis-mount, 2 ea. 7-pin PC mount, 6 ea. 9-pin PC mount)
RCA jumper cables for FX bypass
Enclosure (use fan if unit is enclosed!)
75mm or 100mm fan, 12V 100 mA (if enclosed)

Note:

Many parts such as connectors, pots, etc. can be obtained from any Mode Electronics distributor (visit www.mode-elec.com for the distributor nearest you). The Mode part number for many such components is given above. Tubes and sockets can be ordered online from many suppliers; a search for tubes and sockets will return many such suppliers. The transformers, filter choke, and chassis components can be ordered from any Hammond Manufacturing distributor (www.hammondmfg.com).

6: TWEAKS AND HACKS

A: Limiting Frequency Response

The "FX Loop" jacks can be used to insert a series capacitor to limit low-frequency response. A value of 0.1 uF will typically be desirable to eliminate subsonics that could un-necessarily rob you of available power.

Similarly, a capacitor can be placed in parallel (to ground) to limit high-frequency response, should this be desirable. Experimentation is the best approach, you might find that a couple hundred picofarads to ground will help eliminate the harshness of record pops on old vinyl or 78's.

B: Increasing Driver Filament Voltage

I admit it, this one is definitely in the "hacks" genre. (See HACK1L.GIF.) It is a technique commonly used to squeeze a bit more life out of oscilloscope CRT's by increasing the effective RMS filament voltage.

This can be used to advantage to boost the effective filament voltage to the driver tubes. While they run just fine on 5VAC, the do warm up a bit slower than the rest of the tubes as a result. This means that the output 6L6GC's can draw excessive current for a few seconds on power-up, if the user forgets to flip off the Standby switch during warm-up. The result is often a blown fuse.

By speeding up the warm-up time of the driver tubes, the possibility of trouble is greatly decreased. Allowing warm-up before applying B+ to the power amp is still recommended, bu is no longer as vital.

Interestingly, a side effect of this hack is that it slightly reduces the background noise (harmonics of 60 Hz.) ... at least on the prototype.

C: Stabilising the -590V Negative Regulator

The high-voltage zener diodes used to obtain the negative 590 volt reference tend to be rather unstable with temperature. This can cause the regulated negative voltage (and therefore the finals' grid bias) to drift considerably.

A substantial improvement is afforded by the modified circuit shown in TWEAK2L.GIF. The transistor is a general-purpose high-voltage (300 volts VBCE) small-signal unit. The 12 volt, 1/2 watt zener diode is used as a reference instead. Note that the four 120V diodes are still used, except no longer as a reference. Rather, they are kept as level-shifters to prevent excessive voltage across the added regulator transistor.

A 500-ohm trim pot is also added to provide adjustability. Set it for an output voltage of -590 volts. Depending on individual component tolerances, etc. you might have to tweak the 2.7k series resistor.

D: Separate filament supply for Current-Source Pentodes

The filament arrangement for the driver circuit is technically in violation of maximum filament-to- cathode voltage as regards the 6CB6A pentode. If this bothers you, you can use a separate low- power transformer to supply filament power for these tubes. No need to rectify and filter to DC, since there is no audio on these units. Reference the filament to the -590 V rail by tying one end of the filament to the rail. The transformer used should be either 12.6VAC, 0.3A or better (filaments in series) or 6.3VAC, 0.6A or better (filaments in parallel)

I tried it with a Hammond 191F24 (windings in parallel for 12.6VAC, 0.36A). There was no measurable difference in performance, but it might set your mind at ease if you're concerned about possible arc-over or filament emission.

If you implement this tweak, keep the supply to the 12AT7A filaments the same as before, but reference it to the plate of the 6CB6A instead of to the tap on the resistive divider across the negative supply.

E: Tube Substitutions

I don't recommend substituting any tubes in the preamp section, except that the 12AT7 can be replaced with the European equivalent ECC81 or with 12AZ7A, and the 12AX7A's with ECC83.

In the driver section, several common pentodes will work in place of the 6CB6, including 6AU6 (a good choice) and 6BA6 (a workable but slightly less stable alternative). You will have to considerably increase R511/R611 in order to get the required bias adjustment range.

I haven't tried any output pentodes other than the 6L6GC, but quite a few devices (including other 6L6GC variants as 6BG6GA or 807, 6CA7/EL34, and 6550/KT88 or KT90) should work just fine here. An excellent tube, if you can find it at less than $200 a pair, would be the 8417.

