Abstract

The “PA System in a Box” is a fully functional personal-audio system with the quality and features found on professional audio equipment but without the setup and transportation requirements of a full multi-part PA system. A modular approach to design allows the audio system to be broken up into a number of functional modules that can receive, process, and output audio signals independently of one another. The modules include input channels to receive, process, and amplify low level signals, a mixer to combine and distribute audio signals, a compressor to reduce the dynamic range of the sound, a stereo class-G amplifier to drive speakers, and a power supply to provide multiple power rails for the components. These components are all packaged into a portable acrylic enclosure to form a complete and functional PA system in a box.
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Introduction

Motivation

The motivation for this project was to achieve a better understanding of the nuances that are part of designing a well-performing functional audio system. The designers share a common background in the enjoyment and creation of music.

Goals

The project had three tiers of goals, including the minimum requirements that the project was expected to meet, the primary objectives that would result in a solid product, and the stretch goals which would push the project over the top.

The minimum requirements of “PA System in a Box” were as follows;

- Amplification of low-level (microphone) signals to line level
- A mixer able to sum and distribute signals to the system’s output
- Mono amplification of the summed mixer output
- A linear power supply providing ±15 V rails

The main goals for the project were to have;

- Comprehensive input channels
  - 3 input channels
  - Low-pass and high-pass filtering
  - Single-band parametric EQ
- System bandwidth of 20 Hz -> 20 kHz, input to output
- A functioning class G amplifier
- A mixer with switchable send and returns for outboard effects
- A Bucking power supply to convert ±25 V rails to ±15 V rails

Finally, the stretch goals of the project included designing;

- A compressor to limit signals to a consistent dynamic level
- A graphic equalizer to allow boost and cut of multiple frequencies on the mixer’s output
- Class AB Headphone Amplifier
- An enclosure to house the modules

Scope

The scope of the project was to design, build, and package the PA system so that it could pass for a pre-production piece of equipment. Because audio systems are by nature modular, the division of labor was easy and based on interest. This resulted in Tuan designing and building the amplifier and compressor effect, Nathan designing and building the input channels and mixer, and Edwin designing and building the power supply and enclosure.
Design

Background

Audio signal levels in equipment are typically measured in units of dBV and dBu. Units of dBv are typically used with consumer audio equipment, such as CD players or iPods, while dBu are used when working with professional audio equipment such as preamplifiers and effects. Both units have different zero values, but increase and decrease in logarithmic fashion. A value of 0 dBV corresponds to 1.0 Vrms, while a value of 0 dBu corresponds to approximately 0.775 Vrms. Because sound can be a very dynamic signal with many peaks, loud sections, and quiet sections, signal specifications are typically stated as being “nominal”, which is what the device expects on average. Consumer audio equipment operates at a nominal level of -10 dBV, or 0.316 Vrms, while professional audio equipment expects and operates at a nominal level of +4 dBu, or 1.23 Vrms.

Also important to this project is a basic understanding of human hearing and perception of sound. Human hearing is generally accepted to span across the frequency range of 20 Hz through 20 kHz. The perception of musical tones to signal frequency relates an increase of one musical octave to a doubling of frequency. For example, the note A4 oscillates at 440 Hz, while A5 oscillates at 880 Hz and so on. Finally, there is the whole matter of psychoacoustics that complicate the relation between perceived loudness and power. To summarize, it is commonly accepted that doubling the power into a speaker results in a 3 dB increase in sound pressure level, and that a 10 dB is typically perceived as a doubling of loudness. For this reason, volume controls should all operate in a logarithmic fashion.
System Overview

The overall architecture and layout of our system followed industry standards for live sound reinforcement. The system features input channels that allow the connection, amplification, and spectral manipulation of signals from audio devices such as microphones, iPods, and other musical devices. The output of the input modules enters the mixer that provides level manipulation and stereo panning controls for each input. The mixer also allows for the connection of external effects and devices through the use of sends and returns. A stereo compressor is connected to a send and return on the mixer, and the main output of the mixer is then fed to a stereo Class-G amp which provides an average of 20 Watts to each speaker. Power to all modules is provided by a combination linear / switchmode power supply and the entire project is packaged into a custom acrylic enclosure.

