LABESCAPE ULTRASONIC DIRECTIONAL SPEAKER

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Contents

1. Introduction

1.1 Problem and Solution

For the LabEscape escape room, the primary objective is to hide not only the clues necessary to escape, but also science topics through the medium of an escape room. For future escape rooms, there is a desire to hide clues via audio through a directional speaker. This would engage participants in the science of waves, how wavelength is inversely related to frequency, and the fundamental equations that govern waves. A directional speaker is necessary because unlike a traditional speaker, it does not fill the entirety of the space it is in due to its long wavelength. By using ultrasonic waves, we can hide the audio into a clue so a participant must be in a very specific spot of the room to hear the clue. The original pitched idea for this would require two speakers to each have a section of a code, and only at the intersection of the speakers could the entire code be heard. However, for the purposes of this design in this course, we will be designing a system for one of these speakers that can then be easily expanded for the purposes of LabEscape by doubling the analog circuitry. Our overall solution is depicted in the block diagram shown in Figure 2.

1.2 Visual Aid

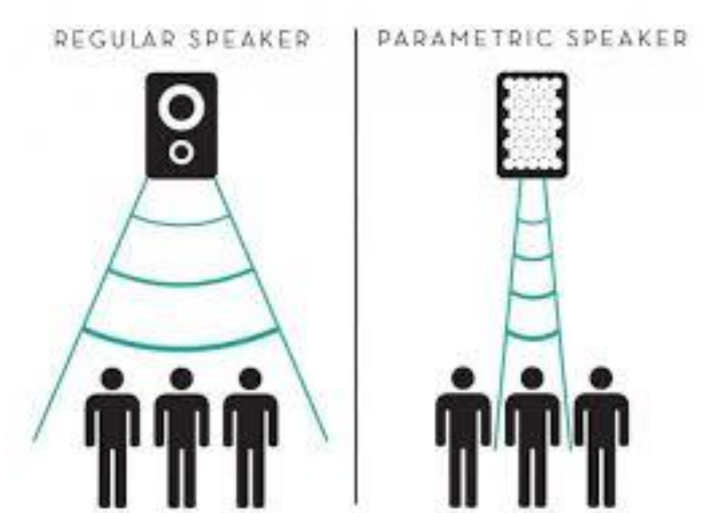

Figure 1: This figure shows the directionality of the directional (parametric) speaker over a traditional speaker.

1.3 High-level Requirements List

If all the following high-level requirements are met, then an ultrasonic directional speaker has been successfully implemented, meaning that only in a narrow distribution can audio be heard in front of the speaker, and a voice recording can be comprehended.

1. The system shall maintain a half-power beamwidth of less than 10 degrees at a distance of 1 meter. According to research from the Polytechnic University of Valencia [8, p.2] this beamwidth is modeled as sqrt(4α/k). Given an absorption coefficient α of 0.15 Neper/m for the ~40 kHz carrier and a wave-number k of 22.91 rad/m (calculated for a 1250 Hz signal at the speed of sound in air 343 m/s with the

equation 2 * pi * frequency / speed, the theoretical beamwidth is 9.27 degrees. Our 10-degree target is consistent with the researchers' experimentally confirmed 4-degree beamwidth at 1 meter [8, p.9], while allowing a realistic margin for independent development. To test this requirement, the speaker array will be mounted on a calibrated rotating base at a fixed distance of 1 meter from a stationary SPL meter. After establishing a peak reference at 0 degrees, the array will be rotated until the measured intensity drops by 3 dB. This angle will be recorded for both the clockwise and counterclockwise directions. The sum of these two angles constitutes the measured half-power beamwidth.

2. The system shall maintain an audio output frequency response flatness of ±6 dB within an operating bandwidth of 300 Hz to 1250 Hz, a moderate range. According to the Polytechnic University of Valencia [8, p. 7], the demodulated sound pressure level is proportional to the square of the modulation frequency. This physical relationship creates a positive slope of 12 dB/octave, which would naturally cause higher-pitched clues to be significantly louder than lower-pitched ones [8, p. 7]. To ensure the audio is balanced and intelligible for escape room participants, the system must compensate for this slope via pre-equalization. To verify this, a frequency sweep from 300 Hz to 1250 Hz will be performed in front of the array. Using an SPL meter at 1 meter, the peak-to-peak fluctuation must remain within a 12 dB window to confirm the equalization of the output signal.