Appendix: Specifications

Here are the measured performance specifications of the prototype "RA-100" Vacuum-tube Reference Amplifier. Voltages are in RMS unless indicated otherwise. The output tubes used for the measurements were un-matched, inexpensive Chinese 6L6GC's, testing done after a burn-in of approx. 100 hours.
The THD measurements were repeated October, 2001 using a matched set of premium Tesla 6L6GC's, for comparison.

Physical size: (not including enclosure)	
Dimensions:	**483 mm (19") W x 178 mm (7") H x 280 mm (11") D**
Weight:	**16 kg. (35 lb.)**

Maximum Power Output:		
Test Conditions: Input to CD/Aux/Tape inputs, Test Freq. = 1kHz, Input signal = 1V sine wave, Tone controls = bypassed, Vol. = max., Finals Grid Bias = -72V, "Clean" mode, Supply = 120 VAC.		
Impedance	Each Channel (singly)	Both Channels (together)
4 ohms	**50 W RMS**	**100 W** (short-term) **85 W** (cont.)
8 ohms (design centre)	**54 W RMS**	**108 W** (short-term) **90 W** (cont.)
16 ohms	**60 W RMS**	**120 W** (short-term) **100 W** (cont.)

Frequency Response:	
Test Conditions: CD/Aux/Tape inputs, Input = 0.1 V sine wave, Tone controls = bypassed, Volume = maximum, Mode = "Clean" Finals Grid Bias = -72V, no inline filtering at FX loops.	
Low 3 dB corner	**7 Hertz**
Flat +/- 0.5 dB	**12 Hz - 25 kHz**
High 3 dB corner	**34 kHz**

Power Consumption:			
Test Conditions: Input to CD/Aux/Tape inputs, Test Freq. = 1kHz, Input = 1V sine wave, Tone controls = bypassed, Volume = max., output load = 8 ohms resistive, Supply Voltage = 120 VAC. ***Phase factor not considered.***			
Condition:	Bias = -80V	Bias = -72V	Bias = -60V
Standby:	**- - - -**	**138 V-A**	**- - - -**
Quiescent:	**245 V-A**	**286 V-A**	**375 V-A**
Full Power:	**456 V-A**	**480 V-A**	**518 V-A**

Channel Separation:
Test Conditions: As above. 54 Watts output from left channel.
37 dB typical

Sensitivity:			
Definition: Continuous sine-wave signal (volts RMS) required for full output at maximum volume setting. **Test Conditions:** As above, test frequency = 1000 Hz.			
Input:	Volts RMS	Voltage gain	Freq. Response
CD/Aux/Tape	**1.00 V**	**26 dBv**	**as above**
Mag. Phono: (at 1000 Hz.)	**12 mV**	**65 dBv**	**per RIAA**
Mic/Line, Min. gain	**640 mV**	**30 dBv**	**as above**
Mic/Line, Max. gain	**4.4 mV**	**73 dBv**	**3dB down at 16 kHz.**

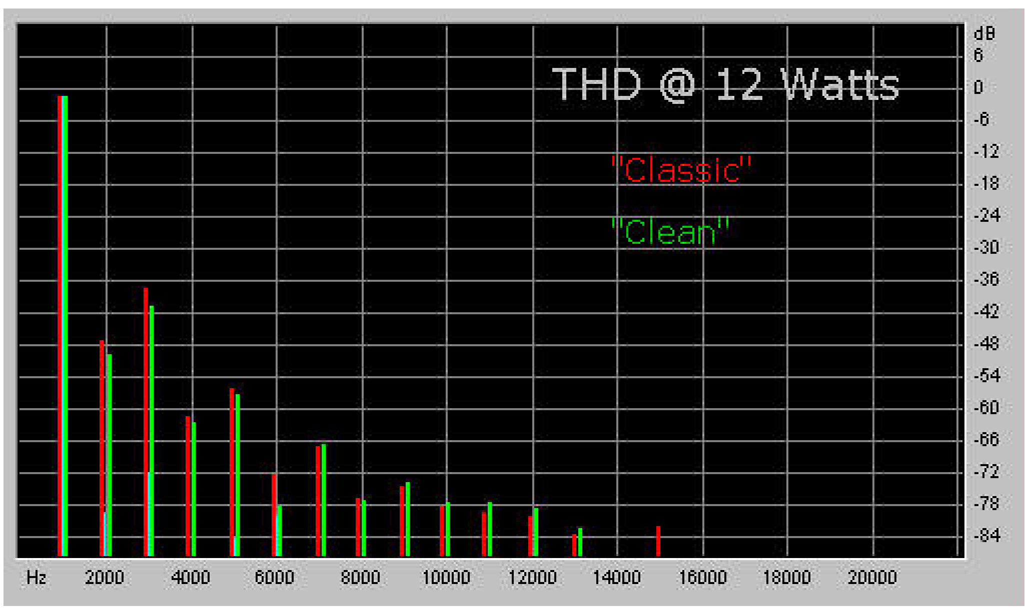
Hum and Noise:			
Definition: RMS power and "Equivalent Input Voltage" of total hum and noise (unweighted), average of both channels. dB equivalents relative to full output power (54 Watts)**Test Conditions:** As above, no shielded enclosure, inputs open.			
Input:	Volume = min.	Volume = max.	Equiv. Input
CD/Aux/Tape Inputs:	**3 µW** (-72 dB)	**435 µW** (-51 dB)	**2.9 mV**
Mag. Phono Input:	**3 µW** (-72 dB)	**903 µW** (-48 dB)	**50 µV**

