The College Budget Tube Amp

Advanced Audio Electronics
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1. Introduction

Some say that playing the guitar is a dying art, but recent statistics tell otherwise. A report from research firm IBISWorld (which tracks guitar manufacturing in the U.S.) shows consecutive growth in the last five years and a projected upswing through at least 2022 [1]. Further, statistica.com reports that guitar pedal sales have been on the rise since 2013 [2]. While advancements in solid state electronics have shrunk down equipment size and cost, many professionals harken back to the "glory days" of electric guitar—the tube era. At one time, tubes were the only option for electronic amplification; now, the term "tube sound" exists to generally describe the specific characteristics that make tube amplifiers sound pleasant to the ear. Tube amplifiers can generate a great deal of gain, and for guitar and bass, they are commonly designed to "soft-clip" the signal to produce second-order harmonics, yielding that pleasant sound. Solid state devices though—transistors and op-amps—are notoriously "clean," since they operate in a linear current-voltage region (as opposed to nonlinear I-V operation of tubes), and therefore do not naturally produce the preferred harmonics that tubes generate when pushed. Another often-overlooked contributor to the pleasurable and nostalgic sound of the tube era is a phenomenon known as crossover distortion, or tube compression, happening in the power amplification stage, visualized by a delay in the zero-crossing of a signal. According to Jack Sondermeyer of Peavey, *"clipping in a tube amplifier sounds less 'dirty' than a solid state amplifier under the same conditions, because a large portion of the overloaded output waveform is forced or compressed into the severe crossover distortion region,"* sounding more musical and "cleaner" than solid state at overload [3]. The tube compression, or distortion near the crossover, reduces the peak clipping yielding a more tolerable series of harmonics, also contributing to what makes tubes aurally preferable.

This report details the design and implementation of an all-in-one, plug-and-play tube amplifier emulation, comprised of three solid state sub-circuits: a tube preamplifier emulation, a tube compression (crossover distortion) emulation, and a speaker simulator. The preamplifier makes use of a tube-to-solid-state conversion theory called the Fetzer Valve by runoffgroove.com [4], the tube compression is emulated by a replication of Sondermeyer and Peavey's Bandit amplifier (which aimed to accomplish the same task), and the speaker simulator provides a line-level output of the circuit to a PA system for a quick and easy setup. The speaker simulator is inspired by Tom Scholz, who included such a device in the well-known Rockman pedal as well as in Boston's stage setup to maintain a semi-consistent sound across a wide range of venues. A brief overview of tube theory will be discussed, followed by a detailed description of the three sub-circuits.

A tube by itself consists solely of a positively charged plate and relatively negatively charged cathode, conducting directionally for either the positive or negative half of an AC signal. A triode, on the other hand, is what the term "tube" typically refers to. It consists of the plate and cathode, but with a third terminal called the grid that is biased to conduct current through to the massively-positively-charged plate within a designed operating region. With a grounded cathode, at the lower/negative cutoff, electrons are trapped in the grid and thus do not flow to the plate, while at the upper 0V limit the maximum amount of current flows to the plate. Since the plate is so positively charged, high output gain (V_{out}/V_{in}) is easily achievable. Several stages of this biased-input triode configuration in succession can yield a great deal of gain, ideal for amplification. A look at the load lines of a triode shows that the distance between the points intersecting grid voltages increases nonlinearly with linearly increasing grid voltage. This is where the desirable harmonic distortion of tubes arises, and thus the motivation for designing a solid state version that emulates this nonlinear behavior [5].

When the grid is driven to the point that it becomes positive with respect to the cathode, the triode is overbiased and begins to act as a forward-biased diode, having the effect of clipping the "bottom," or zero-crossing area, of a rectified signal. In a class B system where each half of the waveform is gained independently, clipping the portions of the waveforms near the zero-crossing has the effect of "spacing out" the peaks. When the halves of the waveform are recombined, the result is easily identifiable crossover distortion between peaks.

In this design, JFETs are chosen as the solid state emulation device for the preamp, while op-amps are used to emulate the class B crossover. Since the terminals of transistors and op-amps do not exhibit the same voltage biases as the plate, grid and cathode, the phenomena of tube amplification/clipping and tube compression/crossover distortion must be emulated independently as sub-circuits. Lastly, as part of the design, the output may be taken from the preamp, the preamp into the class B crossover distortion module, or from the speaker simulator which sees the preamp into class B as its input.

2. JFET Preamplifier

2.1 Mesa Boogie DC-3

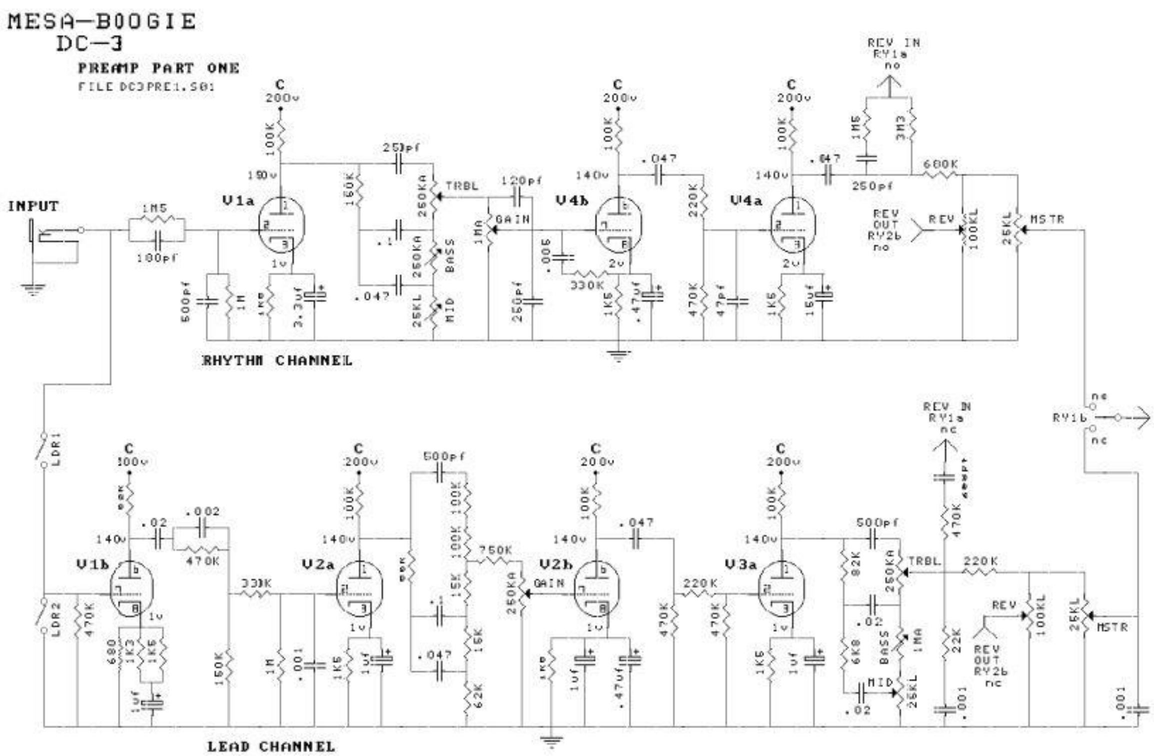


Figure 2: Mesa Boogie Dual Caliper DC-3 Preamp Schematic

The Mesa Boogie DC-3 inspired the JFET preamp seen below in Figure 2.1. While the DC-3 has both a lead and rhythm channel, our JFET emulation is based solely on the lead channel. Due to the JFET's low gain compared to a vacuum tube, the design required a minimum of four JFET stages in order even amplify the guitar signal to the appropriate line level.

2.2 JFET Conversion

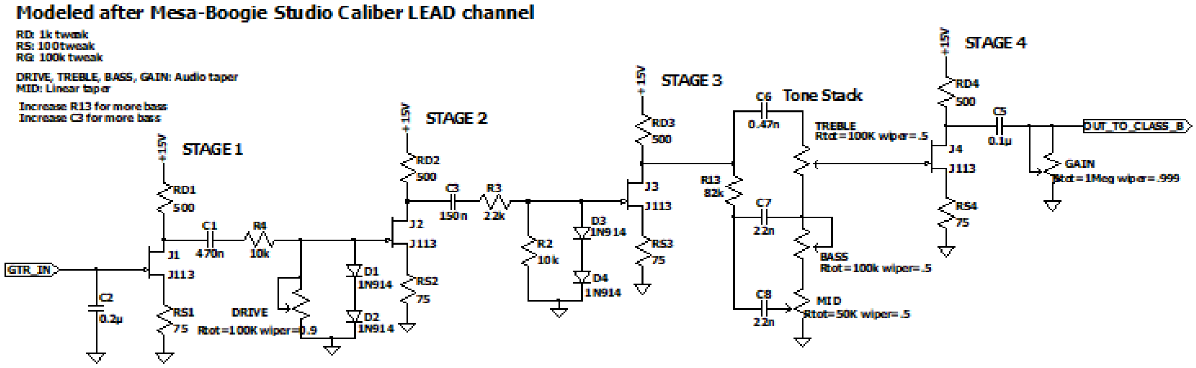


Figure 2.1: JFET Preamp Schematic

The conversion from vacuum tubes to JFET's rests upon the work done on the "Fetzer Valve", a simulation of a Fender 12AX7 input stage using JFET's. [4]

The basis of the conversion, as stated on the Fetzer Valve, is that the "plate voltage of a triode is related to the grid voltage by the three-halves power law, which produces both even and odd harmonics. As the signal level increases, the 2nd harmonic rises, followed by the 3rd harmonic, and then higher ordered harmonics start to appear."

