LABESCAPE ULTRASONIC DIRECTIONAL SPEAKER

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Abstract

We designed, constructed, and tested an ultrasonic directional speaker. Our design uses a microcontroller unit that interfaces with a digital-to-analog converter receiving an audible signal and a waveform generator that creates an ultrasonic carrier signal. Our design modulates the audible signal onto the carrier signal using amplitude modulation, amplifies the resulting signal, and outputs it with an array of ultrasonic transducers. We constructed this entirely on a breadboard, except for the transducer array, which we soldered onto a printed circuit board. We found that our output signal is intelligible, directional, and equalized, confirming functionality. However, we found that the signal is quiet, which we hypothesize is due to the rail voltage limitation of the amplifier in our design.

Contents

1. Introduction

This project is a directional speaker, meaning a speaker whose output can only be heard in a narrow beam in front of it. Its purpose is for use in the LabEscape escape room, pitched by Professor Paul Kwiat, where a clue can be hidden for an escape room participant in a narrow section of the room. In our research, we found that a directional speaker can be created by sending an audio signal modulated onto an ultrasonic carrier signal through an array of transducers; with sufficient sound pressure achieved, greater than 110 dB, the audio signal demodulates in the air, becoming audible; directivity is achieved since the ultrasonic carrier has a small wavelength [1].

As shown in Figure 1, our design uses the ESP32 microcontroller unit programmed to receive an encoded audio signal, such as a voice speaking, which interfaces with the AD5761 DAC to convert it to an analog audio signal. The ESP32 also interfaces with the AD9833 waveform generator to generate the 40 kHz carrier signal. We achieve amplitude modulation with the AD633 analog multiplier, which receives the audio signal and carrier signal. Subsequently, our design filters the modulated 40 kHz signal with a band-pass filter that utilizes the TL072 operation amplifier, which is then amplified by the LM2876T amplifier to achieve sufficient amplitude corresponding to sound pressure greater than 110 dB. Our transducer array transmits the amplifier's output, which demodulates the audio signal in the air, becoming audible and directional. Our design also includes an inductive load in parallel with the transducer array to match impedance. We use a DC Power supply to provide ±22 V to the amplifier; we regulate down the 22 V and -22 V power rails to 15 V and -15 V (for the TL072, AD633, and AD5761) using the L7815 and L915 linear voltage regulators, respectively. We also regulate down the 15 V power rail to 5 V using the L7805 voltage regulator for the ESP32, which has an onboard 3.3 V regulator, and the AD9833 receives 3.3 V.

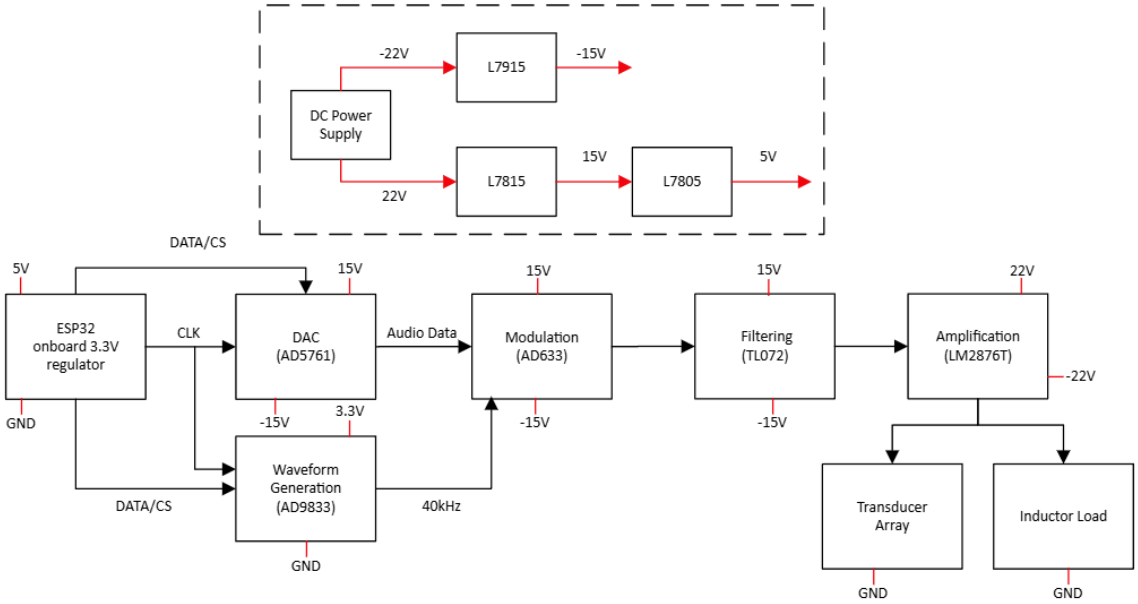


Figure 1 Block diagram.

Our ultrasonic directional speaker has three measures of high-level functionality. The first high-level requirement is that it should maintain half-power beamwidth of less than 10 degrees at a distance of 1 meter. This requirement is important, as it ensures that the speaker is directional enough to be used in the escape room. The second high-level requirement is that it should maintain an audio output frequency response flatness of ±6 dB or less within an operating bandwidth of 300 Hz to 1250 Hz. This requirement is important, as it indicates that the output audio is equalized, meaning that neither high nor low frequencies overpower each other, ensuring that the audio quality is not distorted. The third high-level requirement is that the speaker achieves a sound pressure level of at least 110 dB, which ensures that the audio signal demodulates in the air, becoming intelligible.

2. Design

2.1 ESP32 Carrier Wave Generation

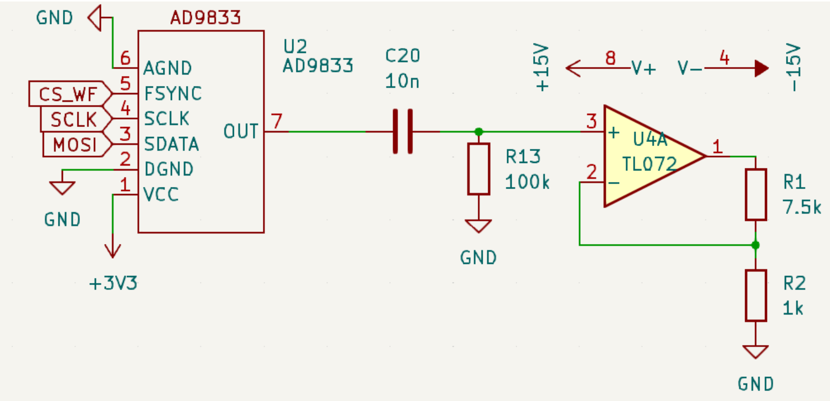


Figure 2 ESP32 carrier wave generation schematic. Showcases the AD9833 output pins, output of the system into the RC filter, and TL072 non-inverting amplifier.