Figure I: A simple block diagram of PA System in a Box

Input Channels (NG)

The input channel is the system’s connection to sound generating devices. It allows the operator to connect microphone or line-level signal sources with either balanced or unbalanced connections. Each input channel consist of a number of simpler modules, including a preamp, lowpass and highpass filter, single-band parametric EQ, and a visual LED clip indicator. The input channel has a bandwidth of 20 Hz through 20 kHz to allow signals in the full audible range while reducing higher frequency noises. The module requires ±15 V rails to supply the op-amps and an optional 48V low-current rail to supply phantom power for the condenser microphones, thereby increasing the capabilities of the system.
Differential Input and Gain

The preamp’s adjustable gain allows a variety of signal sources to be connected to the system, such as a microphone or line-level device with balanced or unbalanced connections. Balanced connections allow long cable runs without noise pickup, since any noise is common mode and is rejected by the differential amplifier. The switchable -15 dB pad, in conjunction with the variable gain, allows the operator to reduce or increase the signal level to professional line level (+4 dBu). The input expects an input signal in the range of 10 mVrms through 7 Vrms and outputs a single-ended professional line level signal at +4 dBu nominal.

Highpass Filter

In live situations, connected microphones may pick up substantial low frequency vibrations caused by microphone movement or passed through the floor from other sources. Excess low-frequency content can cause the speakers to work harder, reducing their sound quality in higher frequency ranges. The high-pass filter serves to eliminate unwanted low-frequency content from the incoming signal.

Lowpass Filter

The low-pass filter eliminates unwanted high-frequency content and reduces the possibility of feedback. Because the human voice contains frequency content that’s at most 3 kHz, a low-pass can be used to reduce the possibility of feedback as the microphone is moved around stage. In other situations, such as recording a bass amplifier, even less high frequency content is necessary and so a low-pass can eliminate extraneous sound that may be picked up.

Single-band Parametric EQ

In an audio system, the EQ allows the operator of the system to apply gain or reduction to a selected frequency. A single-band parametric EQ provides the user with control over a single selected frequency. This can be useful to de-emphasize unwanted vocal harmonics or to compensate for deficiencies in the connected speakers.

Clip Indication

The clip indicator circuit serves to prevent the user from clipping and to aid the operator in setting levels to the nominal +4dBu. Clipping is an undesirable event that occurs in sound systems when the signal exceeds the maximum output swing provided by the op-amps and distorts in very unexpected and unpleasant ways. The clip indication circuit informs the user when they are approaching the circuit’s limit by lighting an LED and allows them to take appropriate steps to rectify the issue, whether by reducing gain or enabling the 15 dB pad. Peak indicators should check the signal wherever there is the possibility of gain, and ideally measure both the positive and negative peaks of the waveform. Additionally, clipping can occur quickly and for very short amounts of time, so the visual indication must be slowed down so that the operator can actually witness it.
Mixer (NG)

The mixer is where the signals from the multiple input channels are combined. It receives a single-ended mono output from each input channel and allows the user to attenuate signals logarithmically and pan (convert from mono to stereo) signals. It then combines the three channels into a stereo pair of signals, henceforth referred to as the “bus”.

To allow the processing of signals by external effects, the mixer provides two switched and buffered stereo sends and returns. Sends take a copy of the signal on the bus and “send” it to a connected device, while returns receive the externally processed signal that replaces the original signal on the bus for downstream modules.

The mixer also features a logarithmic master volume control that controls the strength of the output signal, which is then offered as a stereo pair of differential +4 dBu line-level outputs.

Input to the mixer should be at +4 dBu nominal, while within signals are expected to average around +4 dBu. The mixer does not have any effect on the bandwidth of the system and has a flat response from 20 Hz to 20 kHz. The mixer requires ±15 volt rails to supply the op-amps.

Power Supply (EA)

The power supply provides power to the various modules of the system using a mix of linear and bucking topologies to generate 4 voltage rails from dual transformers. As designed, it provides ±15 and ±30 volt rails with a maximum current of 2 amps each. At higher currents, the ±30 volt rails may drop to ±25 volts.

Enclosure (EA)

The enclosure provides ventilation, easy transport, and protection for the modules of the system. Black laser-cut acrylic was used to create the walls of the enclosure. Laser-etching allowed for the addition of designs and control markings on the enclosure panels.
Differential Input and Gain

The input channel features single-ended and differential RFI filtering provided by two 100 pF capacitors to ground. These capacitors along with the output impedance of the connected device (usually around 200 ohms for a standard low-z microphone) create a passive low-pass filter to block frequencies in the megahertz range that could be picked up by the incoming cable leads.