To account for this relationship, very particular source and drain resistor values must be chosen in relation to each JFET's specific pinch-off voltage (V_P) and I_{DSS} (drain current for 0V bias).

The calculations are as follows:

$$Rs = 0.83 * |Vp| / Idss$$
$$Rd = 0.9 * (Vcc - 2*|Vp|) / Idss$$

Equation 2: Calculations based on Dimitri Danyuk's article "Triode Emulator" (as presented at the 116th Audio Engineering Society Convention, May 2004 in Berlin, Germany). [4]

Below is a table of our the JFET's V_P and I_{DSS} from the project build, along with their proposed source and drain resistances. Due to the small fluctuations, trim potentiometers have been substituted for all drain and source resistances. These potentiometers can and should be trimmed for the best sound.

	|Vp| (V)	**Idss (mA)**	**Rs (Ohms)**	**Rd (Ohms)**

J1	1.83	16.4	92.6	622.3
J2	1.82	17.0	88.9	601.4
J3	1.82	18.0	83.9	568.0
J4	1.81	18.3	82.1	559.7

Table 2: Specific JFET values

2.3 Stage 1

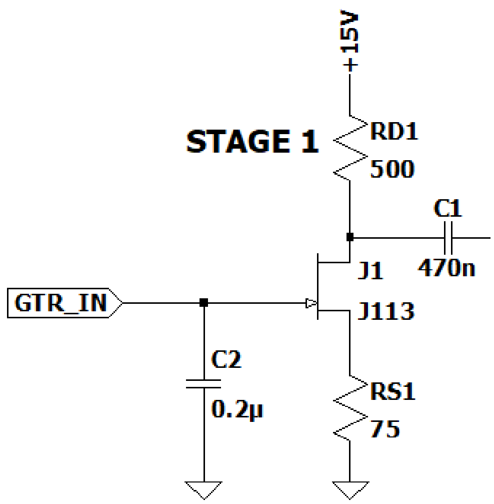


Figure 2.2

The input stage of any guitar preamplifier must account for natural fluctuations in output and impedance that come with different guitars. Ideally, two different inputs would be used; one for both low and high output guitars. The preamp above is based on low-output guitars; an additional high-level input would simply contain a voltage divider to knock the incoming signal down to the low input level. So for the preamp above, high output, active guitars will need to lower their own output or the drive knob on the device to achieve a quality sound.

Managing the input and output impedance of any device is vitally important. To ensure the guitar signal is delivered to the preamp and prevent loading, a sufficiently high impedance must be presented to the guitar (a good rule of thumb is the 10:1 rule). For this reason, the input guitar signal couples directly to the gate of the first JFET. Since the impedance of such JFET's can be on the order of 10^10 Ohms, virtually the entire guitar signal is delivered to the preamplifier input. [6]

Notice Capacitor C2 on the input stage; this limits the high frequency response of the amplifier to better emulate a triode. Triodes have an intrinsic Miller capacitance that can limit the amplifier bandwidth down to 20 kHz. A JFET does not possess nearly as high of an intrinsic capacitance; C2 emulates a triode by shunting high frequencies to ground. [4]

The gain of every JFET stage in this preamplifier, when biased with the proper source and drain resistances, can be calculated with the following equation:

Av = 0.5 * Rd / Rs

Equation 2.5: Source and drain resistor calculations based on Dimitri Danyuk's "Triode Emulator" [4]

For the ideal Rd = 500 and Rs = 75:

Av = 0.5 * 500 Ohms/ 75 Ohms = 3.33V/V

20log(3.33) = 9.5 dB

Stage 1 and 2 are coupled together by 470 nF capacitor C1. This capacitor's sole purpose is to remove the DC bias on the audio from the +15V power supply of the drain of J1.

A coupling capacitor connects each stage and the output of the preamplifier to remove this DC offset. While necessary, a side effect of these capacitors is an overall high pass filter on the device. Adjusting these capacitors affects the bass response of the circuit.

Below is the high-pass frequency response formed by this coupling capacitor. The cutoff frequency, or -3dB point, is below the audio spectrum (20Hz) and will not perceptibly affect the audio signal.

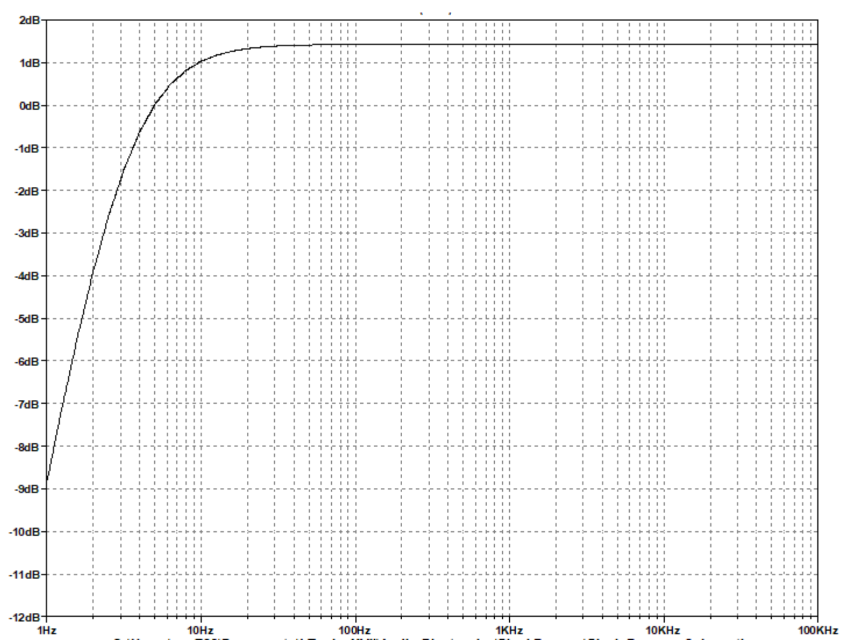


Figure 2.3: Low pass filter formed by Stage 1 coupling capacitor

2.4 Stage 2

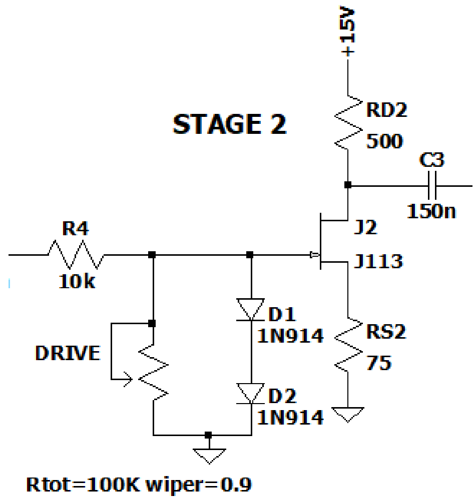


Figure 2.4

The voltage divider formed by resistor R4 and the Drive potentiometer form the heart of the distortion capabilities of this preamp. Notice the diodes D1 and D2: instead of clipping the JFET at its intrinsic Vp, which varies between devices, two IN914 diodes with forward voltages of 0.6V are placed in series to ground. Once forward biased, each diode becomes a short circuit and holds a constant 0.6V across it. Together, these diodes together clip any audio signal that swings above 1.2V, the sum of their forward voltages.

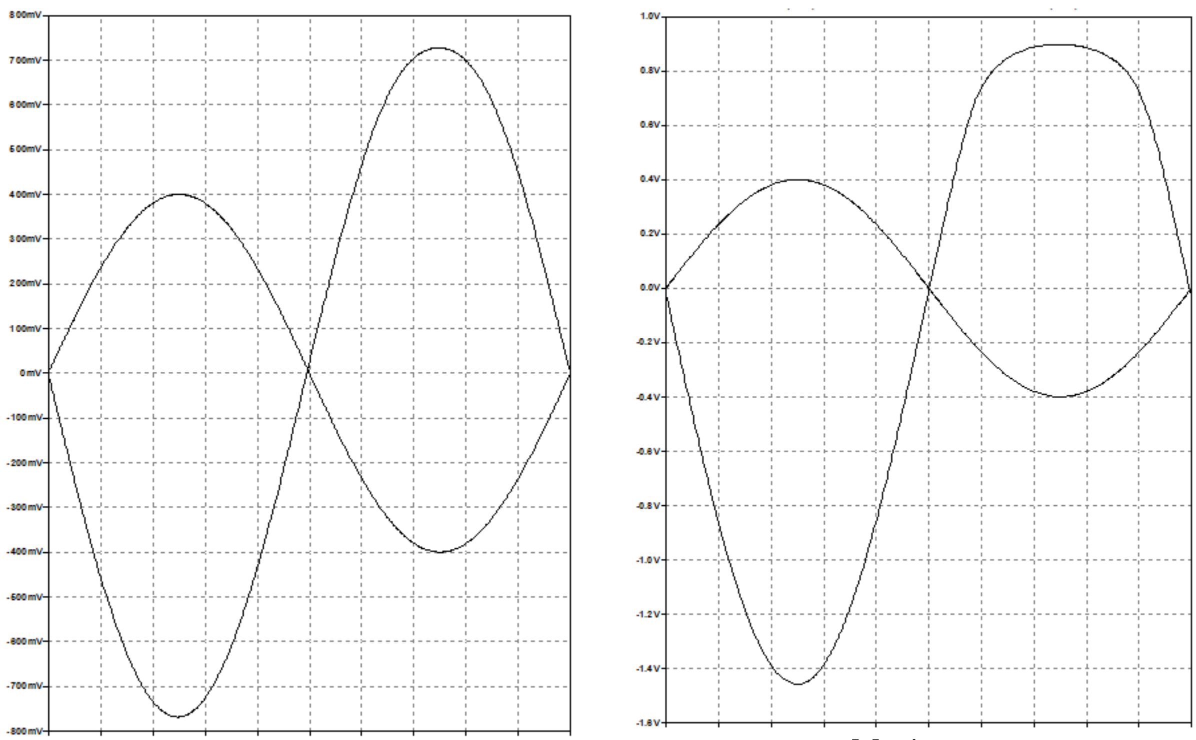
By changing the type and number of diodes in the input path, unique modifications to the distortion sound can be made. For example, adding silicon diodes with Vp at 0.7 each achieves a higher clipping voltage (1.4V) and cleaner sound. Adding a germanium diode on top of the silicon brings the clipping voltage up to 1.6V, directly under the JFET Vp.