Shown in Figure 2 above, we designed this subsystem to generate the ultrasonic carrier wave to be used in our amplitude modulation subcircuit. In our system, the frequency of this wave was 40kHz. In terms of signal progression, we connected the AD9833 programmable waveform generator chip to an ESP32-S3-WROOM development board with an on board 3.3 V regulator. We powered the ESP32 by a 5 V output from the TL7805, which turned on the microcontroller as soon as the full system was powered on. Before any operations occurred, we programmed delay statements in Arduino software to ensure that the ESP32 would be fully settled before trying to program the AD9833. For the wiring, we connected pin 13 of the ESP32 to the FSYNC pin of the AD9833, acting as a chip select. We idled high the chip select bit as data can only be written while FSYNC is low. Pins 11 and 12 were shared pins between the carrier wave generation subsystem and the DAC subsystem. We used pin 11 for the MOSI, or data line, while we used pin 12 for the system clock (SCLK). For the carrier wave generation subsystem, we connected the MOSI line to the SDATA pin of the AD9833, and we connected pin 12 to the SCLK pin of the AD9833. Finally, we grounded both the analog and digital ground pins, and we set VCC to 3.3 V to interface with the ESP32 correctly. To set an output frequency and waveform type, we wrote the chip select bit to low to begin the writing process. Then, we performed two 16-bit writes. The first two bits simply signal to the system's internals that we are writing a sinusoidal wave and that the next 14 bits represent the data. Doing this twice, we end up with 28 bits of data. This 28 bit code encodes a specific frequency based on the following equation per the AD9833 datasheet [4]:

$$f_{out} = f_{code}\frac{f_{MCLK}}{2^{28}} \tag{1}$$

In Equation 1, f_{out} is the desired output frequency of 40 kHz, f_{code} is the 28 bits of internal data in decimal form, and f_{MCLK} is from a built in oscillator of 25 MHz. We calculated the correct frequency code in software for the desired output frequency. Now that the wave is output, it is then input into a passive RC high-pass filter with a frequency cutoff of 159 Hz using a 10 nF capacitor and a 100 kΩ resistor. This removed the DC offset on the output of the AD9833 to center the carrier wave around 0 V. After this filter, we amplified the magnitude of the carrier using a non-inverting op-amp amplifier configuration using a TL072 operation amplifier. We used this to achieve a gain of about 8.5, which moves the initial amplitude of 300mV to around 2.5V. This is used to increase our modulation index for improved audio quality. This is then input into the amplitude modulation circuit.

2.2 ESP32 to Digital-to-Analog Converter

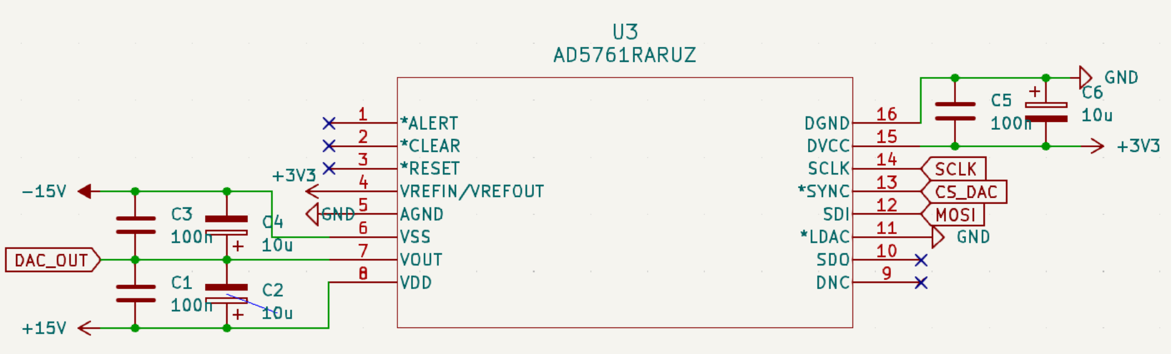


Figure 3 ESP32 to DAC schematic. Showcases the AD5761 ESP32 pin connections and decoupling capacitors along the power lines.

Shown in Figure 3 above, the ESP32 to DAC subsystem achieved two distinct outputs. The first was an audio wave of Sam's voice counting the digits 1 through 5. This was to showcase the audio quality possible for general audio capabilities and to determine and characterize what sound resolution is possible. The second was an audio array encoding a sinusoidal wave, which provided a constant audio tone. This was used to provide consistent audio to best showcase the directionality of the speaker. Both audios were stored in the flash memory of the ESP32, pulled element by element in a for loop in Arduino software. For the hardware of the circuit, we used the AD5761 since it uses dual 15V rails, which matches our amplitude modulation circuit, reducing the need for additional regulators. Regarding the wiring to the ESP32, Pin 9 was the chip select pin, which connects to the SYNC pin on the DAC. SCLK was connected to the shared SCLK line on pin 12, and SDI, or serial data in, was connected to the shared MOSI line on pin 11. Since we did not do readback, and we would reset the whole system at once, we left the ALERT, CLEAR, RESET, and SDO pins disconnected along with the DNC pin. DVCC and VREF were both set to 3.3V, since DVCC needs to be 3.3V to interface with the ESP32, and VREF was conveniently the already accessible 3.3V. Both digital and analog ground were grounded, and VSS and VDD were set for ±15V respectively with capacitors as per the datasheet typical applications circuit [3].

In this system, we first set the DAC operation mode. In our system, we set this output to be the ±10V mode. It's important to note that while the name for the mode of our operation is listed as ±10V in the datasheet, the datasheet assumes a reference voltage of 2.5V. Since we have a reference voltage of 3.3V, we technically operate up to ±13.2V. After setting up the operation mode, we can begin writing voltages to our system. We do this by reading a 16 bit data value, writing x03 to the DAC, which is the operation code for "write and set", and then sending the 16 bits of data to the internal DAC register. This then reads the 16 bit data code, and it outputs a voltage based on the code. This output voltage is determined by the following equation:

$$V_{out} = V_{ref}\left(8 * \frac{V_{code}}{2^{16}} - 4\right) \qquad (2)$$

Equation 2 finds the ratio between the applied code and the maximum code to find out where a data point should be placed, scaled, and shifted to center the response around 0 V, subsequently scaling the response to ±13.2 V. We then apply this transfer function to all the data points in our array, providing a positive or negative value based on the value stored in the audio array. This audio output goes through a smoothing RC low-pass filter composed of a 10 nF capacitor and a 10kΩ resistor, yielding a cutoff frequency of 1591.5 Hz.