The differential signal is then fed into a switchable 15 dB pad. This pad is a standard U-pad configuration that acts as a voltage divider. A switch allows the pad to be bypassed completely when not required. When engaged, the output of the u-pad can be calculated with the following equation;

\[ V_{\text{out}} = V_{\text{in}} \frac{R_5 + R_4}{R_3 + R_5 + R_4} + V_{\text{in}} \frac{R_3}{R_3 + R_4 + R_5} \]

In order to achieve a pad of -15 dB, the ratio of output to input must equal \(10^{-15/20} = 0.178\). Since \(V_{\text{in}} = -V_{\text{in}}\) and \(R_3 = R_4 = R\), this equation can be simplified into the following;

\[ V_{\text{out}} = V_{\text{in}} \frac{R_5}{R_5 + 2R} \]
Resistor $R_s$ is chosen to “set” the output impedance of the connected device, and is similar to the actual output impedance of a typical microphone. $R$ is then calculated with the remaining values.

The output of the pad is then fed into a passive high-pass filter to prevent damage to the opamps from phantom power (+48 V) or other large DC offsets. Two 47 uF capacitors and 2.2k resistors were chosen to create a highpass with a corner frequency of 1.5 Hz.

The filter resistors are connected to a floating ground which is then referenced to ground through the high value 110k resistor. The use of a floating ground allows for variation in component values and leads between the differential signals without affecting common-mode rejection through imbalances.

Note that because 48 V phantom power may be switched off and on at will, the capacitors must be rated for the voltage and also require a path to safely discharge without damaging the connected opamps. To handle this discharge, the capacitors have 10 ohm resistors in series to limit the surge current, while back to back 5.6 V zener diodes provide a path to ground for voltage larger than ±6.2 V. These diodes also have the extra benefit of clipping signals that may be too high.

Another high-pass filter follows in order to reduce unwanted RF noise that could cause oscillation problems in the gain stage. A 1k resistor followed by a 1 nF capacitor was used to create a filter with a corner frequency of 160 kHz, or three octaves above our required frequency response of 20 kHz in order to ensure a completely flat frequency response in the pass band.

Differential gain is provided by an instrumentation amplifier. An instrumentation amplifier topology was chosen since they do not require the inputs to be impedance matched, while still allowing for high-gain with high common-mode rejection. Since the resistors were chosen to be equal, the gain equation was simplified to the following:

$$\frac{V_{out}}{V_2 - V_1} = \left(1 + \frac{2R}{R_{gain}}\right)$$

$R_{gain}$ is determined by the series combination of the 100 Ω resistor with the 10 kΩ gain potentiometer (set up as a rheostat). With the above equation, the maximum gain of 201, or 46 dB, occurs when $R_{gain} = 100$ Ω. The minimum gain of 3, or 10 dB, occurs when $R_{gain} = 10.1$ kΩ.

Due to the high gain of this circuit, it was discovered that the parasitic capacitance of breadboards caused the opamps to oscillate. Once the design was constructed on copper-board, oscillation was no longer an issue. However, it should be noted that copper-board construction takes a considerable amount of time and thought when first laying out a many part circuit.

Filtering

Each input channel includes a switchable and frequency variable high-pass and low-pass filter. These filters utilize active Sallen-Key topologies in order to isolate their interaction and achieve consistent response to control inputs.
Filter parameters were chosen so that the filters would possess a butterworth response. Butterworth filters were chosen over Bessel or Chebyshev due to their flat pass-band response and linear roll-off after the -3 dB point, both of which are desirable characteristics in the audio domain. A filter Q of .707 was required to achieve

In order to achieve this Q in the high-pass filter, a gain of 1.56 was necessary. However, since it is preferred that the gain of the signal remains unchanged by the filters, the output is taken from the voltage divider, or inverting terminal of the opamp. Care must be taken to ensure that this node is not loaded by the next stage or else the gain, and thus Q, will change. In this design, the low-pass filter’s non-inverting buffer presents a very high impedance, eliminating the problem. To achieve a Q of 0.707 in the low-pass filter, C14 and C20 are placed in parallel to equal double of C13.
The use of 10k potentiometers as variable resistors allows the frequency of the filters to be adjusted over a range of about a decade. Using the given capacitor and resistor values allows the low-pass to be adjustable from 900 Hz to 8600 Hz and the high-pass to be adjustable from 40 Hz to 220 Hz.

**Single-band Parametric EQ**

A parametric EQ allows the system operator to select a frequency and apply an adjustable amount of boost or reduction centered at that frequency. The designed circuit relies on a Wien bridge network to act as the filter component and a differential amplifier for boost and cut.