When the Drive potentiometer is at a minimum, the audio signal is shorted to ground. At a maximum, virtually the entirety of the audio signal passes to the gate of J2. Remember the general equation for voltage divider:

$$\frac{R2}{R1 + R2}$$

Equation 2.1. In this case, the Drive potentiometer and resistor R3 take the place of R2 and R1 respectively. [5]

So by adjusting the potentiometer, the user controls how much of the audio signal is driven into the silicon diodes. Turning up this drive control increases the clipping.



Minimum

Maximum

Figure 2.5: Transient analysis of diode clipping.

2.5 Stage 3

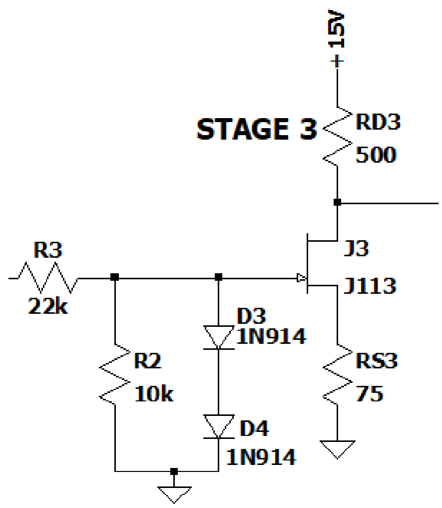


Figure 2.6

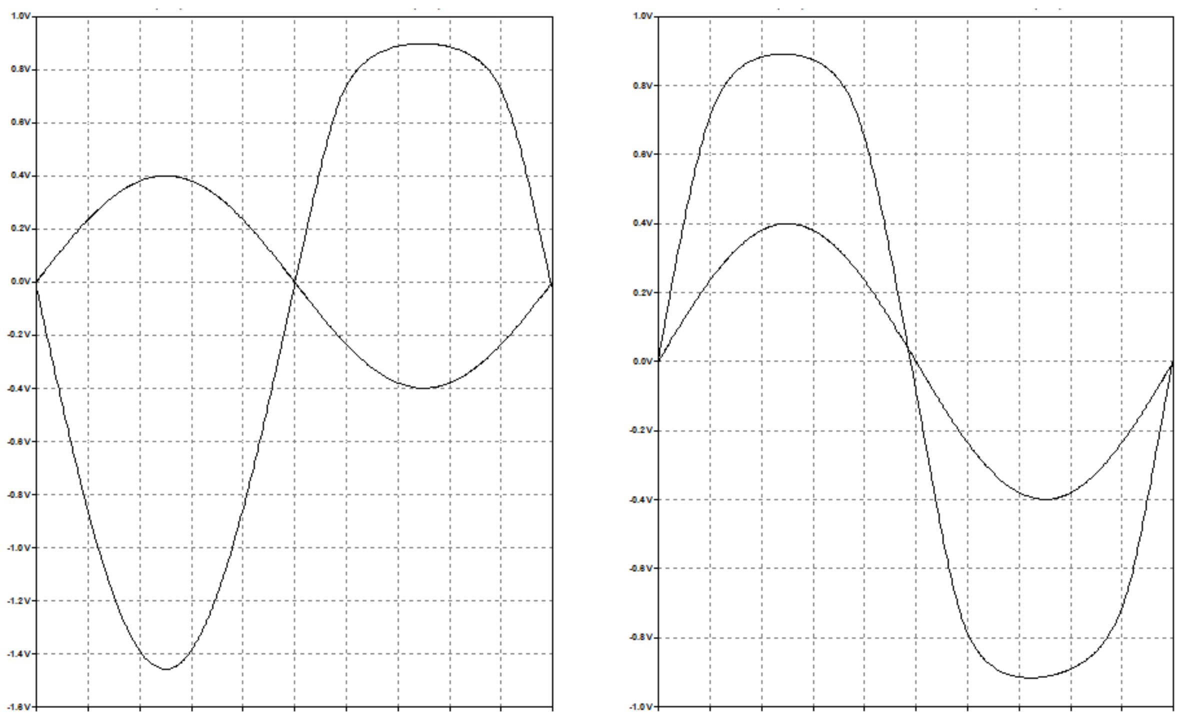
Stage 3 is almost an exact replica of Stage 2. It performs the same operations: voltage division and diode clipping.

The important difference is the lack of a unique Drive control to this stage; the design does not require one. Notice the voltage divider formed by resistors R2 and R3. These resistors serve to step down the output of Stage 2 by exactly the same amount as it was amplified (9.5dB).

$$Av = \frac{10k\Omega}{22k\Omega + 10k\Omega} = 0.313$$

Equation 2.2: Voltage Division before Stage 3 input. [5]

Below in Figure 2.7 is the comparison between the clipped inputs of Stage 2 and 3 vs the input sinusoid.



Stage 2 Stage 3

Figure 2.7:

As visible in Figure 2.7, the signal at the clipping diodes of Stage 3 is virtually the same amplitude as at the Stage 2 diodes. However, the negative signal portion at Stage 2 is still clean. But the JFET amplifier configuration, with the drain as the output, serves to invert the signal. So the diodes of Stage 3 are presented with the unclipped half of the Stage 2 output. The result: symmetrical distortion.

Notice that the drive control of Stage 2 actually controls the distortion of Stage 3, due to the amplitude mirroring that occurs between the Stages. As the drive control increases and more signal is clipped by Stage 2, the clean, negative portion of the signal is greater entering Stage 3, and it is then clipped symmetrically.

Note that resistor R2 is a trim potentiometer. The user can and should adjust the trim potentiometer in order to get the desired distortion.

2.6 Stage 4

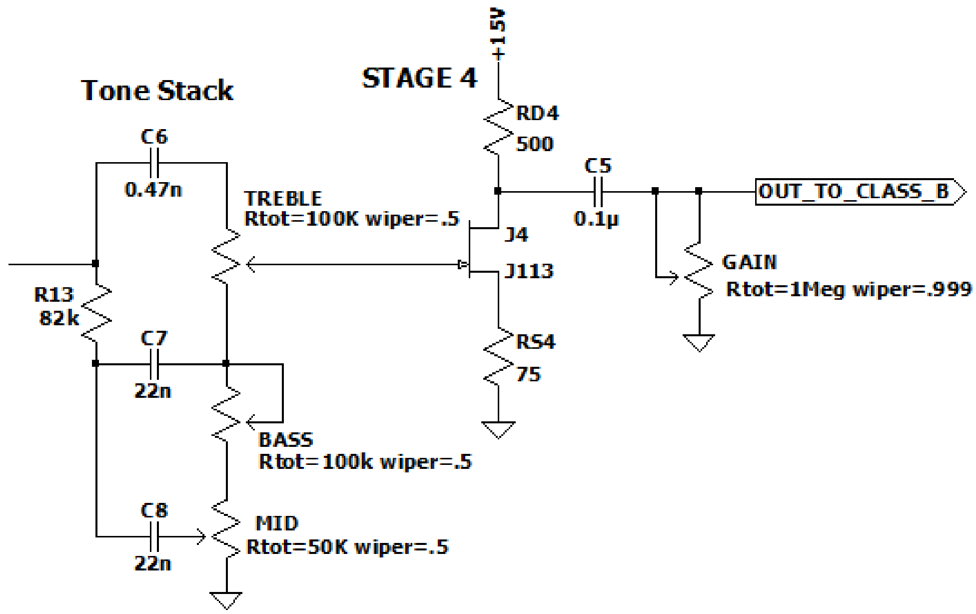
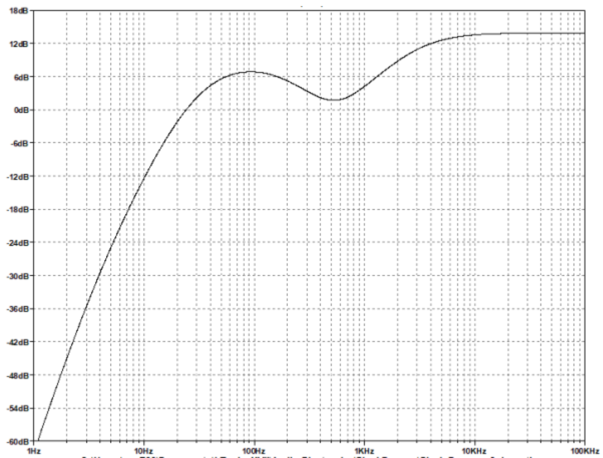