2.3 Amplitude Modulation

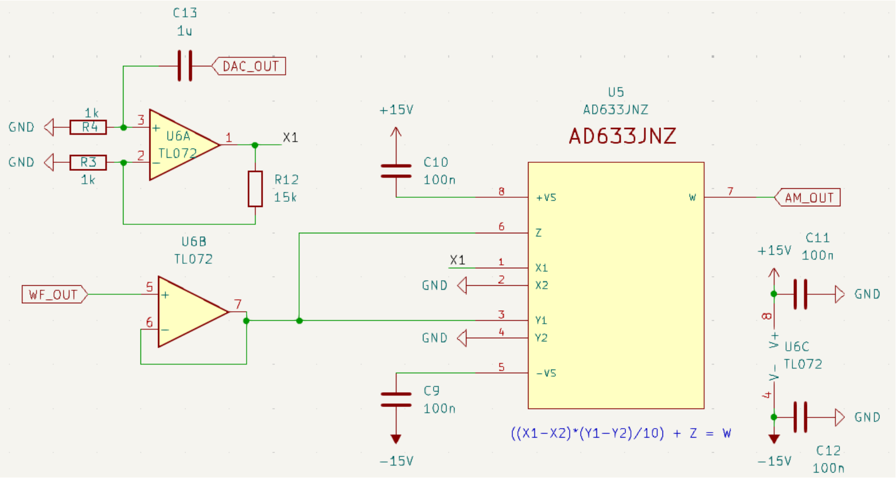


Figure 4 Amplitude modulation schematic.

We designed the amplitude modulation subsystem using the AD633 analog multiplier due to its low cost, precision, and ability to handle ultrasonic frequencies. To ensure the integrity of the signals entering the modulator, we conditioned both the carrier and audio signal paths. We used the TL072 operational amplifier in the conditioning stage, since compared to other operational amplifiers, it performs best with ultrasonic frequencies.

We processed the carrier signal through a voltage follower configuration using the TL072, as shown in Figure 4. This stage implements impedance isolation between the waveform generator and modulator inputs, ensuring that the low-impedance output of the op-amp could drive the parallel inputs of the AD633 without phase distortion or signal attenuation. This output of the TL072 is shorted to the Y_1 and Z inputs of the multiplier, as shown in Figure 4.

We conditioned the audio signal by first applying a passive high-pass filter consisting of capacitor C_{13} (1 µF) and resistor R_4 (1 kΩ) to the non-inverting input of the other side of the TL072, as shown in Figure 4. This filter serves to block low-frequency noise and DC offset originating from the digital-to-analog converted, with the cutoff frequency being 159.15 Hz. Following the filter stage, we amplified the audio signal using a non-inverting amplifier configuration, selecting the feedback resistor R_{12} (15 kΩ) and reference resistor R_3 (1 kΩ), as shown in Figure 4, to provide a gain of 16 V/V. The gain scales the audio signal to the 10 V reference level required by the AD633 for optimal modulation depth.

The output of this amplification stage, X_1, serves as the modulating input for the AD633. To achieve amplitude modulation, we configured the AD633 based on its transfer function [4]:

$$W = \frac{(X_1 - X_2)(Y_1 - Y_2)}{10V} + Z \tag{3}$$

We grounded the differential inputs X_2 and Y_2, so that each input is simply the audio and carrier signal, respectively, as shown in Figure 4. After substituting the carrier signal for both Y_1 and Z, which allowed us to factor the carrier out of the expression, we obtain the equation:

$$W = (1 + \frac{X_1}{10\ V})\mathrm{Y}_1 \tag{4}$$

This configuration results in a “carrier-plus-sidebands” signal, where the term 1 represents the constant carrier component and X_1/10V represents the modulating audio envelope. The resulting output signal preserves the linear relationship between the input audio and the output envelope, which is necessary for clear sound reproduction.

2.4 Bandpass Filter

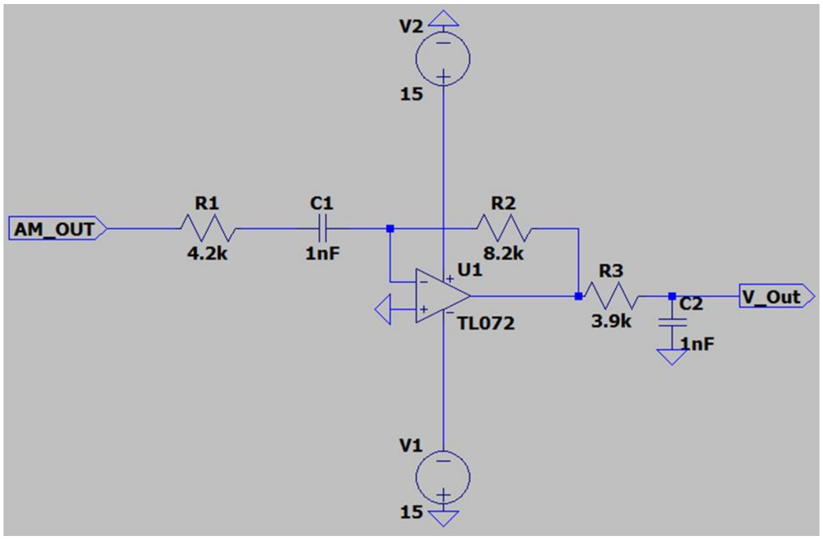


Figure 5 Bandpass filter schematic.

As shown in Figure 5, the filter is an active bandpass filter, which includes a TL072 to keep the signal with a higher peak-to-peak value and prevent any voltage loss that may occur through a passive filter. First, the signal travels through the high pass filter, which is connected to the amplifier, and using the feedback resistor, it has the magnitude of the gain set at approximately 1.952. It is important to note that the inverting amplifier configuration is being used, so the signal has a phase offset of 180°. The offset does not affect the audio signal at all, so it is acceptable. After going through the amplifier, the signal is sent through a low-pass filter to complete the bandpass. Once the signal was sent through the bandpass filter, it was sent into the amplifier subsystem. The cutoff frequency for the high pass filter was 37894 Hz, while the low-pass filter had a cutoff of 40809 Hz making the total bandwidth from 37894 – 40809 Hz. The following Equations were used to calculate the cutoff frequencies and the gain:

$$Gain \;=\; -\left(\frac{R_2}{R_1}\right) \tag{5}$$

$$F_{cutoff} = \frac{1}{2\pi RC} \tag{6}$$

The TL072 was a good fit for the bandpass filter as it offered low distortion to the signal, while allowing proper amplification with the bipolar rail voltage. Other amplifiers, such as the LM741, distorted the signal and added too much noise.

2.5 Amplifier

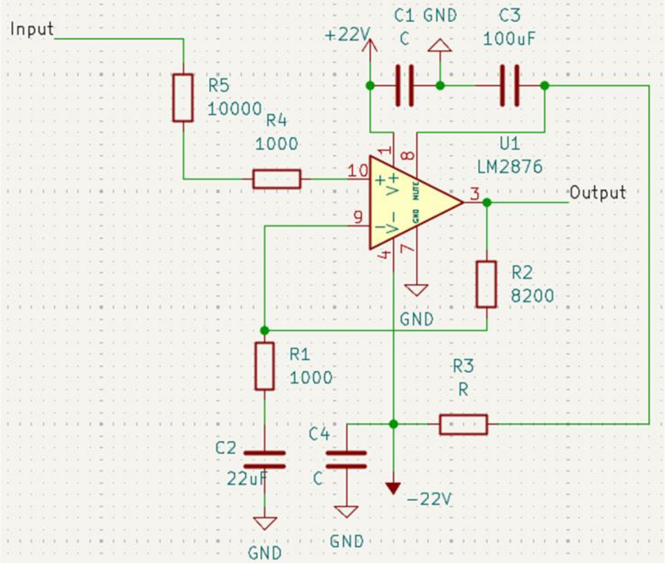


Figure 6 Amplifier schematic.