![Diagram of the single-band parametric EQ with adjustable frequency and gain](image)

*Figure V: A schematic of the single-band parametric EQ with adjustable frequency and gain*

The Wien bridge network on its own acts as a bandpass filter with a fixed Q of about 0.32. The its tuning frequency, when the resistors and capacitors are equal, is determined by \( f = 1 / (2 \pi R C) \). At its tuning frequency, the Wien bridge network causes an insertion loss of 9.5 dB due to the voltage divider network at the output. This tuning frequency is user adjustable by augmenting the standard resistors with a dual-gang potentiometer used as a dual-gang variable resistor.

The amount of boost or cut is determined through the use of a differential amplifier on the module’s output and the circuit’s 9.5 dB insertion loss. The output of the Wien bridge to the non-inverting terminal of the output opamp, while the unmodified signal is passed to the inverting terminal of the output opamp. The input of the Wien bridge circuit is then connected to
a potentiometer with a center detent that feeds into a buffer in order to isolate the filter from the potentiometer. At the center position, the output signal and input signal of the module will cancel out since they are 180 degrees out of phase, causing the non-inverting input of the differential amplifier to be 0 volts. When the wiper is moved toward the input of the EQ module, the filter will output the original signal to the differential amplifier, reducing, or cutting, it from the output. When the wiper is moved toward the output of the EQ module, the filter will output the inverted signal to the differential amplifier, adding, or boosting it to the output. The 9.5 dB insertion loss of the filter allows the EQ module to provide a boost or cut of 9.5 dB, since the output op-amp compensates for the loss through feedback.

While the range of boost and cut may be increased by increasing the gain of the differential amplifier, raising it too much can cause the Wien bridge to begin oscillating.

Clip Indicator

The peak indicator as designed is built around an op-amp comparator and is able to measure multiple points in the input channel through the use of diodes. The output of the comparator is fed into a signal diode for rectification, and then into an RC network. This RC network charges up rapidly and discharges slowly, allowing the connected MOSFET to remain switched on for a longer period of time.

Without an input signal, the comparator is biased so that the inverting and non-inverting terminals have voltages of -460 mV and -1.02 V respectively, a difference of about 540 mV. The diodes feed into the voltage dividers between the 47 k resistors. In order for the indicator to light, the voltage at the non-inverting terminal must be greater than the voltage of the inverting
terminal. This occurs when the incoming signal exceeds the voltage at the diode input nodes and raises changes the voltage at the op-amp inputs.

Values for the RC network were chosen mostly through experimentation using the pulse function of the function generation and feel. In the end, a resistor value of 100k and capacitor of 680 nF was decided to give the best indication for even very short peaks while not remaining visible for too long.

Mixer (NG)

Inputs

The mixer possesses three inputs, allowing the connection of up to three input channels. To achieve this, three copies of the below circuit were used.

![Circuit Diagram](image)

*Figure VII: A schematic of an input as used in the mixer*

Passive High-pass Filter

The mixer features three single-ended inputs. A passive 6 dB / octave high-pass filter is used at each input to both block DC and provide a set input-impedance and path to ground. A capacitor of 690 nF and a resistor of 22 kΩ were used to create a high-pass filter with a -3 dB
point at 10.5 Hz. 22k was chosen as it is a high enough value to not load the output stage of the connected module or device, while not causing excessive noise.

Active Volume Controls

The mono signal is then fed into an active volume control. In order for a volume control to sound and feel natural to the user, the change in signal level measured in dB should be linear with the rotation of the control. Though passive logarithmic potentiometers are available for purchase, these crudely approximate a logarithmic curve by using two linear slopes, which create an inaccurate response and variable input impedance, both of which are unfavorable in professional applications. While more accurate logarithmic potentiometers do exist, they can be prohibitively expensive, especially when taking into account the requirement of multiple volume controls. For these reasons, an active volume control stage was used to create a control that responded linearly in dB. The design topology used was originally published by Peter Baxandall and requires only two op-amps, two resistors, and a standard linear potentiometer.

It should be noted that this volume control inverts the signal, so care should be taken when attempting to preserve signal phase.

Pan Controls

In order to convert the mono signal to stereo, the output from the volume control is then fed to the panning circuitry. In order to pan appropriately, the controls must respond in a sinusoidal / cosine curve. The circuitry should also provide a 3dB difference to the output signal when the controls are centered compared to panned to the extreme left or right. This 3 dB difference is necessary in order to preserve apparent loudness, since outputting an identical signal to two speakers results in an apparent 3 dB boost in level when compared to a single speaker.