Figure 2.8

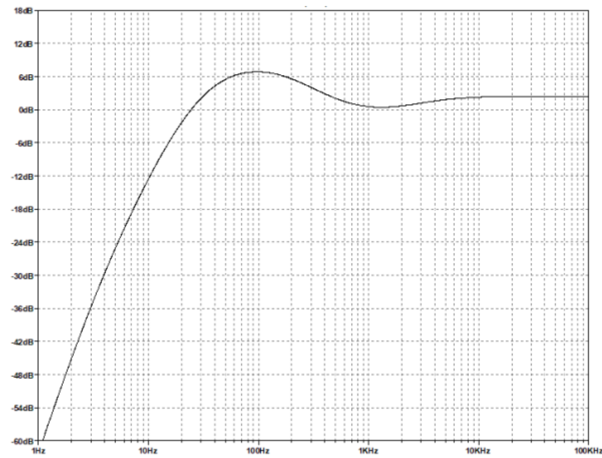
The tone controls for the preamplifier exists completely in this final stage; the output filter (tone stack) from the Mesa Boogie DC-3 has been directly copied to give the best emulation of its sound. But the placement of said tone stack is different than in the original.

While the original Mesa Boogie tone stack is placed on the entire output of the device, the JFET tone stack is placed between Stages 3 and 4. The relatively low supply voltage and gain of JFET's (compared to triode amplifiers) results in the filter attenuating the audio signal far below line level. In order to leave the preamp at the appropriate line-level, the audio signal required additional amplification.

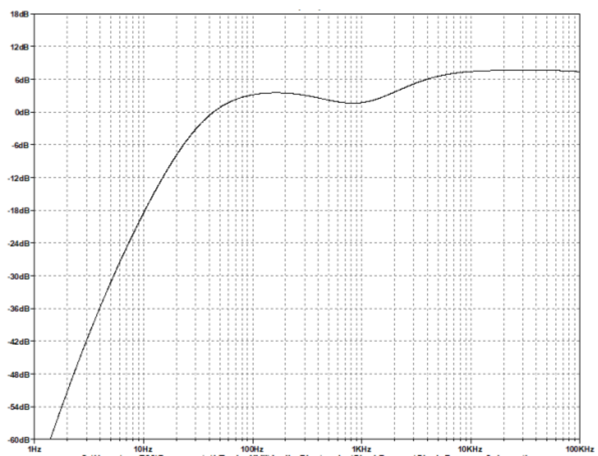
Below are multiple frequency response plots for various adjustments of the Bass, Mid, and Treble controls present in the tone stack.



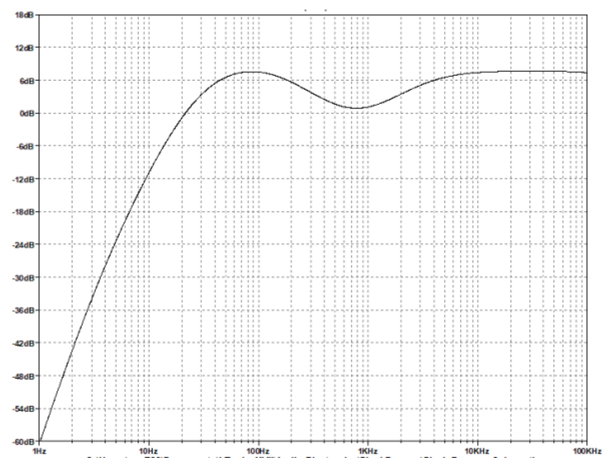
Maximum Treble



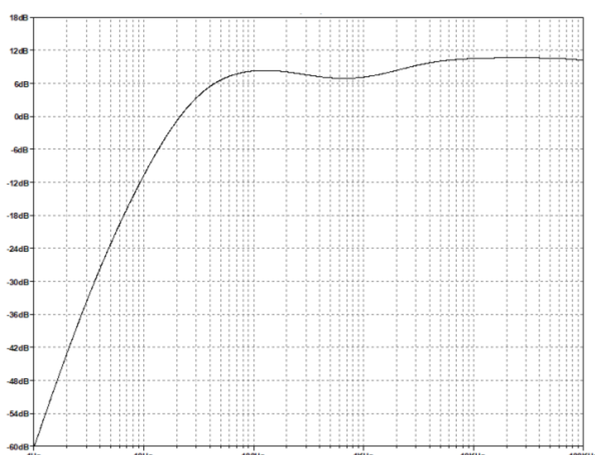
Minimum Treble



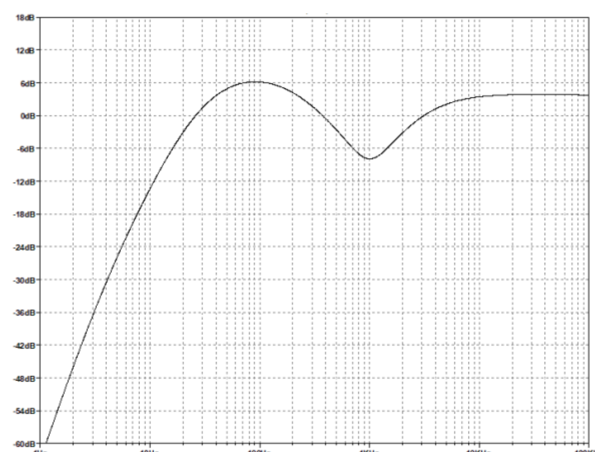
Maximum Mid



Minimum Mid



Maximum Bass

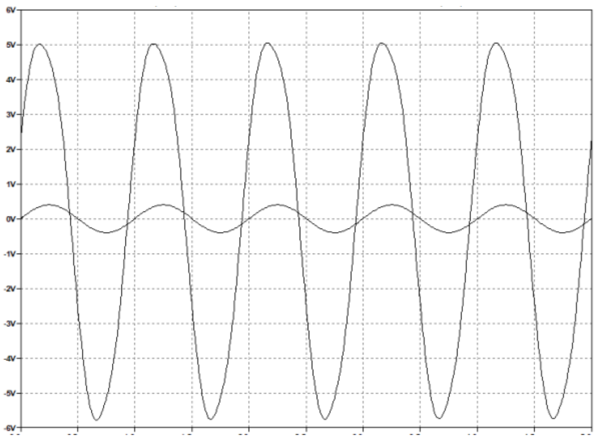


Minimum Bass

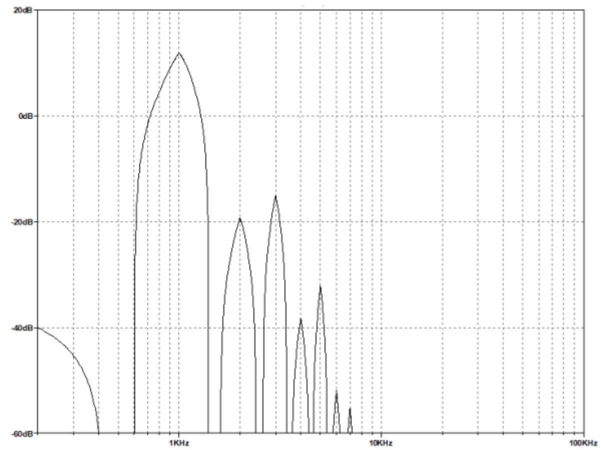
Figure 2.9

2.7. Output

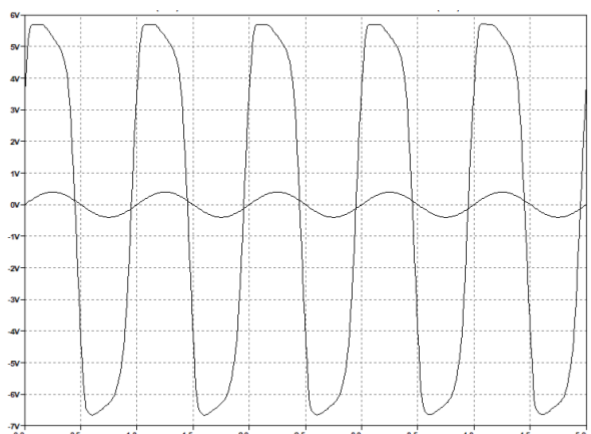
Below are the two main sounds of the preamp; clean and distorted:



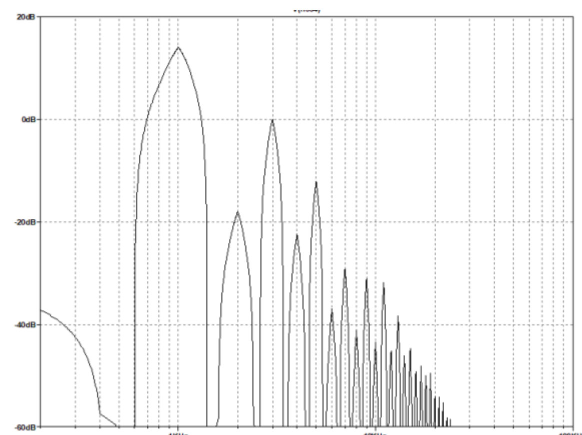
Clean Transient Output, Drive at a Minimum



Clean FFT



Overall Driven Output; Drive at a Maximum



Driven FFT

Figure 2.10

3. Class B Emulator

3.1 Class B Tube Amplifier Characteristics

The design of class B tube amplifiers have several desirable characteristics, including tube compression and increased crossover distortion. Tube compression occurs when a tube-based power amplifier is driven into clipping, and rather than generating harsh hard-clipped harmonic distortion, the amplitude of the signal is dropped to help create what is considered a more “musical” effect in clipping. For class B topologies, the input signal is split into opposite phases, which are then amplified in two output tubes which are summed back into a full waveform. Should the power amplifier in a push-pull configuration be given a high enough signal level to drive the output tube grid into a positive voltage difference with the cathode, then a forward biased diode has been formed.