The amplifier subsystem used the LM2876T for the amplifier chip, as it allowed the signal to reach the necessary peak-to-peak voltage to have an audible signal. As shown in Figure 6, the amplifier takes the input signal that previously went through the bandpass filter. It then amplifies the signal and sends it to the transducer array. For the circuit, the design comes from the “typical usage design” from the LM2876T datasheet [5]. A few things to note about the design are that the R5 in the schematic is a potentiometer that allows us to tune how strong of an input we want into the amplifier. Then, the capacitors connected to the positive and negative rail voltages prevent motorboating or the rail voltages jumping up and down at these high frequencies. The gain we chose was 9.2; this is because the amplifier is set up in the non-inverting configuration, meaning the gain equation is:

$$Gain = \;1 + \;\left(\frac{R_2}{R_1}\right) \tag{7}$$

The amplifier in the system needed to hit the necessary peak-to-peak voltage to output an audible sound, so that the air would de-linearize the 40 kHz signal. Then, it needed to run without overheating while not distorting the signal. We chose the LM2876T since it matches these descriptions and properly amplifies the signal at 40kHz, whereas other amplifiers struggle to control signals at 40 kHz, often creating noise or distortion, while those that can amplify the signal at 40kHz struggle with hitting the peak-to-peak voltage to make the signal audible. The LM2876T was the perfect choice because it could do both things without overheating.

2.6 Transducer Array with Impedance Matching

We designed a 4-by-4 symmetrical array consisting of sixteen piezoelectric ultrasonic transducers. We chose the CUSA-T60-150-2400-TH transducer model, since it operates at a typical resonance frequency of 40 kHz, and it is more directive with a higher typical sound pressure level attained than other transducers we compared it with [6]. We soldered all sixteen transducers in parallel (ensuring a uniform phase across the entire emitter face) on a custom printed circuit board, as shown in Figure 7.

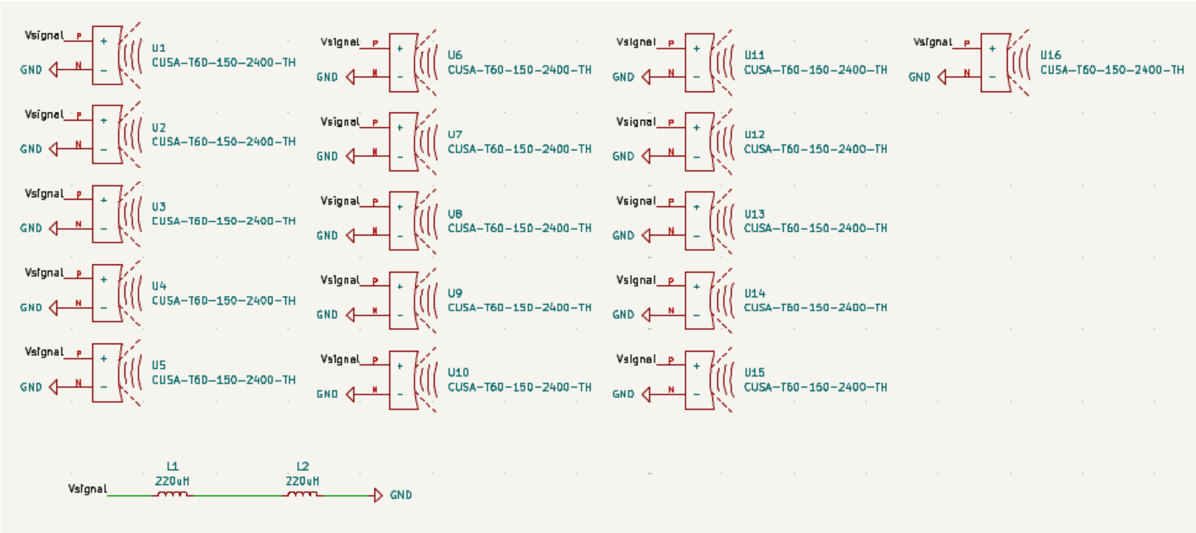


Figure 7 Transducer array with impedance matching schematic.

Since each individual transducer has an internal capacitance of approximately 2400 pF [6], the parallel combination resulted in a significant cumulative capacitive load of 38.4 nF, calculated as the sum of the individual unit capacitances. This high capacitance would normally lead to high reactive current flow, and it could potentially induce high-frequency oscillation and thermal instability in the LM2876T amplifier, due to the phase shift it introduces. To counteract this, we utilized a parallel impedance matching network. We calculated the necessary inductance:

$$L = \frac{1}{C_{tot}(2\pi f)^2} = \frac{1}{38.4\,nF \cdot (2\pi \cdot 40\ kHz)^2} \approx 412.3\ \mu\text{H} \tag{8}$$

As shown in Figure 7, we placed two 220 µH inductors in series with one another and placed this series combination in parallel with the transducer array, approximately matching the 412.3 µH system inductance. We effectively created a resonant circuit at 40 kHz. By aligning the inductive reactance with the capacitive reactance, we ensured that the LM2876T amplifier perceives the transducer array as a primarily resistive load at the carrier frequency, shifting the power factor towards unity and ensuring the stability and signal fidelity of the amplifier output.

2.7 Power Supply and Regulation

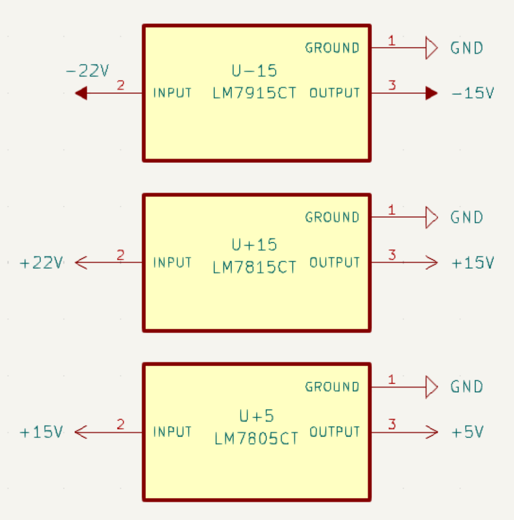


Figure 8 Voltage regulation schematic.