3. The transducer array must generate a minimum ultrasonic sound pressure level of 110 dB at the emitter face. If this threshold is not met, the air fails to delinearize, which is necessary for the ultrasonic signal to become audible. Verification will be conducted using a SPL meter placed 1 cm from the center of the 4x4 array. The system will be driven at its carrier frequency of 39.25 kHz. A measurement of 110 dB or higher will confirm that the system has reached the power level necessary to trigger the non-linear propagation.

2 Design

2.1 Block Diagram

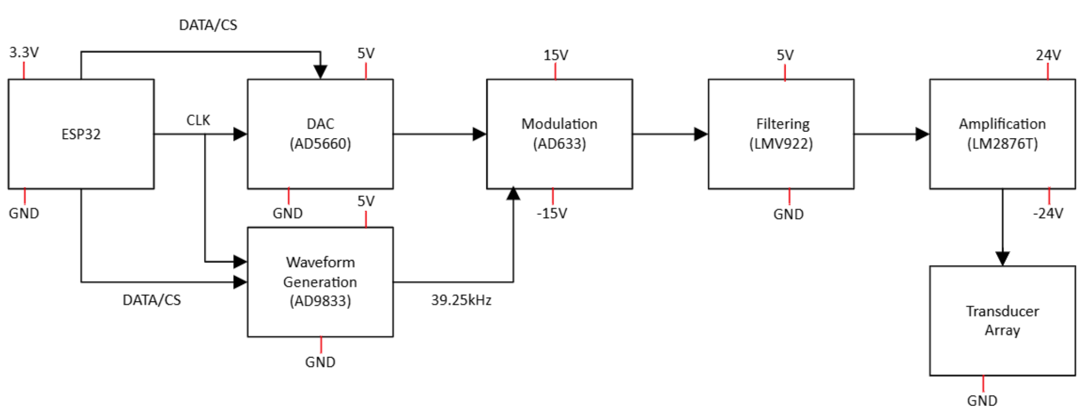

Figure 2: This image depicts our overall block diagram for our proposed solution. Rail voltages are marked with red.

2.2 Physical Design

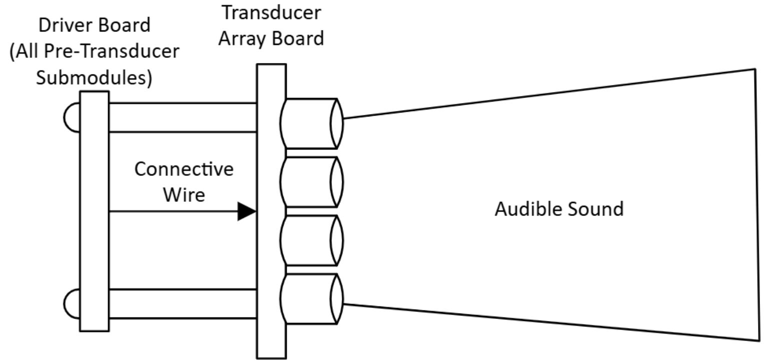

Figure 3: This figure shows our project configuration with 2 separate boards. One will drive the signal. Then, a wire will carry the signal over to the transducer array board, where it will output the modulated signal.

2.3 ESP32 to AD5660 Subsystem

This submodule will take a digitally stored version of an audio clip and output it as an analog signal using the AD5660 16 bit serial load DAC, allowing us to send the desired audio through the rest of the driver circuit and output it through the directional speaker. The ESP32 will also use 3 GPIO pins to connect to the AD5660. The DAC will be powered by 5V with the output from the DAC being some scaled voltage between 0V and 5V. To use the DAC, we will load the ESP32 with a 16 bit encoded audio clip sampled at 22050 Hz. The data will be serially loaded out of the ESP32 memory through a GPIO pin and into the AD5660 through its DIN pin. Then using software, we can time pulses of SCLK and the SYNC pin of the AD5660 such that for every 16 bits loaded an output voltage is released. There will also be coupling capacitors on the VDD pin as described by Figure 4 below from the datasheet [13, p.22].