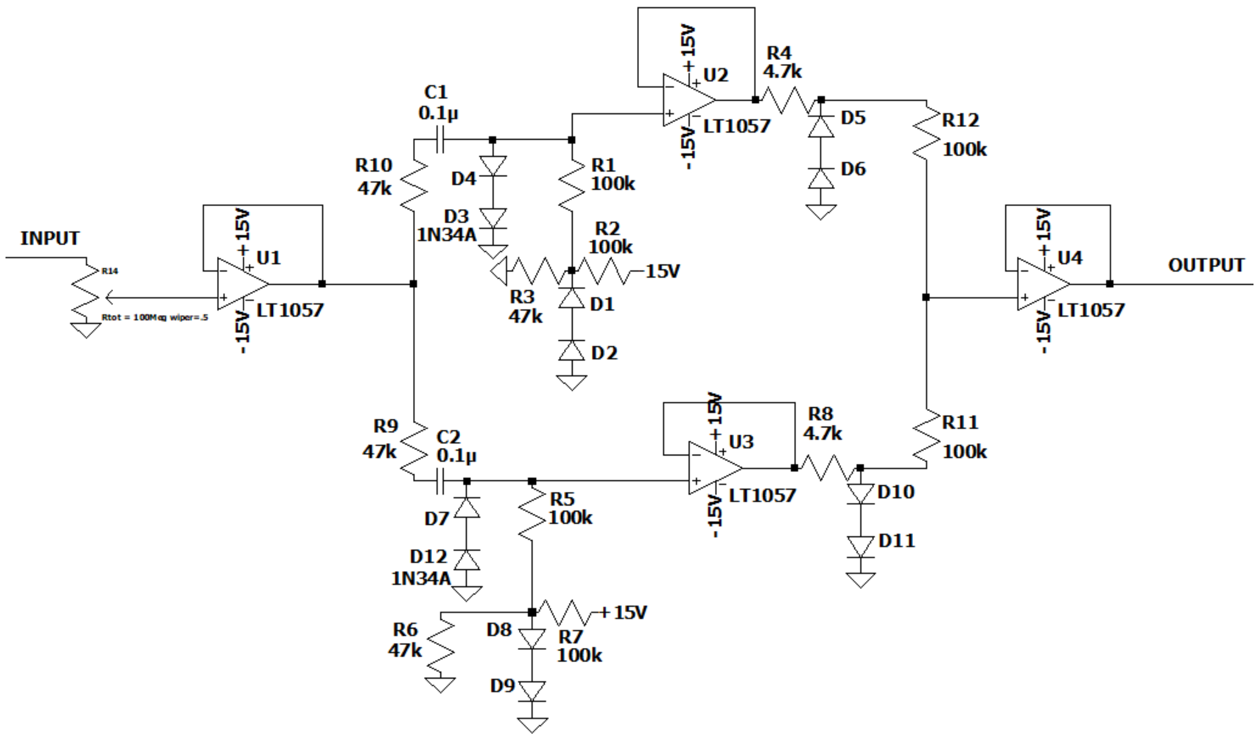
This forward biased diode formed at the output tube grid has the effect of clipping the positive peak of the input signal, but also in affecting the DC bias voltage at the grid of the output tube. This negative DC bias voltage will increase in proportion to the excess input signal being clipped. When the two opposite phase output tubes have become over-biased beyond class B, crossover distortion is generated. Crossover distortion means that there is a delay or flatline as the signal crosses from positive to negative or vice versa.

The conditions to create crossover distortion not only create the distortion, but also a reduction in amplitude since the excess signal is being lost into the crossover distortion region. This allows tube power amplifier designs to create a compressed and less harsh output compared to a standard solid state amplifier when overdriven.

3.2 Solid State Emulation

A notable design for a solid state circuit that would emulate the effects of class B crossover distortion was presented in Jack Sondermeyer’s 1996 US patent. The patent outlined a design using two paths signal flow through op amps, each pathway only affecting one half of the signal. The patent describes a technique for biasing each side of the waveform such that low level signals transfer cleanly, but large signals will modify the DC bias and create uneven clipping. We will detail how these adjusted biases create crossover distortion in the following section.

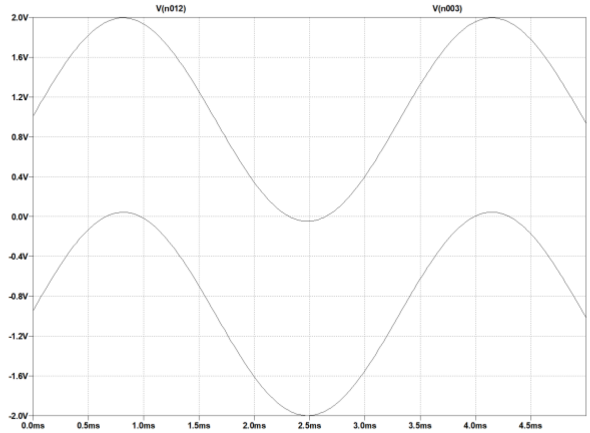
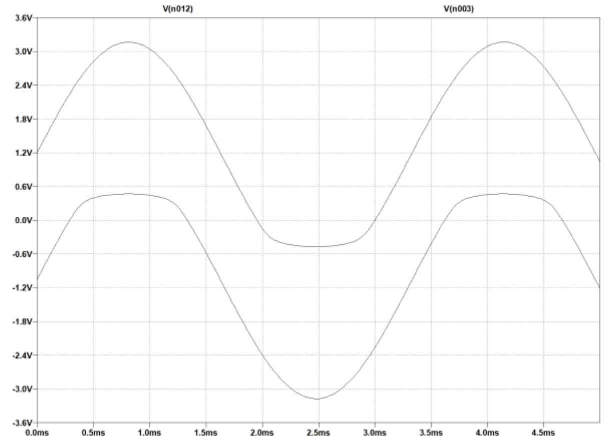
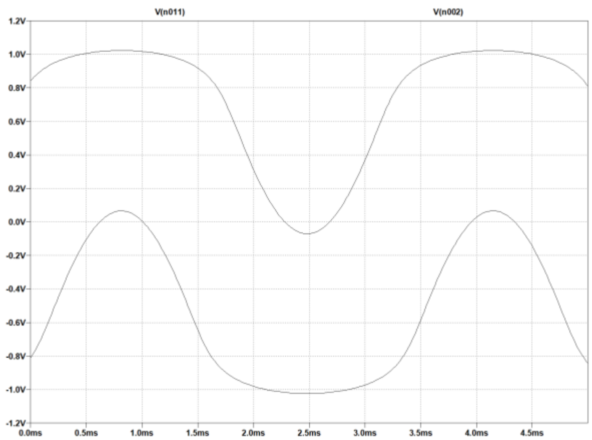
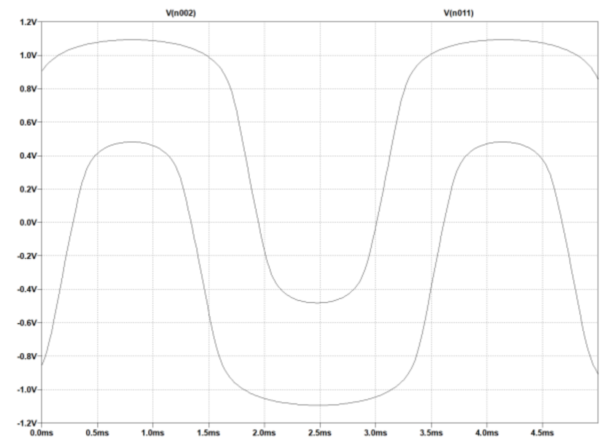
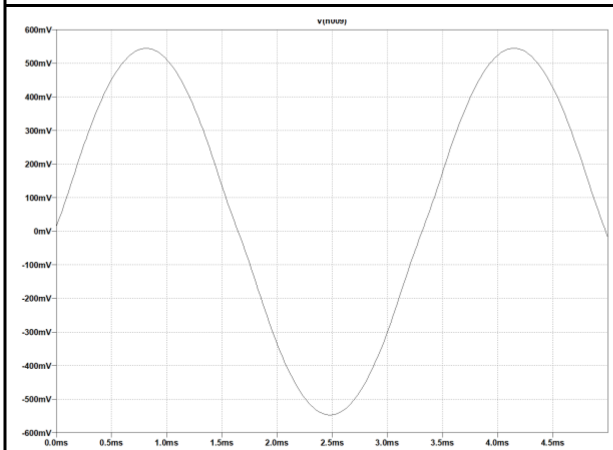
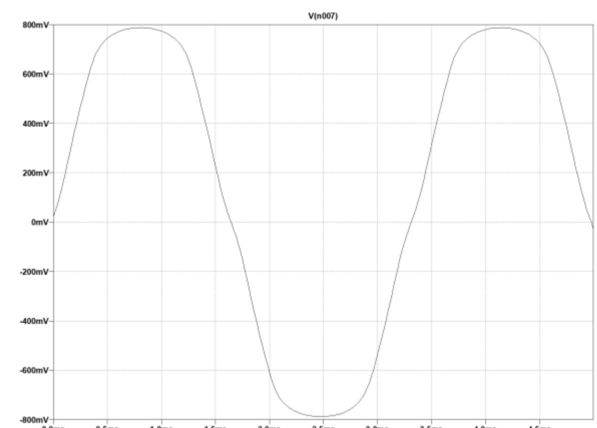
Our class B emulation circuit is shown in figure XX. This design is similar to that in Sondermeyer’s patent and a portion of the Peavey Bandit 112’s power amplifier section in its topology, though some adjustments have been made to better fit our expected input/output levels. All diodes are 1N4148 silicon diodes with a bias voltage of 0.6V, except germanium diodes D3 and D12. The input and output of this circuit are both buffered to prevent loading, and an audio taper potentiometer allows for control of the input level arriving from the preamp stage.