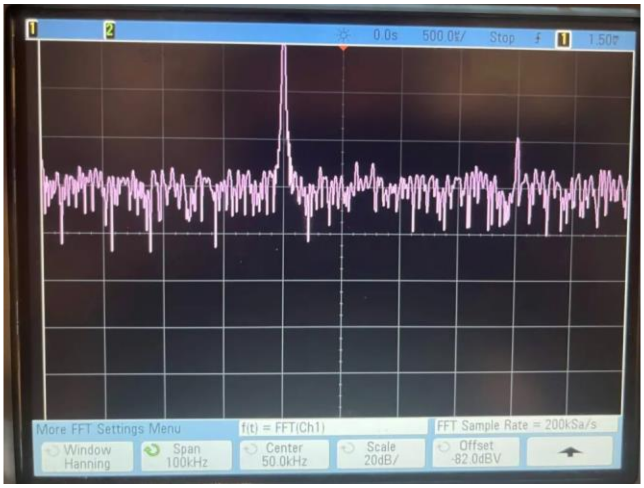
Regarding the power supply and regulation shown in Figure 8, we used two DC power supplies, as they supplied the ±22V. We probed them using a digital multimeter to make sure that they were ±22 V in reference to ground. Once we tuned them to that magnitude of 22 V, we connected them to the circuit and the LM2876T. After that, we used voltage regulators from the LM78 series to regulate the voltages down to positive 15 and 5 V, then from the LM79 to regulate down to –15V. We chose all these regulators and power supplies because they entailed no further cost, being found in the self-service shop or supplies we already possessed. These components were also easy to use and test with a multimeter to ensure safety.

3. Design Verification

As we constructed and tested each subsystem, we verified that our product properly generates, conditions, and transmits tone test signals that are pure sine waves by testing each subsystem requirement, as shown in the Requirement and Verification Table in Table 5 of the Appendix. After we integrated each subsystem into our final product, we verified that the speaker transmits these signals as well as a voice test signal that is a human voice speaking from one to five in a manner that is clearly intelligible, directional, and equalized by testing each high-level requirement. Thus, we verified overall functionality, even as we observed that the speaker's output is quiet, likely due to the limitation of our power supply.

3.1 Signal Generation

Regarding our carrier wave signal generation using the ESP32 interfacing with the AD9833, we verified on the oscilloscope that the output is a clean 40 kHz sine wave with a variance of less than 100 Hz. Additionally, we used the FFT tool on the oscilloscope, and as seen in Figure 9 below we found that while there was a slight secondary peak at 80 kHz, there was a clear and dominant peak at 40 kHz, indicating that no harmonics dominate the signal. Regarding our audio wave signal generation using the ESP32 interfacing with the AD5761 DAC, we confirmed on the oscilloscope that this output wave matches tone test signal frequencies from 300-1250 Hz sine waves. Also, in integration with our remaining subsystems, using the voice test case, we found that the audio could be comprehended, indicating a faithful digital-to-analog conversion.



f(t) = FFT(Ch1)
FFT Sample Rate = 200kSa/s
Window Hanning
Span 100kHz
Center 50.0kHz
Scale 20dB/
Offset -82.0dBV

Figure 9 Post-amplification FFT of carrier wave generation system.

3.2 Signal Conditioning

Regarding the filtering of our audio signal output from the DAC, we observed on the oscilloscope that with a 1 kHz sine wave test case after the low-pass filter, the signal's discrete voltage transitions were smoothed out in a linear way, and no tapering effects were observed. This ensured the most accurate analog representation of the originally digital audio signal, minimizing artifacts of conversion. We achieved a gain of 15 of the audio signal prior to modulation with the TL072 operation amplifier, which accounted for the built-in divisor of 10 V in the AD633 transfer function. Also, we achieved a gain of 8.5 of the carrier signal prior to modulation with a TL072 operation amplifier, which brought it from the initially low amplitude of 300 mV up to 2.5 V amplitude, which proportionally increased our AD633 output amplitude, as seen in Figure 10 below.

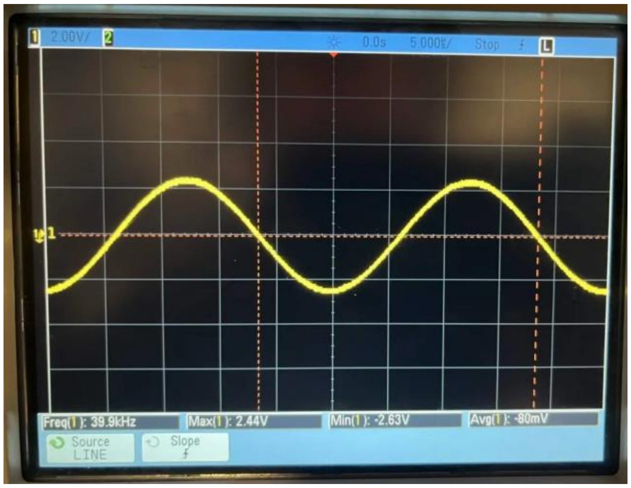
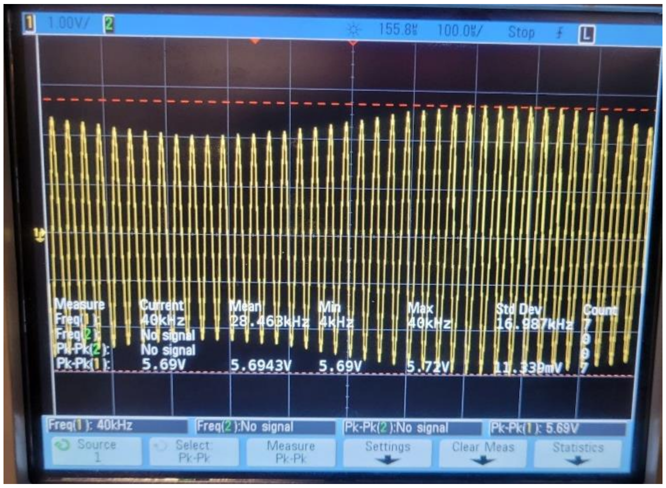


Figure 10 Post-amplification output of the carrier wave generation system.

As shown in Figure 11, we found that after amplitude modulation performed on the AD633, the output signal maintained the carrier signal frequency of 40 kHz and its envelope clearly matched the shape of the 1 kHz sine wave test case. We observed on the oscilloscope that the modulation output signal had no DC offset, which allows the maximal use of the available rail voltage given a roughly symmetric input signal.

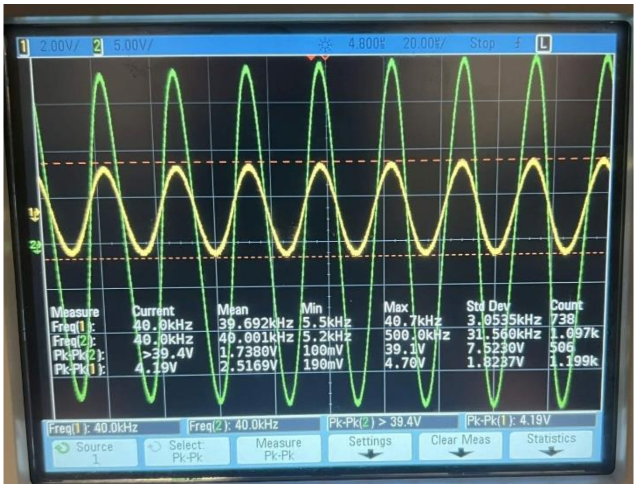


Measure	Current	Mean	Min	Max	Std Dev	Count
Freq(1):	40kHz	28.463kHz	4kHz	40kHz	16.987kHz	7
Freq(2):	No signal					0
Pk-Pk(2):	No signal					0
Pk-Pk(1):	5.69V	5.6943V	5.69V	5.72V	11.339mV	7

Figure 11 AD633 output.