Tone Control Range:			
Test Conditions: Volume = 50%, Input = CD, adjusted for 1 W out with all tone controls centred. Test frequencies = peak response.			
Control:	Test Freq:	Minimum:	Maximum:
Bass	**30 Hz.**	**-14.6 dB**	**+15.4 dB**
Midrange	**600 Hz.**	**-11.4 dB**	**+11.5 dB**
Treble	**4 kHz.**	**-12.3 dB**	**+8.25 dB**

Harmonic Distortion:

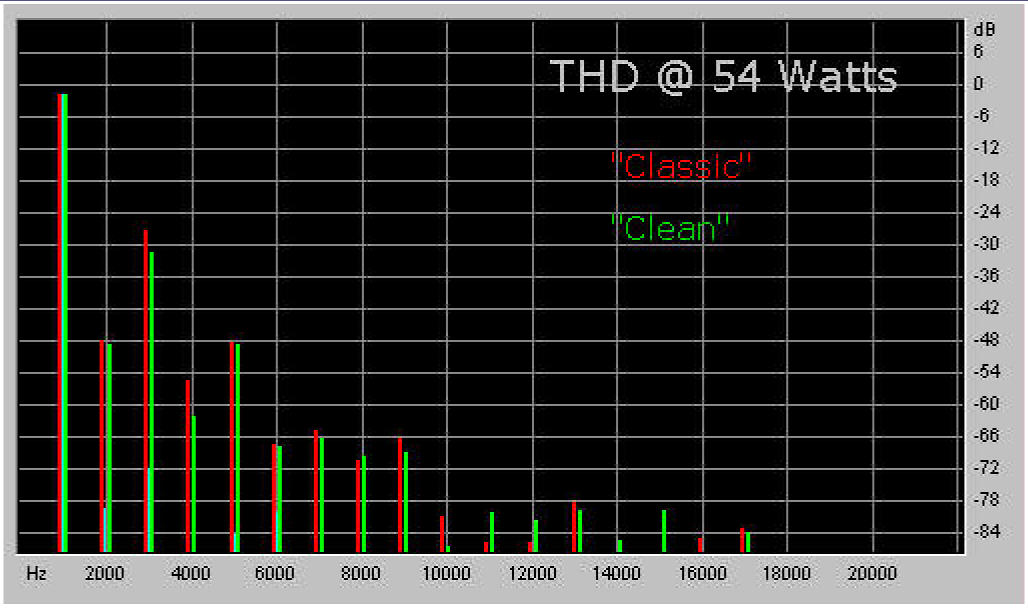
Re-done Jan. 2001 to correct errors in methodology

Test Conditions: Input = Mic/Line @ 50% gain, volume set for 12 W output, tone controls bypassed. Test freq. = 1000 Hz.
Output tubes: unbranded Chinese 6L6GC