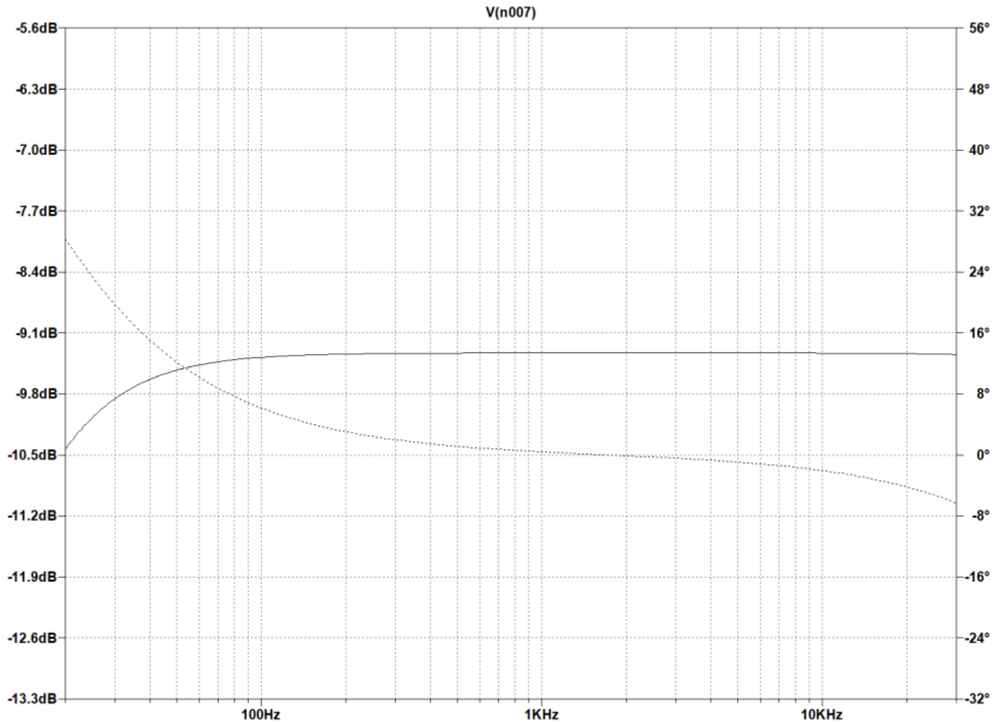
The first component of the circuit to identify are the bias points. At the node joining R1 and D1 on the upper path, a voltage divider applies a negative DC bias. This bias has a fixed value of -1.2V due to the coupling of D1 and D2 to ground. This bias maintains through amplifier U2 and appears across D5 and D6, biasing them idle just about to conduct. Likewise, on the lower path, the node joining R5 and D8 has a fixed DC bias of +1.2V, which appears after U3 and biases D10 and D11 idle, almost into conduction.

For some low level input signal, with an amplitude of 1.5V, the signal will take on the -1.2V bias, and come through the amplifier ranging from -2.7V to 0.3V. Note that where diodes D3 and D4 couple to ground, a minimum of 0.8V would be required to conduct to ground. Our positive peak of 0.3V will therefore not cause any clipping at those diodes. The signal now reaches D5 and D6, where any voltage above -1.2V will pass, but the entire negative half of the waveform below -1.2V will be clipped. The lower path operates the same, but with opposite diode polarities. each half of the waveform therefore passes cleanly and is summed following R11 and R12 into a bipolar waveform.

For a larger input signal, with an amplitude of 3V, the signal will again be biased -1.2V down, swinging from -4.2V to 1.8V, but in this case the positive peaks will allow for D3 and D4 to conduct to ground. Thus, our input signal will begin to clip its positive peaks at 0.8V, and the average bias of the signal following the amplifier will not be -1.2V, but instead nearly -1.8V. D5 and D6 will still clip all voltages below -1.2V, but that is no longer the zero crossing point of our waveform due to the changed bias value. As a result, a positive portion (for which the diodes are overbiased) is clipped in addition to the negative portion of the waveform. When summed with the equivalent process of the lower path, some amount of crossover distortion can be noticed.

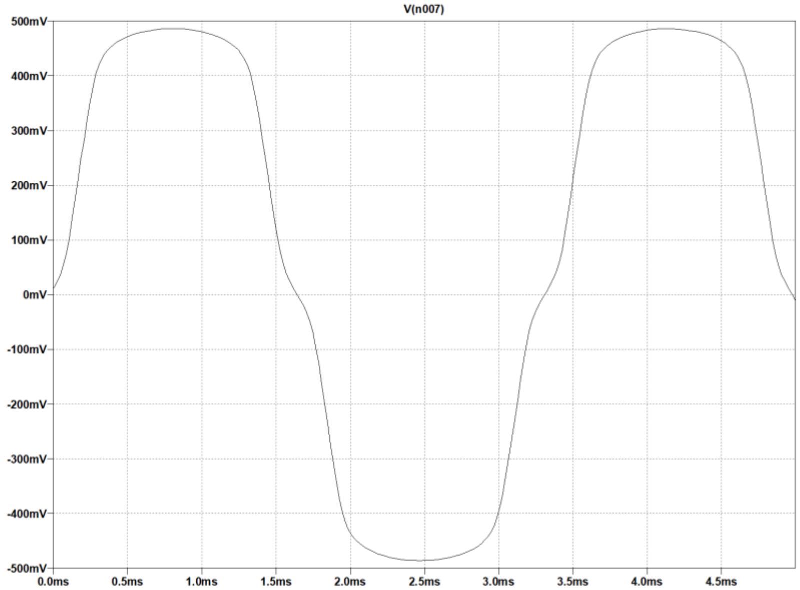
1.5V Input	3V Input
 Biased signals with no clipping before op-amp	 Biased and clipped signals prior to op-amp
 Signals after the idle biased diodes	 Signals following the overbiased diode clipping
 Output signal with minimal distortion	 Output signal is clipped and shows crossover distortion

It is worth nothing that the class B emulator is not designed to have any significant frequency domain effects, apart from the harmonic distortion and the high pass effect of coupling capacitors. A broadband -9 dB attenuation is observed as well.

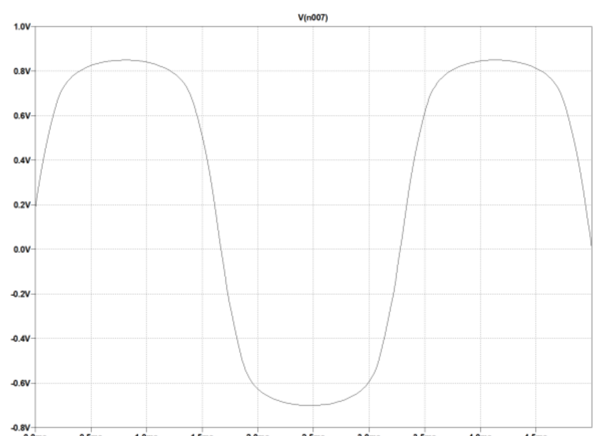


3.3 Modifications

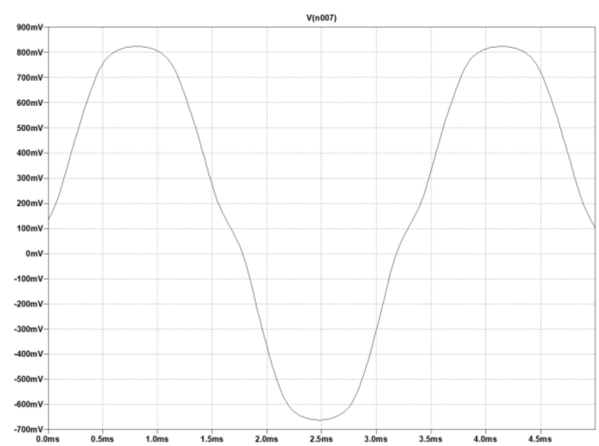
A number of possible modifications can be made to this design. The first, which reflects how Sondermeyer designed how patent's example circuitry, is to use single diodes to reduce our biases to 0.6V. This results in clipping at lower voltages and more pronounced crossover distortion. We also observe that the 3V input has also dropped to a 0.5V output, so the number of diodes to be used for any particular design should take the desired output level into account.