In regard to the wiring of the ESP32, the chip will be powered by a 3.3V source, which will be achieved via a voltage regulator such as the LP2950CZ from the self-service drawers.

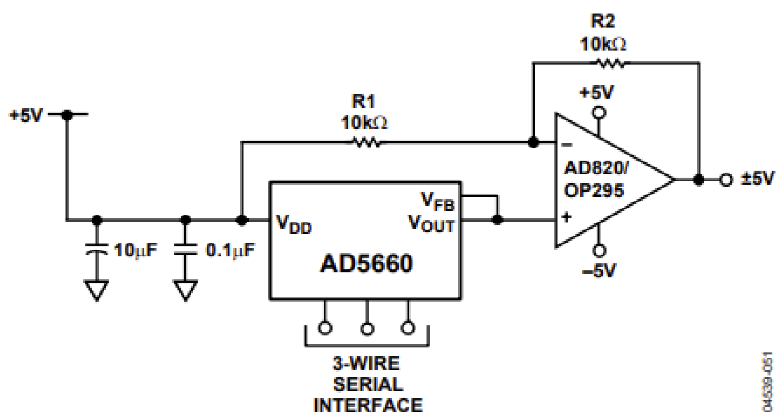

Figure 4: This figure [13] depicts a proposed wiring for the analog signal reconstruction system. We will not have the same post-DAC signal processing.

2.4 ESP32 to AD9833 DD5 Programmable Waveform Generator

The ESP32 will use 3 of its GPIO pins to connect directly to the AD9833 programmable waveform generator chip to create a 39.25kHz sine wave. Note that 40kHz is the standard central frequency for the transducers we have, but to even out the decibel level of the sound that we are expecting, we will shift this frequency to 39.25kHz. This sine wave will be used as the carrier wave input to the modulation subcircuit to allow for our analog signal to operate at the center frequency of the transducers, making our analog signal compatible with our transducer array. The chip will be powered by a 5V supply, and the connections to the GPIO pins will be at the SDATA, SCLK, and FSYNC pins [14, p.6] of the board. Both digital and analog ground will be grounded. Software will provide SDATA with a sequence of inputs that will program the AD9833. Once the AD9833 is programmed, we should be able to put the FSYNC pin low since the input data will no longer be changing, and the waveform will hold its shape throughout the program loop.