To achieve this a single dual-gang linear potentiometer is used in conjunction with a negative impedance converter. The negative impedance bends the resistance of the potentiometer such that it more closely approximates a sinusoidal / cosine curve. The gain set by R7 and R8 provides the gain necessary to create the 3 dB difference when panning from center to either extreme.
Signal Summation

Figure VIII: A schematic of the summing amplifier as used in the mixer

The outputs of the channels are combined into a single pair of stereo signals by a summing amplifier. A summing amplifier was chosen over a passive summer in order to avoid a reduction in signal level and the higher resistors necessary to maintain a drivable input impedance. As before, higher resistor values are to be avoided in order to reduce the amount of johnson noise. This approach was also chosen over a more complex balanced design since the circuitry and signal runs weren’t very long to begin with. Because the summing amplifier is inverting, the signal is returned to its original phase after the volume control.

Sends

Figure IX: A schematic of the send buffer as used in the mixer
The send circuitry consists of dual unity op-amp buffers. A 100 ohm resistor is placed in series with the output to prevent oscillation caused by capacitance on cable leads. Two identical copies of this circuit were used to provide two stereo sends from the mixer.

Returns

![Schematic of the return buffer as used in the mixer](image)

The return circuitry receives the externally modified signal and consists of a passive low-pass and high-pass filter, a unity buffer, and a switch. The filters limit the bandwidth to 10 Hz through 26 kHz. R6 and R23 were chosen at 22k to provide a reasonably easy to drive load to externally connected devices. A dual-pole dual-throw switch allows selection between the original signal and the processed signal.

Two copies of this circuit are used to provide two switchable returns in the finished design. The output of the first switch feeds into the left input of the second switch.
Master Volume & Differential Output

Figure XI: A schematic of the master volume control and differential output as used in the mixer.

The master volume control consists of a similar circuit to that used for the inputs of the mixer. However, to achieve a differential signal, the output is taken from both op-amps. This design allows for logarithmic volume and differential outputs without requiring extra op-amps to create the differential pair. As with the sends, 100 ohm resistors are placed in series with the outputs to prevent oscillation.
Compressor (TP)

We include a compressor/limiter in our project and the purpose of the compressor is to keep the sound in a consistent level of dynamics. This means that whenever the signal is too loud, the compressor will lower the signal to an appropriate level. For the compressor, we intend to use a feedback architecture to construct the system and the system would consist of a gain control element which compress the sound according to a specific set ratio. The compressed sound then get feedback into a control circuit to determine whether the signal has exceeded the set threshold. After evaluating and modifying the input signal, the signal then goes through a makeup gain stage which then give the signal a predetermined level range of the dynamics.

One of the major component of the compressor is the voltage control amplifier (VAC), which is used to attenuate the level of the signal in response to the set level voltage. The VAC we built has a 60-dB dynamic control range and we are basing our model of the VAC using op-amps and JFETs.

![Figure XII: A simplified block diagram of the compressor](image)

For the gain control element, I implemented a topology including a n-jfet and an op amp to configure and control the dynamic level of the compressor. In this implementation, the n-jfet acts as a variable resistor that can control the threshold voltage of the system. In my implementation, the threshold voltage is 5.41 V peak-to-peak. I chose the n-jfet configuration is due to their instantaneous switching and isolation of the input signal and the control voltage. However, due to their unpredictable nature, when I built a second compressor for stereo, the problem I ran into was the difference in characteristics of each individual n-jfet and this in turn did not give me identical compressors for the system. In order to replicate an exact copy, I was trying to match different n-jfts to have the same characteristics to solve this problem. After the gain control element, we have an attack and decay control which refers to the amount of time the signal will be compressed and be recover. For the attack time, in the schematic below, which is being control by R12, is about 5 milliseconds. For the decay (or release) time, which is being control by R15, is about 1 second. This provides a compromise between a fast cut-off speed and not making the signal interruption too abruptly.

When building the compressor, it was hard to see the exact point where the signal was compressing and to solve this problem, I implemented a Schmitt trigger and a LED. The LED will light up when the voltage going into the trigger is 5.33 V peak to peak which is not perfect but it provides an approximation when the signal is being compressed.
In the schematic, to reduce the distortion created by the n-jfet and the op-amp, a noise cancelling circuit was implemented in the design which comprises of R11, R13, C4 and C5. The cancelling circuit has a much higher resistance compared to the control voltage resistance so that the attack and release time will not be affected in theory. However, when the circuit was built, some non-linearity can still be seen at the top of a compressed signal.