Regarding our band-pass filter, we tested three cases. A 20 kHz sine wave's amplitude decreased by 20.66% and a 100 kHz sine wave's amplitude decreased by 35.12%, whereas a 39 kHz sine wave's amplitude only decreased by 3.9%, confirming that frequencies substantially below and above our 40 kHz resonance frequency were de-amplified. It is important for high-frequency and low-frequency noise to be de-amplified to minimize distortions, optimizing audio quality.

Regarding our post-modulation LM2876T-chip based amplifier, as shown in Figure 12, we achieved an output peak-to-peak voltage of 39.1 V with a 1 kHz sine wave test signal, achieving the amplification necessary for audibility, as this 1 kHz tone was audible after being transmitted by the transducer array. When we tested the voice signal, we found that the maximum amplitude of the signal reached our rail voltage of 22 V, indicating clipping; however, for over 95% of the length of the signal, clipping did not occur and the average amplitude when the voice was speaking was about 20 V. We decided that allowing a small amount of clipping was acceptable, since the clipping only occurs for a small portion of the signal, and de-amplifying would result in a quieter signal – after halving the LM2876T's feedback resistor value, effectively halving the gain, we could no longer hear the signal, proving the positive relationship between amplification and volume. Thus, we conclude that we achieved an optimal balance of minimal distortion and maximal volume given our rail voltage limitation of ± 22 V.



Measure	Current	Mean	Min	Max	Std Dev	Count
Freq(1):	40.0kHz	39.692kHz	5.5kHz	40.7kHz	3.0535kHz	738
Freq(2):	40.0kHz	40.001kHz	5.2kHz	500.0kHz	31.560kHz	1.097k
Pk-Pk(2):	>39.4V	1.7380V	100mV	39.1V	7.5230V	506
Pk-Pk(1):	4.19V	2.5169V	190mV	4.70V	1.8237V	1.199k

Figure 12 Pre-amplification (channel 1) and post-amplification (channel 2) signals

3.3 Signal Transmission

We found that our transducer array received the conditioned signal and was able to transmit it, with the impedance matching being important. Without the two series 220 µH inductors in parallel with the transducer array, we observed that the capacitive transducers appeared to charge and discharge, preventing the signal from being steadily transmitted—our transducer array with impedance matching reliably transmitted the ultrasonic 40 kHz signal, demonstrating power efficiency. Below we outline how we verified that our speaker meets overall functionality in terms of high-level requirements under the categories of directionality, equalization, and demodulation.

3.3.1 Directionality

The first high-level requirement is that our speaker maintains a half-power beamwidth of less than 10 degrees at a distance of 1 meter, which is a benchmark for attaining directionality. With a test 1 kHz sine wave as the input signal, we measured the audible frequency range sound pressure level with a decibel meter a meter away from the center of the transducer array. As shown in Table 1, at a perfectly centered angle, we measured the sound pressure level to be 81.8 dB, whereas at angles of 5 and -5 degrees, we found the sound pressure level to be 70.2 dB and 70.6 dB, respectively. A half power-beamwidth is defined as a 50% drop or a drop of 3 dB in both directions, and we find a sound pressure level drop of over 11 dB in both directions, indicating the half-power beamwidth is less than 10 degrees. At angles of much greater than 5 degrees and -5 degrees, we find the sound pressure level to only drop slightly, namely less than 0.2 dB, indicating that there is a sharply defined sound pressure level threshold that is within 5 degrees in one direction. These measurements align with our qualitative observations – upon moving out of focus from the speaker, we cannot hear the voice speaking in the voice test case, whereas in focus we clearly can.

Table 1 Audible Frequency Range Sound Pressure Level at Various Angles

Degrees (°)	**Sound Pressure Level (dB)**
<< -5°	70.5
-5°	70.6
0°	81.8
5°	70.2
>> 5°	70.0

3.3.2 Equalization

The second high-level requirement is that our audio output has a frequency response flatness of at most ± 6 dB, which is a benchmark for attaining equalization. We measured the audible frequency range sound pressure level with a decibel meter a meter away from the center of the transducer array at an angle of 0 degrees, testing a sweep of sine wave test case frequencies between 300 to 1250 Hz. As shown in Table 2, we find there is a positive and linear correlation between frequency and sound pressure level, with the highest frequency of 1250 Hz resulting in the highest sound pressure level detected of 83.5 dB and the lowest frequency of 300 Hz resulting in the lowest sound pressure level detected of 82.6 dB. We find that the difference in maximum and minimum sound pressure levels within a range of 300 to 1250 Hz is less than 1 dB, satisfying the requirement of a frequency response flatness of ± 6 dB. These measurements align with our qualitative observations – while higher frequencies are somewhat louder, the difference in volume is no so great as to cause distortion in the voice speaking test case or cause difficulties in hearing low frequencies.

Table 2 Audible Frequency Range Sound Pressure Level at Various Frequencies

Frequency (Hz)	Sound Pressure Level (dB)
1250	83.5
1100	83.5
1000	83.4
900	83.3
750	83.3
650	83.0
500	82.9
400	82.7
300	82.6

3.3.3 Demodulation

The third high-level requirement is that our transducer array outputs an ultrasonic sound pressure level of at least 110 dB, which is a benchmark for achieving demodulation. In order for an ultrasonic directional speaker's signal to demodulate in the air and become audible, it needs to pass this 110 dB threshold [1]. Our decibel meter, a standard variety, was only capable of measuring the audible frequency range, so it could not detect the greater than 110 dB sound pressure level dominated by ultrasonic frequencies. Due to budget constraints, we could not directly measure the ultrasonic signal at a level as high as 110 dB. However, since all audio test signals were audible, we conclude that this has been met. If the threshold was not met, the signal would not have demodulated, and it would not have been audible. We assessed that since the voice test case speaking "one, two, three, four, five" was clearly intelligibly with minimal distortion, the demodulation affect was achieved.

Without a direct measurement, we can estimate the sound pressure level given our peak-to-peak voltage measurement of the fully conditioned signal of 39.1 V for the 1 kHz sine wave test case, which corresponds to a root mean squared voltage of roughly 14 V. According to the CUSA-T60-150-2400-TH transducer data sheet, with a reference voltage of a 10 V sine wave at 30 cm away from a single transducer, the sound pressure level has a minimum value of 115 dB [6]. The sound pressure level can be estimated as the sum of the base level defined on the data sheet, the voltage gain given our measurement, and the array gain provided by our 16 transducers:

$$SPL \approx SPL_0 + 20\log\left(\frac{V}{V_0}\right) + 20\log(N) \approx 115 + 20\log\left(\frac{14}{10}\right) + 20\log(16) \approx 142\ dB \tag{9}$$

We note that the actual ultrasonic sound pressure attained may be less than the estimated 142 dB, as this estimation is an idealized case assuming perfectly optimized spacing between our transducers to achieving the full array gain added to the base level of a single transducer, however it is clear that the sound pressure level is greater than 115 dB, as even without the array gain factor, this is obtained.