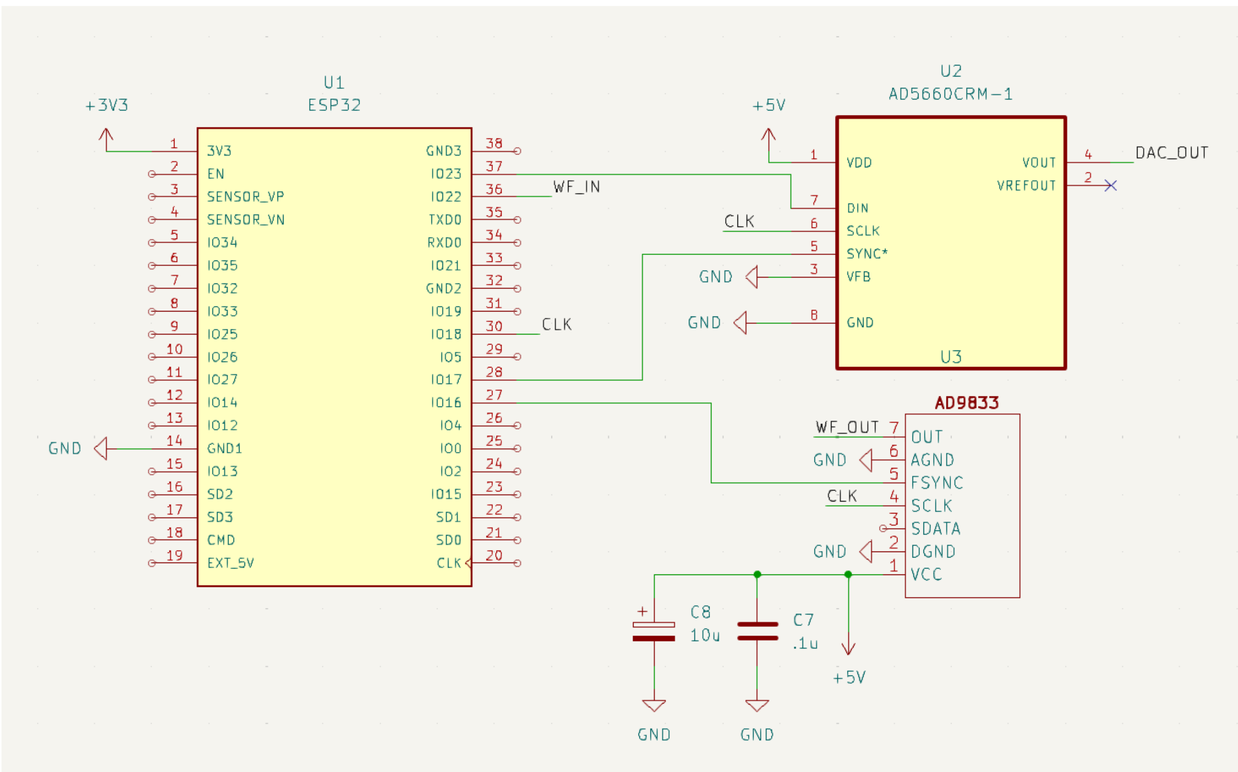

Figure 5: The schematic for the ESP32 connections for both the waveform generation and the DAC signal reconstruction.

2.5 Filters Subsystem

According to the Polytechnic University of Valencia [8, p.3], the demodulated pressure is proportional to the square of the modulating frequency, and it is necessary to equalize the audio signal before modulation. However, we believe that signal resolution may be affected by any pre-modulation filtering. We will test the DAC output on an oscilloscope to experimentally determine if any pre-modulation filtering is needed. Thus, we focus on the post-modulation filtering in this report.

The bandpass filter will be used as an intermediate step taking in the output of the AM circuit and outputting a filter signal into the audio amplifier. An indicator of success is that the filter will successfully reduce noise and eliminate any harmonics outside of the bandwidth. The active bandpass filter was chosen as it was deemed important to keep the frequency in a specific bandwidth and possibly amplify the signal if needed before it being sent into the audio amplifier. The filter is important in making sure that the signal is as clean as possible.

For the bandpass filter, we are looking to do an active bandpass filter with a bandpass of 37.25kHz-41.25kHz. We are using this initial range to make sure that none of the signals are lost from the filter. From Medical Instrumentation: Application and Design [15], the following equations were used to find the resistor and capacitor values for the filters: $f = \frac{1}{2\pi RC}$. We will use the calculated values from the formula and the closest available components in self-service and the supply shop. The following resistor and capacitor values were chosen. C = 1nF, R_{HP}= 4.3kΩ and R_{LP}= 3.9kΩ. The bounds of the bandpass may not be exactly 37.25kHz- 41.25kHz. However, they will be the closest they can be given the situation. With the active bandpass, a gain can be applied to the signal as well; for the initial gain it will be 2, so a third resistor is added with the value of 8.2kΩ, as it is the closest available value in the supply shop. The gain will be adjusted as necessary; this will be dependent on the signal coming from the AM circuit. For the op amp in the filter, the LMV922MX was chosen as it has low noise, minimal distortion, and a good Gain Bandwidth Product. The rail voltage being supplied to the LMV922MX would be 5V. The bandpass filter components may be changed if the bandpass or gain need to be increased or decreased respectively.