Another modification is to alter resistor values at the bias voltage point. This has the ability to pronounce either the peak clipping or crossover distortion. Their peak output levels are very similar, though crossover distortion will sound quieter will less energy under the curve. This demonstrates how crossover distortion relates to emulating tube compression as described earlier.



R2, R7 = 200k; R3, R6 = 2k
Peak clipping is pronounced



R2, R7 = 2k; R3, R6 = 200k
Crossover distortion is pronounced

The Tube-Dynamics Control that Peavey implemented in their Bandit 112 would be another worthy modification to our design. By using a dual-ganged potentiometer to control input and output gain simultaneously (and with inverse controls), the amount of distortion introduced can be controlled independently of simple in and output gain.

4. Loudspeaker Simulator

4.1. H&K Triamp

The purpose of the loudspeaker simulator is to filter the signal in a way that mimics the frequency response of a guitar cabinet and speaker. Common characteristics of speaker simulators include a steep high-frequency roll-off, resonances, and a mid-frequency notch. The H&K Triamp speaker simulator [7] features a fourth-order low-pass filter with resonance. Its schematic is shown in Figure 4.1.

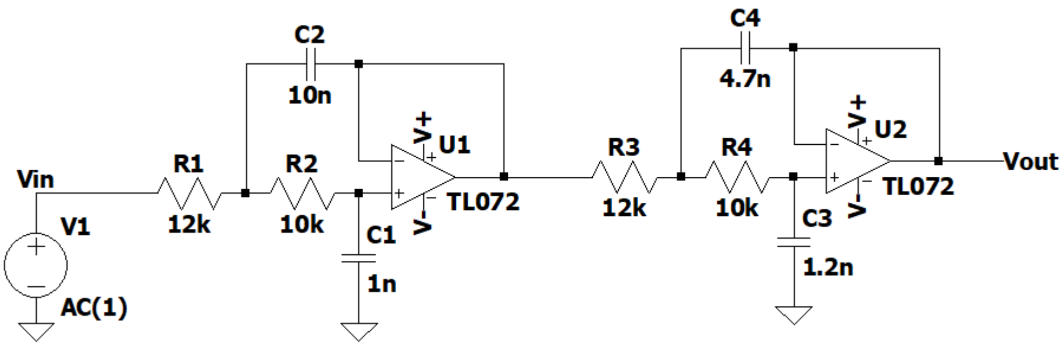


Figure 4.1: H&K Triamp speaker simulator schematic

The circuit consists of two active low-pass filters. Performing Laplace-domain nodal analysis and solving for the output of U1 in terms of the input leads to the following transfer function for one of the two filters:

$$H(s) = \frac{1}{1 + (R_1C_1 + R_2C_1)s + (R_1R_2C_1C_2)s^2}$$

The form of this transfer function shows that it is a second-order low-pass filter, and so two stages of this makes the circuit a fourth-order filter. Given that a second-order low-pass filter takes the form:

$$H(s) = \frac{1}{1 + (\frac{1}{Q})(\frac{s}{2\pi f_c}) + (\frac{s}{2\pi f_c})^2}$$

The cutoff frequency and Q-factor for each stage can be calculated as:

$$f_c = \frac{1}{2\pi\sqrt{R_1 R_2 C_1 C_2}}, Q = \frac{1}{2\pi\ (R_1 C_1 + R_2 C_1)\ f_c}$$

For the first stage, the values used give a cutoff frequency of 4.59 kHz and a Q-factor of 1.57. A Q-factor greater than 0.707 is indicative of resonance. For the second stage, the same formulas are used to calculate a cutoff frequency of 6.12 kHz and a Q-factor of 0.985. The cascading of these two filters in series should therefore produce a resonance around 4-5 kHz and a steep roll-off afterwards. The magnitude response from a SPICE simulation of the circuit is shown in Figure 4.2 to verify the analysis.

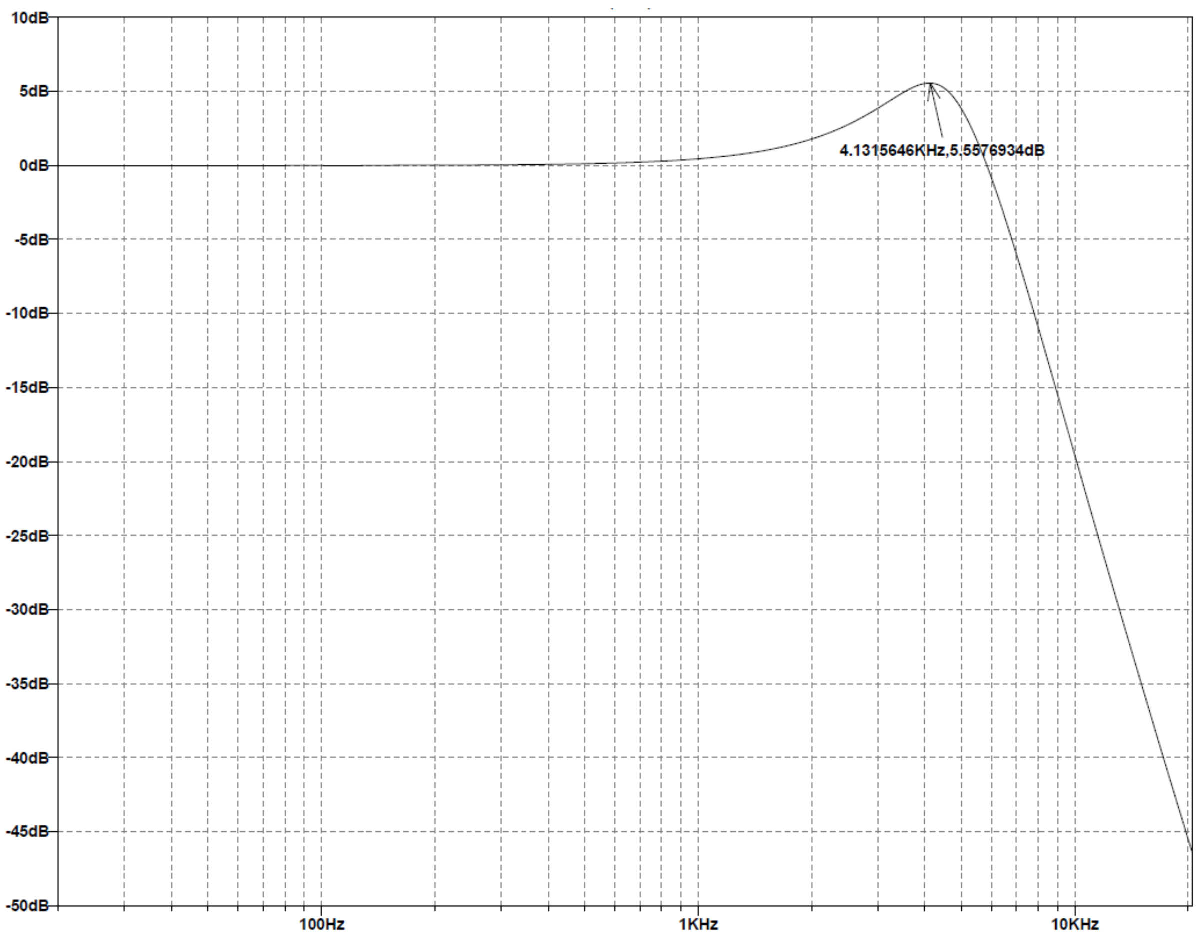


Figure 4.2: Magnitude response of H&K Triamp speaker simulator

The simulation shows a resonance at around 4.1 kHz, with a peak gain of 5.5 dB. The filter response is unity gain at low frequencies, which can be seen both by the transfer function (by substituting in $s = j\omega = j(0) = 0$) and by the fact the when the capacitors become open circuits at DC, the resulting circuit is simply two unity-gain buffers.

The output of the loudspeaker simulator will be the final output of the system, so it will be amplified to a suitable line level. The input to this stage comes from the class B emulator circuit, whose output peaks at around 0.8V. A typical professional line level is +4 dBu, which is roughly 1.2 V_{RMS} or 1.7 V_{peak} for a sine wave. To reach this approximate level, a non-inverting op amp with a gain of 2 is added to the output, followed by a potentiometer so the user can control the master volume of the circuit. With a non-inverting op amp having a gain equation of

$$A_{non-inv} = 1 + \frac{R_f}{R_i},$$

a gain of 2 can be accomplished by setting the input and feedback resistances equal to each other. The circuit with this amplification is shown in Figure 4.3.