4. Costs

In Table 3 below, we document all costs due to parts bought, and in Table 4 below, we document our schedule of work and associated tasks for this project. We then make a calculation of labor costs based on our schedule.

4.1 Parts

Table 3 Parts Costs

Part	Manufacturer	Retail Cost ($)	Bulk Purchase Cost ($)	Actual Cost ($)
AD9833 Dev Board	NOYITO	16.99	N/A	16.99
AD5761	Analog Devices	8.98	5.80	8.98
ESP32 Dev Board	Espressif	15.00	6.33	15.00
AD633	Analog Devices	21.48	14.00	21.48
TL072	Texas Instruments	0.32	0.13	0.00
CUSA-T60-150-2400-TH	Same Sky	5.15	3.45	72.54
RLB9012-221KL	Bourns Inc.	0.62	0.32	1.74
LM2876T	Texas Instruments	3.57	3.57	0.00
0805 SMD Resistor Kit	ATNSINC	8.99	N/A	0.00
0805 SMD Capacitor Assorted	MCIGICM	7.99	N/A	0.00
LRS-50-24	Mean Well	11.40	10.40	0.00
LM7815CT	Texas Instruments	1.86	1.365	0.00
Lm7915CT	Texas Instruments	1.86	1.365	0.00
LM7805CT	Texas Instruments	1.86	1.365	0.00
Total				**136.73**

4.2 Schedule

Table 4 Schedule

Week:	Piotr	Arthur	Sam
2/9	Hours Spent: 12 Task Worked on: Researching the general project to have a better understanding. Researched different types of filters and amplifiers.	Hours Spent: 15 Task Worked on: Researching the general project and possible ways to design modulation, transducer array, and impedance matching.	Hours Spent: 15 Task Worked on: Researching and designing ESP32 system and carrier wave generation.

2/16	Hours Spent: 15 Task Worked on: Picking out amplifier chips, resistors, and capacitors for filter. Research more about amplifier chips and worked on making PCB.	Hours Spent: 20 Task Worked on: Continued research for modulation, transducer array, and impedance matching, simplified overall design to one-beam method.	Hours Spent: 15 Task Worked on: Final research before Project Proposal, making final design block diagram after finalizing to AM circuit, first PCB order out.
2/23	Hours Spent: 15 Task Worked on: Worked on Design Document, finished up PCB order. Looked at in house components to pick up.	Hours Spent: 24 Task Worked on: Made schematic for modulation subsystem, worked on design document, began searching for parts.	Hours Spent: 20 Task Worked on: Design Document, making BOM sheet to order parts.
3/2	Hours Spent: 20 Task Worked on: Design Review with TA, Testing amplifier chips for filter, building and testing the filter. Looked at voltage regulators. Fixed PCB to be sent out in second round order.	Hours Spent: 13 Task Worked on: Searched for parts in supply shop and Digikey, compared different parts, helped send out orders, prepared and went to Design Review.	Hours Spent: 15 Task Worked on: Design Review with TA, setting up Arduino IDE for coding, looking into self-service voltage regulators.
3/9	Hours Spent: 20 Task Worked on: Adjusting bandpass filter, with different components. Starting to work with amplifier subsystem.	Hours Spent: 15 Task Worked on: Did breadboard testing for modulation subsystem.	Hours Spent: 15 Task Worked on: Starting initial code, switching design to have dual power rails, ordering parts.
3/16	Hours Spent: 5 Task Worked on: Researched more about amplifier, built the wooden housing for the project.	Hours Spent: 4 Task Worked on: Researched possible issues with modulation subsystem.	Hours Spent: 10 Task Worked on: AD9833 Code over Spring Break.
3/23	Hours Spent: 25 Task Worked on: Working on amplifier chip, debugging circuit. Incorporating voltage regulators.	Hours Spent: 20 Task Worked on: Debugged issues with modulation subsystem on breadboard.	Hours Spent: 25 Task Worked on: AD9833 debugging of software, adding HPF and amplifier, soldering DAC.
3/30	Hours Spent: 23 Task Worked on: Connecting filter, amplifier,	Hours Spent: 20 Task Worked on: Worked on combining modulation,	Hours Spent: 25 Task Worked on: Start DAC software, make

	and regulators to AM subsystem. Incorporating onto one breadboard. Ordering more components.	filter, and amplification subsystems.	final driver board PCB.
4/6	Hours Spent: 20 Task Worked on: Fixing the single breadboard design after burning the AM chip, Rewiring and debugging circuit.	Hours Spent: 20 Task Worked on: Worked on combining all subsystems, worked on modulation and transducer array subsystem testing.	Hours Spent: 15 Task Worked on: DAC rebuild after burnout, debugging DAC system software and hardware, interface with rest of system.
4/13	Hours Spent: 20 Task Worked on: Soldering PCB with Transducers, Fixing amplifier chip, testing and verifications on the subsystem.	Hours Spent: 20 Task Worked on: Worked on inductor configurations for impedance matching, researched DC power supplies.	Hours Spent: 25 Task Worked on: Debug whole system, improve software to output a tone, look to increase modulation depth.
4/20	Hours Spent: 20 Task Worked on: Preparing for mock demo and presentation. Incorporating the entire system on one breadboard and attaching PCB to the wooden box. Soldering amplifier chip for more optimal wiring.	Hours Spent: 25 Task Worked on: Made final breadboard testing changes, debugged wiring issues, tested different combinations of feedback resistors, acquired and wired DC power supplies, prepared for mock demo and presentation.	Hours Spent: 25 Task Worked on: Debug whole system to improve modulation depth and increase volume, tune system to sweep tones, software to have both tones and speaking audio, prep for mock demo and presentation.
4/27	Hours Spent: 25 Task Worked on: High Level requirement testing, final testing, adding heat sink to amplifier, and making the final presentation and demo document.	Hours Spent: 26 Task Worked on: Worked on high-level requirement testing with decibel meter and prepared for final demo, presentation, and report.	Hours Spent: 25 Task Worked on: Final testing to prove subsystem requirements for final demo and presentation, making final presentation.

4.3 Labor

For costs of labor, with a $43.00 dollar an hour average for Electrical Engineering graduates [7], we worked a total of 672 hours, as summed from Table 4, yielding a total labor cost of $28,896.

5. Conclusion

We conclude by discussing the accomplishments and uncertainties of our project, and we explain how we abided by ethical considerations. With our insights and experience, we make suggestions for future work.

5.1 Accomplishments

This semester we reached several milestones and achieved many successes that future groups can build upon. Most notably, we got an audible output from our speaker, proving that amplitude modulation can yield the non-linear demodulation effect that we desired. This was an accomplishment, since most systems use pulse width modulation to drive the effect, and there is very limited research on the amplitude modulation effect of directional speakers. By proving this effect works and determining that an amplitude of 20 V is merely the threshold for this effect using amplitude modulation, we have gained necessary knowledge for future groups. Additionally, our system met all high level requirements, which means that we achieved beyond a minimal level of functionality.

5.2 Uncertainties

There are a few uncertainties about the effects of our speaker that we could not measure directly. We could not directly confirm the dB level of the ultrasonic output using an SPL meter since we were at a frequency beyond what typical SPL meters can measure. This led to the assumption that we are above the 110dB level because we heard the output, but not by how much we cleared it, leading to some level of uncertainty. Additionally, different sources give different reasons for the directionality of the speakers. Some sources, such as the Polytechnic University of Valencia, state that it is the small wavelength of the ultrasonic waves that put the audible beam in a narrow cone [1]. Other sources, such as Electron Impressions on YouTube, suggest that this phenomenon is from constructive and destructive interference, and the placement of the transducers promotes constructive interference along the front of the wave, but destructive interference along the side [8]. Confirmation on the true origin of the effect would get rid of any design uncertainty.

5.3 Ethical considerations

For the ethics of this project, we need to consider the effects of ultrasonic waves on individuals. On the effects of 40 kHz waves on individuals, a study published in *Applied Acoustics* was conducted to see the effect of 40kHz wave on cognitive function [9]. The results of the study showed that there was no correlation between cognitive function and the ultrasonic waves when the SPL dB level was at or below 120dB. A similar study was conducted by the same researchers in 2019 for the *International Congress on Acoustics* [10]. This study, also capped at 120dB, confirmed that there was no correlation between TTS, which is a temporary loss in hearing, and 40kHz ultrasonic frequencies. Given there is no relationship between cognitive function nor temporary or permanent hearing loss at the dB levels our system operates at, the system we have designed should be safe.

5.4 Future work

We have several suggestions for future work based on our experience designing and constructing this project. First, our rail voltage limitation resulted in us not driving the transducers at their maximum rated voltage. Increasing the bounds of the output voltage will likely increase the output volume; it will also increase the modulation depth, given additional room for voltage to move and encode an audio wave. We recommend those doing future work to design their own AC to DC conversion system that plugs into the wall and converts the grid voltage into the highest DC voltage possible with their transducers and amplifier.

Second, we observed that our speaker created patches of sound directionally as opposed to a unified front of sound, likely due to the transducer placement, where the transducers left large lines of space unoccupied and thereby without sound. We recommend those doing future work to study and test the optimized distance between transducers that best promotes constructive interference at the front of the wave and destructive interference on the sides of the speakers.

Additionally, while this would considerably change the project design, it may be sensible to use pulse-width modulation as opposed to amplitude modulation. Such a design would likely be simpler, cheaper, and more power efficient, saving the budget for more transducers, which allows for a greater sound output, and eliminating the need for very high DC voltages. Such a design may entail H-Bridge amplifiers and a single 24 V voltage source. A possible downside to pulse-width modulation is worsened audio quality; since using pulse-width modulation would entail a complete redesign of the system, it may be difficult to compare methodologies with our project.

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Appendix A Requirement and Verification Table

Table 5 System Requirements and Verifications

Requirement		Verification status (Y or N)
1. Waveform Generator Requirements a. The wave has a high resolution without any noticeable gaps or quantization. b. The created wave from the waveform generator will not create large amounts of harmonics in the output signal.	1. Verifications a. An oscilloscope will be able to read a frequency of 39.25kHz within a tolerance of .05 kHz using the frequency and the built in waveform analysis tools. b. Use the FFT tool on the oscilloscope to determine that the frequency breakdown is primarily 39.25kHz and does not have peaks within 10% of the value at 39.25kHz.	Y
2. DAC Requirements a. The output wave audio can be comprehended upon output. b. The output waves do not have any artifacts from the conversion. c. The conversion of the wave at any given point is within a .01V range through multiple operations.	2. Verifications a. We will use an analog speaker and a simple op-amp to play the converted analog signal. b. We will use an oscilloscope to look at the reconstructed audio signal to make sure the reconstructed wave is smooth and without unexpected spikes. c. We will use the oscilloscope to record the conversion process multiple times and make sure the wave form is consistent across multiple trials.	Y
3. Bandpass Filter Requirements a. It filters out noise and frequencies outside of the range. b. It amplifies the signal if needed. c. It keeps the signal centered around a specific frequency.	3. Verifications a. It sends various signals through the filter should be successfully filtered out. Frequencies that can be tested are the middle of the bandwidth, still in the bandwidth on the edge, outside the range on the edge, and a signal far outside of the bandwidth. For example: 39.25kHz, 37.3kHz, 36.75kHz,20kHz. b. If amplification is necessary, we will be sending in a signal at different frequencies and peak to peak voltages, checking that the output is	Y

	acting accordingly; if the gain needed is 2, then adjusting the resistors to have the gain of 2 and seeing the output on the oscilloscope will be done. c. Check the output of the filter on an oscilloscope to see if the signal is still properly centered and if it is relatively close to 39.25kHz.	
4. Audio Amplifier a. It amplifies the signal to the necessary dB to hear the signal through the transducer. b. The amplifier can run without overheating and possibly melting. c. The amplifier does not cause too much noise and distort the signal.	4. Verifications a. Display the output signal on the Oscilloscope and see the gain of the amplitude. Once the output of the signal is obtained, the dB can be calculated to see if the necessary dB is obtained. b. Running the circuit for a set given time such as a 1 minute, so that the audio can be heard clearly multiple times, will be done, and during this time checking the state of the amplifier to make sure it does not overheat will also be done. c. The output of the signal keeps its initial shape (relatively close); there is not an excess amount of noise.	Y
5. Modulation Circuit a. The modulation index must be between 0.5 and 1.0. b. The output DC offset should not exceed ±50 mV after AC coupling. c. The AD633 must maintain the carrier frequency at 39.25 kHz (±0.5%).	5. Verifications a. Connect the AD633 output to an oscilloscope. Measure peak carrier amplitude and modulating envelope. Verify if the ratio is between 0.5 and 1.0. b. Measure the signal at the X1input of the AD633 using a multimeter in DC mode. Verify the 1 µF capacitor has removed any offset. c. Measure the output of the AD633 with an oscilloscope.	Y
6. Transducer Array and Impedance Matching a. Resonant frequency of the LC circuit must be 39.25 kHz (±1%) b. Total series inductance (L) must measure 428 µH (±5%).	6. Verifications a. Use an oscilloscope to find the frequency where the voltage across the transducer array is at its maximum peak-to-peak value. b. Use a meter to measure the combined value of the inductor(s) before soldering to ensure it/they match the calculated resonance requirement.	Y