Figure 6 shows the schematic of the bandpass filter. The concept was taken from the Medical Instrumentation: Application and Design manual [15] and course work in ECE 415. The input signal first goes through the high-pass filter, which contains the resistors and op-amp; then it goes through the low-pass portion of the bandpass filter, outputting the filtered signal.

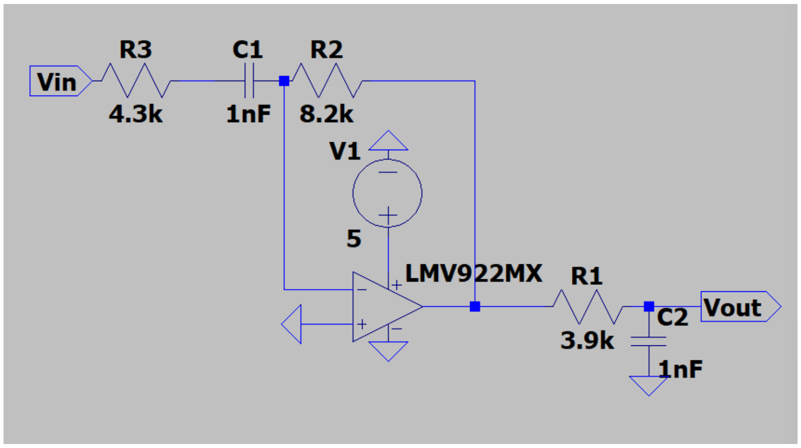

Figure 6: Bandpass Filter

2.6 Amplifier Subsystem

The amplifier subsystem will be an audio amplifier chip that takes in the output of the filter and goes through amplification. The amplified signal will then be sent into the transducer array, with the output being audible noise. An indicator of a successful amplifier subsystem would be that the amplifier can take a signal at a given frequency and amplitude. Then, we output an amplified signal at the necessary amplitude without destroying the frequency of signal or adding too much noise.

For the audio amplifier, the LM2876T was chosen as it works in the required frequency range; it is accessible from the self-supply center, and it has the necessary amount of rail voltage to supply the necessary dB to be audible enough. $\pm$ 24V is chosen as the supply voltage, as it should be large enough for proper amplification. In Figure 7, the schematic of the amplifier subsystem is shown, currently with unassigned resistor and capacitor values as representation for the amplification, this will change as testing occurs and necessary components will be added or changed that affect the overall gain of the amplifier.

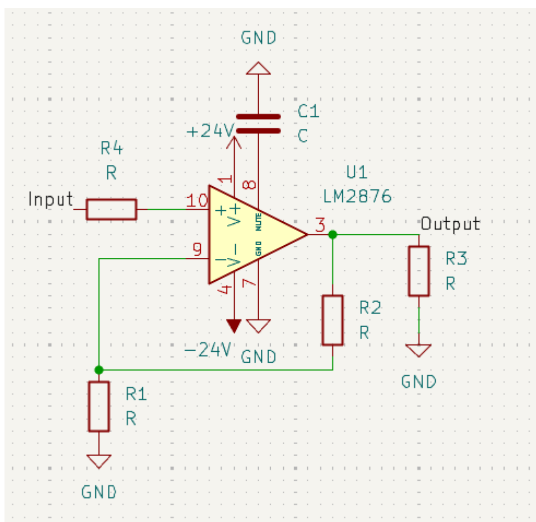

Figure 7: Amplifier Circuit (Simplified)

2.7 Modulation Subsystem

We will be performing amplitude modulation, with the carrier signal being the 39.25 kHz sine wave, and the modulating signal being the filtered audio output from the digital-to-analog converter. According to the International Research Journal of Engineering and Technology (IRJET) [7, p.2] and the Polytechnic University of Valencia [8, p.2], amplitude modulation is suitable for ultrasound directive speakers using a transducer array.