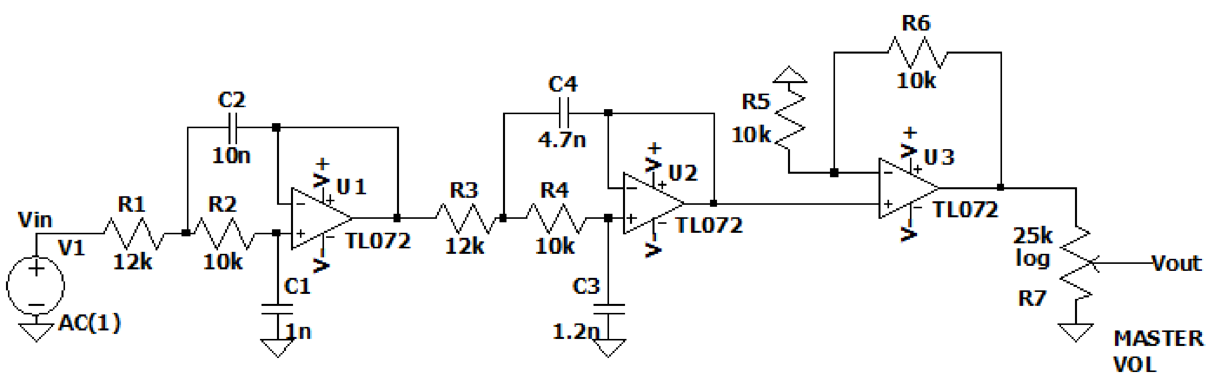


Figure 4.3: Speaker simulator with output amplifier

This is the circuit that was implemented for the build, but a potential improvement could be made by adding a mid-frequency notch to the circuit. The following section describes how this notch could be designed and implemented.

4.2. Addition of Notch Filter

Typical guitar loudspeakers have this notch around 1.5 kHz. This could be done by implementing an active notch filter after the two low-pass filters. The modification is shown in Figure 4.4.

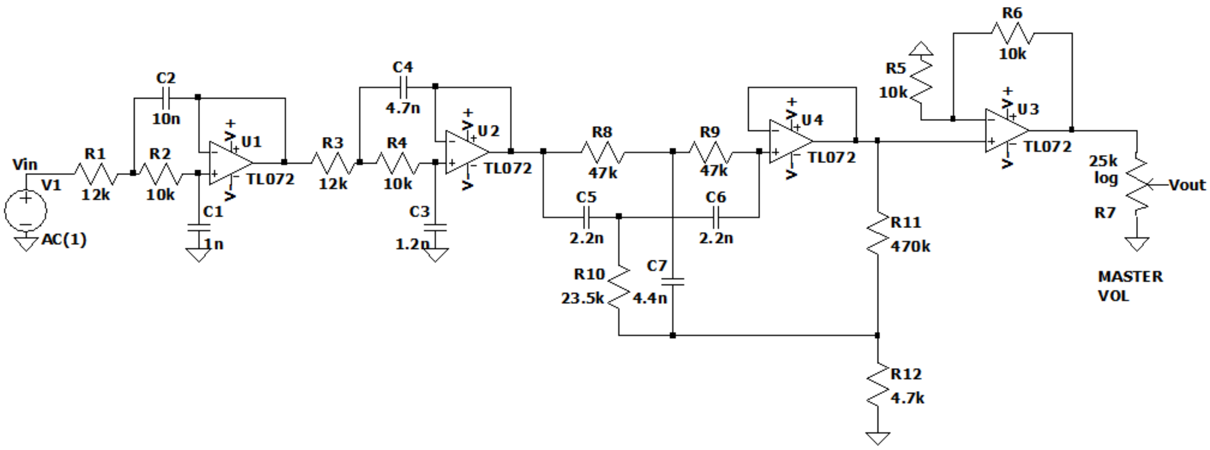


Figure 4.4: Speaker simulator schematic with notch filter

This notch filter circuit, presented and analyzed in [8], requires that C7 be twice the value of C5 and C6, and that R10 be half the value of R8 and R9. With these conditions, the notch frequency can be calculated as

$$f_n = \frac{1}{4\pi C_5 R_{10}}.$$

The values chosen give a notch frequency of 1.54 kHz, which is perfect for a loudspeaker simulator. The voltage divider established by R11 and R12 determines the depth and Q-factor for the notch. These values were determined experimentally in a SPICE simulation until a reasonable frequency notch was achieved. The overall frequency response of the circuit is shown in Figure 4.5.

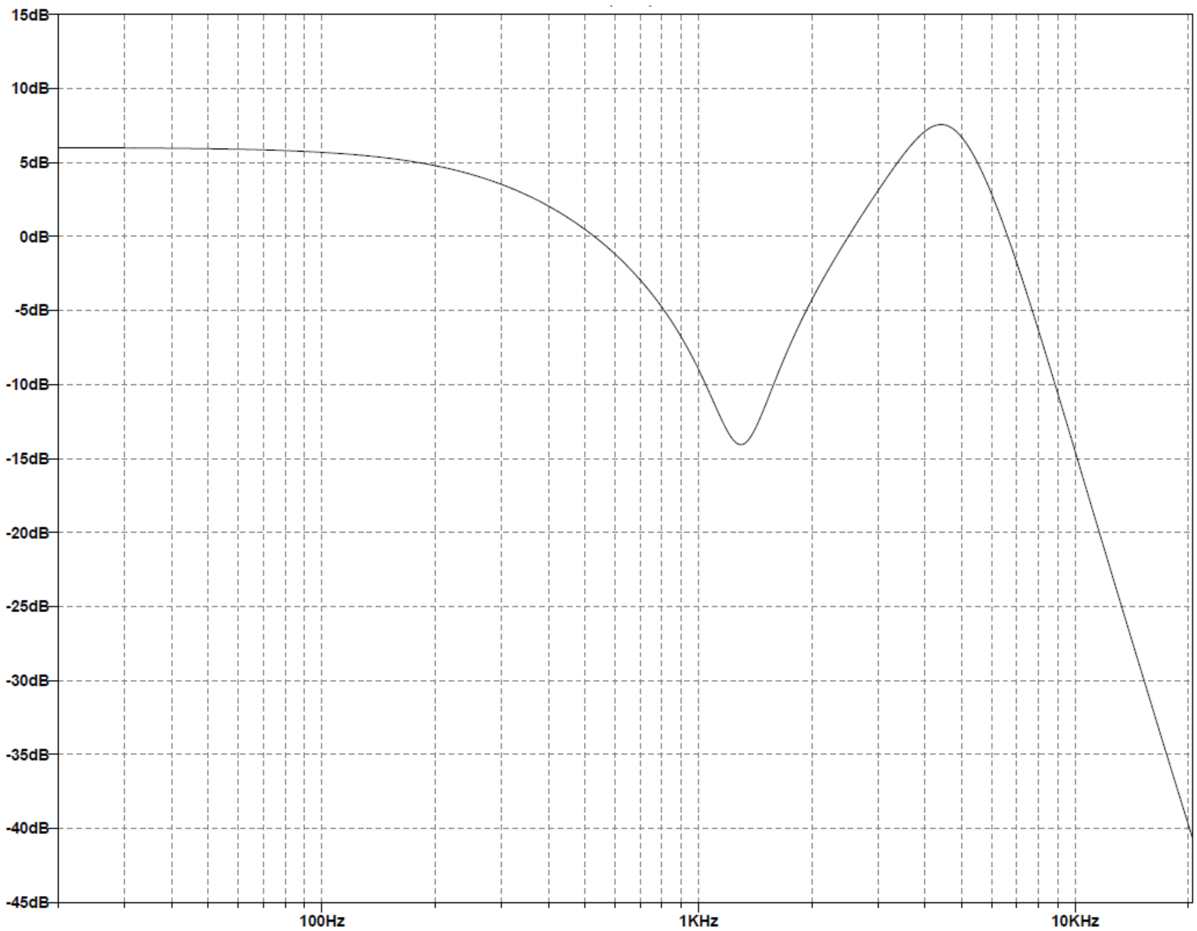


Figure 4.5: Magnitude response of speaker simulator with mid-frequency notch

The 6-dB gain at low frequencies is due to the final op-amp's gain factor of two. The added notch at 1.5 kHz is clear.

5. Conclusion

This report details the design and implementation of a circuit emulating a tube amplifier into a cabinet. The final design consists of a single unbalanced input, requiring 16 VAC (~1 A), with three outputs—preamp out, class B out, and cabinet sim out. The preamp and class B output ~5 VAC, while the speaker simulator knocks the signal down to line level (+4 dBu).

The device models the Mesa Boogie Studio Caliber nicely, replicating both the shape of the waveform and the overall tone of the amplifier. Several improvements could be made however, the most pressing being the tone stack, as the mid control cuts out a majority of the signal at the low position, and the high frequency control adds only subtle presence to the response. Additionally, it might be preferable to knock down the tapped outputs from the preamp and class B emulations so as to be able to run them into external amplifiers. Experimenting with a voltage divider quickly raised impedance issues, suggesting that buffers may be required to knock the tapped output down to line level without losing gain through to the subsequent stages.

The College Budget Tube Amp is the solid state all-in-one, plug-and-play, modern-day replacement for the vintage, sought-after sound of the classic tube amplifier.

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