Sending the signal through the system loses some of the signal strength and therefore a voltage make-up gain was necessary to restore the signal to its original and send it back to the mixer. For the voltage makeup gain stage, I used an inverting op-amp configuration with the gain of 1.6.

Figure XIII: The schematic of the compressor circuitry
Amplifier (TP)

For the system, we decided on a class G amplifier with class AB for both the inner and outer drive. A class G amplifier is an enhanced version of the class AB amplifier allowing for greater efficiency and prevent the effect of clipping of signal. Compare to class AB, class G can have multiple rails and for our project, specifically we will have two rails, one, \( \pm 15 \) V, to operate the inner power drive and one, \( \pm 25 \) V, to operate the outer drive in the case when the signal exceeds the limit of the lower power rail. The trade-off by using a different level for the higher rail is whether how much head room we have to operate and the power dissipation in the system. Since most of the input signal will unlikely be higher than the low power rail, the class G amplifier will operate in its lower power amplifier for the majority of the time. While at this operation point, the higher power rail will turn off and conserve energy for efficiency purposes. Also with the higher power rail accompanying the amplifier, even at the low power operating point, efficiency also increases compare to a stand alone class AB amplifier. The amplifier works with little distortion with the signals’ frequencies ranging from 20 Hz - 20 kHz and this is to accommodate standard music input signals. Since class G amplifier generates a large amount of current, about 0.8 - 1.4 A therefore the power dissipation in the circuit will be relatively high, from around 20-30 W, heatsinks were used to prevent the circuit from burning up. Since for our project, we were using a built power supply, there are a lot of bypass capacitors at all the rails in order to reduce the noises from the power supply. The amplifier is itself can be considered a modular system with three components: the input stage, the voltage amplifying stage and the class G output stage.

Input Stage

**Supplies: 2N3904 and 2N3906 transistors, 1N914 diodes.**

For the input stage I implemented a differential amplifier with a pair of transistors. The input stage can be implemented using a single transistor but there are many pros that would justify the use of the differential pair. Compare to the single transistor, the differential pair has a low DC offset because the current does not flow through the feedback network. Also, the pair provides a more superior linearity compared to a single transistor configuration. According to the Audio Power Amplifier Design book, the input stage can also be implemented using FETs instead of BJTs, however, BJT is a better choice because the characteristics of BJT are more predictable compared to FET and BJT has higher transconductance compared to FET.

Using a current mirror configuration, I was able to lower the DC offset that would make the incoming signal unbalance and by proving this balance, we can eliminate harmonic distortion at the input level, which reduce noises at the output stage where every small noises would amplify greatly. The pair of transistors which includes Q1, Q2, Q5, and Q6 also have complementary negative local feedback with an degenerative emitter resistors (R3 and R4). By increasing these two resistors, linearity will improve however the gain would slightly decrease and the Johnson noises would increase. The local feedback system is used to reduce the DC offset and gain stability.
I used a current source to drive the input stage because with the current source, you can adjust the amount of current going through the transistors to reduce the slew rate according to the characteristics of the transistors. For the amplifier built in the system, the amount of current driving the input stage is 2.7 mA which can be adjust with R8. Increasing R7 can increase the linearity of the current source but introduce more noises to the system. Increasing the gain of the input stage, can have a stronger feedback but it also introduces more instability of the system. The gain can be adjusted by changing R8 or changing R5 and R6. For future project, the designer must take into account this trade off.

For the differential input stage, there are two inputs: Q1 and Q6. Q1 takes in the AC-coupled signal of the mixer and Q6 takes in a negative global feedback. Even though the system has some local feedback, a global feedback aims to reduce harmonic distortion with adjustments using C3. The global feedback also reduces the impedances of the system which can help drive the speaker with more power, and can also increase the power supply rail rejection ratio. The four diodes connection with C1 is to protect the feedback path against high current that can potentially damage the transistors of the input stages therefore reduces the amount of parts need to be replaced if something short out in the circuit.
Figure XIV: The schematic of amplifier's differential input
Voltage Amplifying Stage (VAS)