To perform amplitude modulation on the audio signal, we have designed the schematic shown in Figure X. The core of this subsystem is the AD633ARZ-R7, a four-quadrant analog multiplier. We chose the AD633 because it has a relatively low cost and it can perform amplitude modulation. According to the AD633 datasheet [1, pp.5,10], it can be used as a linear amplitude modulator with the transfer function W = ((X1 − X2)(Y1 − Y2) / 10V) + Z. Grounding the inverting inputs by setting X2 = Y2 = 0, and substituting

X1 = Audio, Y1 = Carrier, and Z = Carrier, yields W = (Audio × Carrier / 10V) + Carrier, which factors to W = (1 + (Audio / 10V)) × Carrier, where Carrier = A_c × sin(2π × 39.25 kHz × t).

However, the audio and carrier signals cannot be connected directly to the AD633 pins without additional conditioning, as described below.

Since the carrier signal is centered around 0 V, the audio signal must be as well for proper modulation. The audio signal from the digital-to-analog converter may have a DC offset. According to a Keysight article [2], placing a capacitor in series with a signal path eliminates the DC component — a process known as AC coupling — and a capacitance in the µF range is recommended for signals in the low-to-medium kHz frequency range. Thus, the audio signal first passes through a 1 µF series capacitor, as shown in Figure 8.

After AC coupling, the audio signal's amplitude may still be too low or too high relative to the carrier. According to Number Analytics [3], the modulation index — defined as the amplitude of the modulating signal divided by the amplitude of the carrier — must be kept between 0 and 1 to prevent overmodulation; we also believe it should be at least 0.5 to ensure a clear, detectable variation in the carrier's amplitude. To achieve this, the audio signal passes through a standard non-inverting amplifier, with its output connected to the X1 pin of the AD633. As shown in Figure 8, the amplifier uses a 1 kΩ input resistor and a 10 kΩ potentiometer as the feedback resistor, giving an adjustable gain of 1 + (R_pot / 1 kΩ), ranging from 1 to 11 V/V. Combined with the ability to adjust the carrier amplitude via the waveform generator, the modulation index can be tuned to fall within the desired range of 0.5 to 1.

According to a Texas Instruments blog post [4], a bias current cancellation resistor equal to the parallel combination of the feedback and input resistors is recommended at the non-inverting input to reduce output offset voltage — a standard practice for non-inverting amplifier designs. As shown in Figure 8, we connect a 1 kΩ resistor between the non-inverting input and ground, since the parallel combination of the feedback and input resistors ranges between 0 and 909 Ω. This value may be adjusted after experimentation if needed.

The carrier signal must be sent to two pins of the AD633, but directly splitting the signal risks loading down the waveform generator and causing a voltage drop. According to the Wikipedia article on buffer amplifiers [5], "the interposed buffer amplifier prevents the second circuit from loading the first circuit unacceptably and interfering with its desired operation." Thus, as shown in Figure 8, the carrier is first sent through a voltage follower, whose output connects to both the Y1 and Z pins. The follower's high input impedance preserves the carrier amplitude at the source, while its low output impedance drives both pins without voltage drop.

Finally, proper power supply connections and decoupling are required. The AD633 datasheet [1, p.3] specifies a typical supply voltage of ±15 V, so the +15 V and −15 V rails are connected to the VS+ and VS− pins, respectively. The TL072 datasheet [6, p.15] likewise specifies ±15 V as the typical supply, so the same rails are connected to the VCC+ and VCC− pins of the operational amplifier. Both the AD633 [1, p.10] and TL072 [6, p.32] datasheets recommend placing a 0.1 µF bypass capacitor between each supply pin and ground to minimize errors from noisy or high-impedance power supplies; these decoupling capacitors are included in the design, as shown in Figure 8. In our PCB layout, we will keep all decoupling capacitors as close as possible to their respective pins, as the datasheets and standard practice recommend.