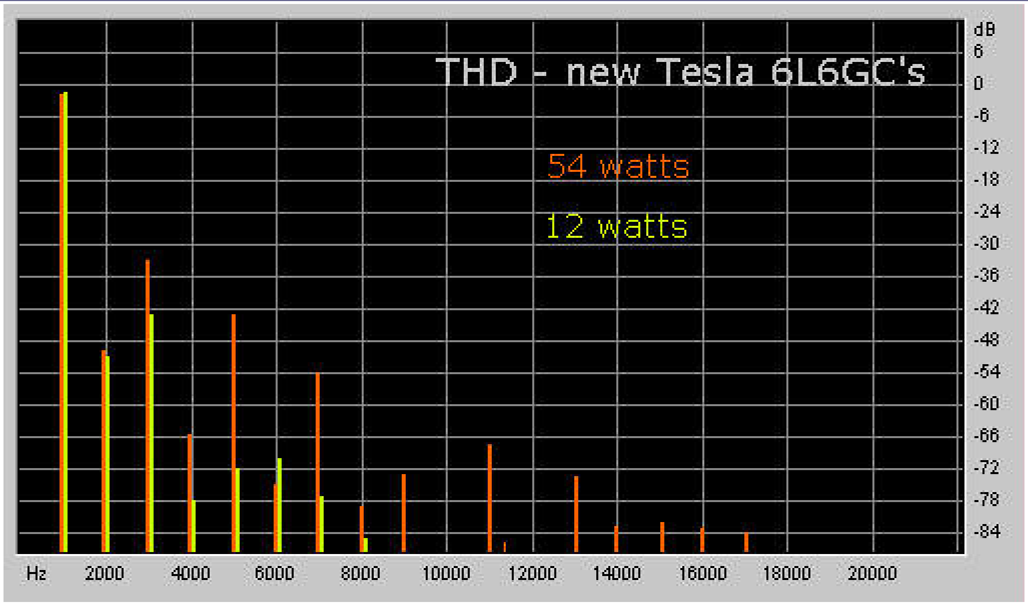
Most significant term is 3rd harmonic, at -37 dB ("classic")
THD (by voltage) [classic] = **1.49%** [clean] = **0.96%**

Test Conditions: Input = Mic/Line @ 50% gain, volume set for 54 W output, tone controls bypassed. Test freq. = 1000 Hz.
Output tubes: unbranded Chinese 6L6GC



Most significant term is 3rd harmonic, at -26 dB ("classic") or -32 dB ("clean").
THD (by voltage) [classic] = **4.51%** [clean] = **2.88%**

Test Conditions: Input = Mic/Line @ 50% gain,
volume set for specified output (54, 12 watts)
Tone controls bypassed, "clean" mode. Test freq. = 1000 Hz.
Output tubes: new, matched Tesla/JJ 6L6GC

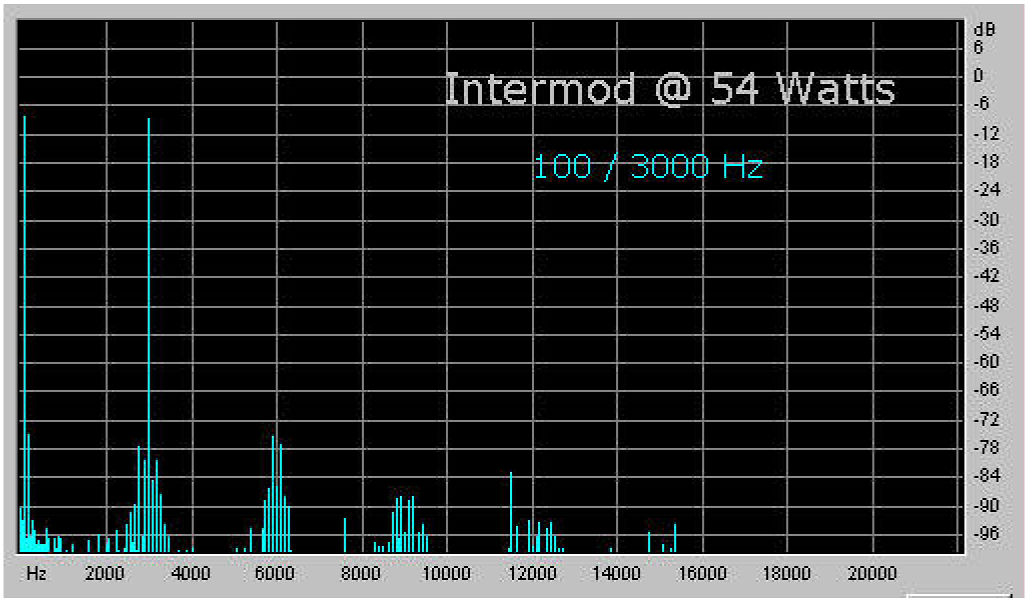


Most significant term is 3rd harmonic, at -30 dB (full 54 W output) or -40 dB (at 12 W output).
THD (by voltage) [54 Watt] = **2.38%** [12 Watt] = **0.76%**

Intermodulation Distortion:

Re-done Jan. 2001 to correct errors in methodology

Test Conditions: Input = Mic/Line @ 50% gain, volume set for 54 W output, tone controls bypassed, Clean mode.
Test freq. = 100, 3000 Hz. (equal amplitudes)
Output tubes: unbranded Chinese 6L6GC



This measurement is in some ways a superior measurement of amplifier distortion, as it demonstrates the amount of "mixing" that occurs due to non-linearity. This design exhibits very low IMD in both "clean" and "classic" modes, presumably thanks at least in part to the accurate differential phase inverter.

NOTES: