The "PA System in a Box" is a fully functional personal-audio system with the features found on professional audio equipment but without the setup and transportation requirements of a full multi-part PA system. To achieve this, all the modules necessary for a complete system will be neatly packaging into a single portable enclosure. We hope to accomplish the build and test stages soon so that we may have custom circuit boards built for the project.

The workings of audio PA systems lend themselves to modular design; as a result we’ve managed to break our project into a number of functional blocks that can work independently of other components in the system. These components are themselves designed modularly, consisting of multiple sub-components. Each component is discussed in detail after the block diagram.

**Input Channel (Line / Microphone) (NG)**

The microphone and line input channel allows users to connect a microphone or line level signal source with a balanced or unbalanced connection and increase or reduce the signal level to professional line level (+4 dBu). It expects single ended or differential input signal in the range of 10 mVrms - 7 Vrms Balanced or Unbalanced Mono.
10 mVrms through 7 Vrms. It will output a single-ended professional line level signal of 1.2 Vrms (+4 dBu) nominal. The input channel should have a bandwidth of 20 Hz through 20 kHz to allow signals in the full audible range while reducing higher frequency noises. The module will require ±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 our system.

   Each input channel will consist of a number of simpler modules, including a preamp, lowpass and highpass filter, single band parametric EQ, and a visual LED peak indicator.

   The module will be designed, built, and tested by Nathan. In order to test the input channel, signals will be produced using function generators, microphones, and line level devices (like phones) and fed into the channel. Various setting configurations will then be tested and the results will be recorded. There should be a configuration of settings that allows the output to reach nominal +4dBu levels without distortion or excessive bandwidth reduction.

   Multiple channels are required to allow for multiple inputs. Our current goal is a minimum of 3 copies in order to allow a stereo line-level device (like a phone) and a microphone to be used at the same time. Dual gang 10k potentiometers will be necessary for some functionality, but can be uncommon.

Mixer (NG)

   The mixer is where the signals from the multiple input channels are combined. It receives the single-ended mono output of each input channel and allows the user to attenuate and pan (convert from mono to stereo) the signals while combining them into a stereo bus. It expects single ended input signals at a voltage of 1.2 Vrms (+4 dBu) nominal. Within the mixer, signals are expected to average around +4 dBu. The mixer will output stereo single-ended +4 dBu nominal signals to its connected components while also offering stereo differential +4 dBu line-level outputs to interface with outside recorders. The mixer should not have any effect on the bandwidth of the system and is expected to have a flat response from 20 Hz to 20 kHz. The mixer will require ±15 volt rails to supply the op-amps.

   This module will be designed, built, and tested by Nathan. To test the mixer, line level signals from function generators and other music devices will be fed into the module and the signal will then be measured at various points throughout the mixer, including the outputs, to verify that there has been no reduction in bandwidth or increase of noise or distortion on the signal.

Compressor (TP)

   We intend to include a compressor 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 and whenever the signal's level is too low the compressor acts like a booster providing gain, making the sound louder. 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. Our goal is to have at least three sound compression ratios???

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
intended to build would have a 60-dB dynamic control range and we are basing our model of the
VAC using op-amps and JFETs. Using this linear model we can expect an error of around 1% of
the dynamic range. We are also considering a transconductance amplifier replacing the gain
resistor in an op amp configuration.

To determine the functionality of the compressor, we would use the function generator to
input signals at different levels of amplitude and then we can calculate the maximum value and the
minimum value the compressor will output to determine the range of the system. We can also test
how responsive the compressor is by fluctuate quickly the incoming signal at extreme range.

Graphic EQ (EA)

The graphic equalizer will allow the user of the system to compensate for sonic
deficiencies in the speakers or space by providing boost and cut controls for predetermined
frequencies before they reach the amplifier stage of the system. This will be achieved by using a
series of multiple-feedback bandpass or sallen-key bandpass filters. The graphic equalizer will
accept a single-ended two channel signal at +4 dBu nominal and will output a single-ended two
channel signal again at +4 dBu. The bandwidth of this component should be 20 Hz to 20 kHz, as
to maintain the human audible range and not reduce to bandwidth of the system. The individual
frequencies that may be boost and cut will likely be spaced at least an octave apart from the
previous and following filter. This module will require ±15 volt rails to supply the op-amps.

This module will be designed, built, and tested by Edwin. To test the graphic equalizer, a
function generator set to the appropriate frequencies will be used to test the filter values and gain /
reduction. A musical test will be performed by playing music through it and listening to whether the
selected frequencies were useful. During this time it will also be verified that the system’s
bandwidth is maintained and that there is low noise and distortion.

In regards to special parts; if one set of knobs controls the EQ for both channels, this will
require dual gang 10k linear potentiometers. However, if the left and right channel may be
adjusted separately, then single gang 10k linear potentiometers are required. Having both
channels adjusted at the same time is more user friendly, and will cover most use cases, while
individually adjustable channels allows for more flexibility but at the expense of user friendliness.
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, ±15 V, to operate the inner power drive and one, ±35 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. We also want the amplifier to work with signal's frequencies ranging from 20 Hz - 20 kHz ideally to accommodate the input signals. For this range we want the signal to be clean and stable to prevent from hearing buzzing sounds. Since class G amplifier will generated a large amount of current, about 1.8 - 2.2 A therefore the power dissipation in the circuit will be relatively high, from around 25 - 40 W, and heatsinks will be needed to prevent the circuit from burning up.

One of the major problems is dealing with linearity of the amplifier when switching from low to high power rail. The diodes switching time will create noise at the output and will create buzzing sound at the speaker. Heat dissipation will be another problem concerning driving a high current for the speakers and depend on how the circuit will be implementing in details, with high level of heat, the amplifier will create noises within the system and can potentially burn out the components of the amplifier.

We can test the amplifier by input signals from the function generator. One of the very first challenge for building the amplifier will be testing a working amplifier with the low power supply rails at 1 kHz signal and then gradually increase the frequency of the signal to 20 kHz or greater to test the stability of the amplifier at a high frequencies. This will be challenging due to the limitation of the transistors. Then to test the high power supply rails, we have to use an input signal that can exceed the low power supply rails and since at the high power rail, heat dissipation will be greater therefore noises will increase.

LED Bar Graph Metering (TP)

Metering is essential to any audio system to provide visual feedback to the operator and let the user know how close to the system’s limit they are. To implement metering, we will use an LED bar graph display to show the independent output signal levels for both stereo channels. Each pair of LED’s will correspond to a preset signal level, allowing the user to know the strength of the summed signal in the mixer.

To reduce the number of power rails that the project requires, we will be operating this module off the ±15 V rails. However since there may be switching noise due to the digital-like
behavior of this circuitry, its ground will be kept separate from the signal grounds and will have its own lines routed to the power supply.

This module will be designed, built, and tested by Tuan. To test it, a variety of input signals at different levels will be generated using the function generator. The levels presented on the meter should then correspond to the levels being generated by the signal generator (after \( V_{rms} \) to dBu conversion).

Due to the lack of special design in this portion of the circuit (it can be constructed as a number of op-amp comparators), we may source out a metering chip that provides this functionality for us.

**Power Supply (EA)**

In order to ensure that the system is portable we must furnish it with its own power supply running on the AC wall input. The design of the system calls for the use of two sets of power rails, ±15 V and ±35 V. In order to minimize the space taken up by the power supply, the project will make use of one single high voltage rail from which the four power rails will be derived. This will be accomplished by making use of buck converter topologies in tandem with a standard linear AC-DC rectifier.

The challenge associated with this design is related to the amount of noise in the power supply. There will be a need to use filters and associated circuitry to remove as much noise as possible and also minimize ripple voltage from the output of each one of the rails. This task is compounded by the use of one rail to power the lower rails in the system.

This power supply approach has the advantages of ensuring modularity and reliability in the case of failures of different rails. It also allows the work in the other components of the system to go unimpeded until the power supply has been thoroughly tested.

**Enclosure (Stretch Goal) (EA)**

The enclosure of the system will house all the components of the system and place. The goal of our project is to make the system portable so the user can carry from and to places with ease. The enclosure will allow the users to adjust settings fairly easily, while protecting and allowing cooling for the internal components. To achieve this, we plan on constructing a plexiglass enclosure using laser cutting to create the shapes necessary.

This will likely be a collaborative effort and likely the final portion of our project, since its dimensions and layout will depend on the final designs of the modules being installed.

**Headphone Amplifier (Stretch Goal)**

It is sometimes required to use an audio system with headphones, such as when recording or rehearsal. The headphone amplifier will receive a stereo +4 dBu signal from the mixer’s output bus and amplify it to drive the stereo headphones. Output current should be limited in order to provide user safety and not cause damage to connected devices. Because headphones are relatively efficient for a given volume, this module will only use the ±15V rails, and not the higher
ones. Bandwidth should reach from 20 Hz through 20 kHz as to not limit what is audible, and
noise should remain low enough to be unnoticed during typical usage.

This module will be designed, built, and tested by whoever finishes their previous parts
first. In order to test it, a line level signal at various frequencies will be fed into the amplifier and
measured while connected to a dummy load. This load will then be replaced by a small speaker,
and finally with actual headphones once it is known to be operationally safe.

Hi-Z Input Channel (Instrument) (Stretch Goal)

If time allows, this module will allow the user to connect high-impedance sources, such as
guitars and other instruments, to the mixer without a loss of signal fidelity. It will be nearly identical
to a standard input channel except it will feature a single-ended high impedance input instead of a
balanced input. Power, bandwidth, and other requirements will remain the same.

This module will be designed, built, and tested by whoever finishes their previous parts
first. It will be tested in much the same way, except we will use real instruments (including a bass
guitar) to test the input.