**Parts:** 2N3904, 2N3906, TIP35C power transistor, 100 ohms power resistor.

The output of the input stage is then feed into the base of Q10 for the VAS. The VAS is arguably the most important stage of the whole amplifier. The VAS provide the full voltage swing of the amplifier allowing the usage of the higher +/- 25V and to provide this voltage gain efficiently using a common emitter configuration. At this stage, frequency compensation also occurred and the component controlling this function is C2, also known as Cdom in many power amplifier topologies. The value of C2 varies and depend on the system. The C2 prevents high frequency oscillation at the output stage and to prevent the voltage gain of the amplifier to fall below one. In my implementation to increase the power supply rejection ratio, I used a current source to adjust the amount of current needed to drive the voltage amplifying stage. Using the current source allows the designer to adjust with more ease. With this configuration, the amount of current driving the VAS is about 16.66 mA. The outputs are at different bias points to drive the transistors at the class G output stage.

For the biasing scheme, I used four 1N914 diodes and a VBE multiplier with R11 a 10K rheostat. With the VBE multiplier, I can change the voltage bias easier and this would help with the DC offset at the load. The VBE multiplier is also used to prevent crossover distortion at the class G output stage. The voltage biasing scheme also vital to turn on and off the different drivers of the next stage in the output stage. In theory, using the assumption that the diode is 0.7 V drop, if the top signal is higher than two diodes drop higher than the input point going into the output stage, the outer drive would turn on and drive the signal with the higher voltage rail (+/- 25V). If we set the voltage bias too low or too high, beyond the point mentioned, problems such as delays in operation of a drive or too much heat reducing the efficiency of the amplifier. In practice, however, the working of the biasing was not simple due to the mismatch in the diodes’ characteristics. At the beginning of building the biasing scheme, I first used a string of eight diodes however, through trials and errors, using a VBE multiplier improves the design tremendously compared to a string of 4 diodes. In order to improve this biasing scheme further, it is best to apply the same scheme, the VBE multiplier, across the other two pairs of diodes so that we can freely adjust the biasing of the VAS.
Class G Output

Parts: 2N2219 and 2N2905 switching transistors, TIP35C and TIP36C power transistors, MUR460 rectifier diodes, 1N4001 Rectifier diode, two 10 ohms power resistors

The class G output stage consists of two power devices: an inner power device and an outer power device. In this implementation, based on recommendation of the Audio Power Amplifier Design Book, the inner power device is a class B amplifier and the outer power device is a class C amplifier. Using a standalone class B or class C has different problems. Class B has crossover distortion but gives a decent amount of power. On the other hand, class C can be extremely efficient but generates less power compare to other amplifiers. When the two drives combined together, we have a more powerful amplifier compare to class AB along with a slightly
less efficiency. The upper and the lower transistors (Q15 and Q18) turn on depending on the biasing at the previous stage. The biasing in a sense push the signal up to the outer drive or pull down toward the inner drive. To be efficiency, the upper power rails are not used since most music signal will not get past above that point. This stage acts like a buffer stage where most of the current gain is provided. To further improves the swing of the class G output, the emitter degeneration resistors should be as small as possible. However, these two resistors also keep the transistors from burning up so no resistors is not advise for improvements. At the output there is a global feedback which you can determine the voltage gain using the feedback resistor. The feedback provides a voltage gain of approximately two. Since class G is a power amplifier, we are more concern about the current gain. If we can increase the current going into the load (by reducing the R18 and R19) the amplifier can drive the speaker much louder. A Zobel network was also implemented to neutralize the effects of speakers’ inductance.

Figure XVI: A schematic of the class G output stage circuitry.
Figure XVII: The completed schematic of the class G amplifier
Power Supply (EA)

In order to ensure that the system is portable we must furnish it with its own power supply running on the wall AC input. The design of the system calls for the use of two sets of power rails, ±15 V and ±30 V. The ±15 V rails are used for delivering power to the input channels, mixer, compressors, and class G amplifiers; this rail also drives the output speakers at lower volumes. The ±30 V rails are meant to provide additional power to the Class G amplifiers when required in order to improve efficiency. In order to minimize the space taken up by the power supply, the project makes use of the +30V rail as the source for the ±15 V rails, this allows us to cut the number of transformers from 4 to 2.

Figure XVIII: A schematic of the ±30 V rail regulators.

The above schematic describes the design and operation of the ±30 V rails. Each is constructed making use of one transformer providing an approximate AC Voltage which then undergoes linear rectification through a bridge rectifier. The rectified signal is then smoothed by use of capacitors. While the above schematic lumps the total capacitance into one 11µf capacitor, the actual design calls for 11 separate parallel 1µf capacitors. This arrangement takes into account the ESR, Effective Series Resistance, that each capacitor provides. The parallel arrangement of these capacitors ensures that the ESR seen by the circuit is minimized, improving power efficiency and minimizing ripple. The smoothed voltage is then fed through a voltage regulator, a darlington pair TIP driven by a BJT feedback loop, which adjusts the output current to maintain the output voltage constant.
The second stage of the power supply consists of both ±15 V rails. The schematic above illustrates how the +30V rail is converted to the lower rails by use of a buck converter topology. The LT1074 is a step down switching regulator which takes in feedback to adjust the duty cycle associated with the desired output voltage. The buck topology uses an inductor to generate a back EMF which supports a voltage across it, the LT1074 acts like a switch connecting and disconnecting the power source, this process changes the rate of change of the current through the inductor and thus the voltage across it. In regulating the duty cycle of operation we can ensure that a specific voltage appears at the output and is maintained through feedback detecting changes in the current being drawn at the output. This topology is highly
efficient which allow us to minimize power dissipation. Like in the ±30 V rails the displayed 8mf capacitor represents a lump capacitance accounting for ESR.

There were several considerations when designing and implementing this power supply. It was essential to consider the power dissipation that would occur throughout the circuit, in the case of the TIP darlington pairs a voltage drop of 3 volts at up to 2 amps meant that heatsinks would be required. The buck converter topology required the use of a high power diode which had to endure high reverse voltages and high forward currents which called also for heatsinking. Issues related to the amount of ripple detected in the circuit did not become apparent until testing of the +30 volt rail, where a ripple of almost 1 volt threatened accurate operation. This resulted in an analysis of the capacitor’s ESR and was fixed by doubling the number of capacitors, this change is reflected in the final schematic. Testing of the -15V rail revealed an undesired signal related to the switching of the LT1074. This low frequency signal was alleviated by the use of bypass capacitors in the class G amplifier as well as making use of shorter leads coming out of the power supply. The signal however persisted at an audible level, it is recommended that in future uses of this design a different integrated circuit be used for the -15V rail. The transformers used had a larger than expected ESR which resulted in lower than expected output voltages at higher currents (> 2 amps), this made the effective voltage rating at high volumes drop to around ±25 Volts under heavy current drain, in order to compensate for this, some of the class G amplifier biasing was redesigned to use ±25 Volts as the expected power supply. This lower voltage rating was not noticeable at the volumes tested in the lab and would require enhanced testing to further minimize.

Enclosure (Stretch Goal) (EA)

The decision of making the system portable added the complexity of designing an appropriate enclosure for transportation and utilization. This enclosure provides the housing for all of the circuit components and provides the user with an appropriate interface with all of the system’s functions. The design called for the use of a lightweight material, given the choices available in EDS, Engineering Design Studio, black acrylic was selected due to its aesthetics and availability.

The frontal panel was implemented at an angle of 60 degrees to provide the user with an interface that can be manipulated from a wide range of relative positions. All appropriate apertures were cut making use of a laser cutter, labels were also rastered making use of the same laser cutter. This panel was made making use of ¼ in acrylic to ensure that it laid flush with the thicker panels used in the rest of the enclosure.

The remaining panels for the enclosure were also cut and rastered making use of the same laser cutter. The side panels include a series of vertical apertures designed to allow airflow to cool the internal components. The horizontal apertures act as handles for transport. The back panel has also several outputs which are assumed to be less commonly used and where placed here to minimize the clutter of the interface in the front panel. The box was then
built together making use of screws at predetermined locations. $\frac{1}{4}$ in acrylic was used for all of these panels.

Several issues were encountered when constructing the enclosure. Due to the relative brittle nature of acrylic, some of the drilling efforts into it resulted in small parts of the acrylic to break away. The use of thinner acrylic for the front panel resulted in flexing of this panel during operation. The last minute decision to introduce a stereo setup to the system resulted in the interior of the box to be unexpectedly packed. The third input channel circuit was never built and thus masking tape was used to cover up this panel, the result was aesthetically pleasing.

Future incarnations of this enclosure would ensure that a larger volume is used, about 15% larger to compensate for the added circuitry. The introduction of some form of active cooling would remove any concerns regarding the lifetime of the system. A more scratch resistant material would serve the design.

Figure XX: The panel designs for the acrylic enclosure