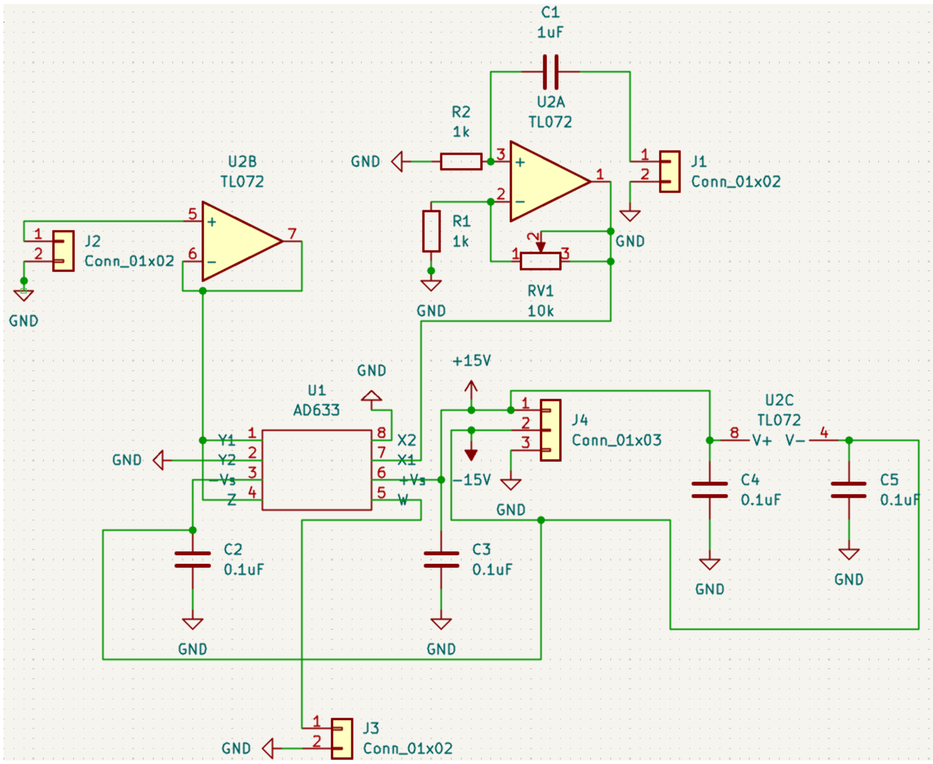

Figure 8: This image depicts the schematic for the modulation subsystem.

2.8 Transducer Arrays and Impedance Matching Subsystem

We have designed a transducer array to transmit the modulated signal. According to the International Research Journal of Engineering and Technology [7] and the Polytechnic University of Valencia [8], an array of transducers must be used to transmit the ultrasonic signal, which will subsequently be demodulated in the air.

Our transducer array is designed to have 16 transducers in parallel, with their positive inputs being connected to the input signal, and their negative inputs being connected to ground, as shown in Figure 9. Each transducer array will be a separate PCB, with its PCB layout being a 4-by-4 square grid, as shown in Figure 10.

According to the International Research Journal of Engineering and Technology [7, p.3], transducers have the property of a capacitor, and a series resonant circuit can be created by placing an inductor in series with the transducer array. According to the CUSA-T60-150-2400-TH datasheet [9], the typical capacitance of a transducer is 2400 pF. Thus, we have decided to place an inductor in series with each transducer array circuit to match impedance. The inductance value is determined by the resonance frequency formula, f = 1 / [2 × pi × sqr(L × C)], where f is the carrier frequency value, 39.25 kHz and C is the overall capacitance, 2400 pF * 16 = 38.4 nF. Solving for L, we obtain L = 1 / [C × (2 × pi × f)^2] ≈ 428 µH. Therefore, for both signal paths, we will use an inductor with a value close to 428 µH or two inductors in series with an equivalent value close to that.

We have chosen the CUSA-T60-150-2400-TH transducer. According to its datasheet [9], it has a directivity of 60 degrees at -6 dB. According to the Polytechnic University of Valencia [8, p.1], demodulation in the air can only occur if the sound pressure level is at least 110 dB. Thus, the minimum sound pressure level for this device is suitable. Compared to other transducers we searched for, it is more expensive and has a larger size, which means that less can fit on the PCB. However, its relatively lower directivity means that the speakers will have a narrower distribution, a key feature in directional speakers, which justifies the higher cost.

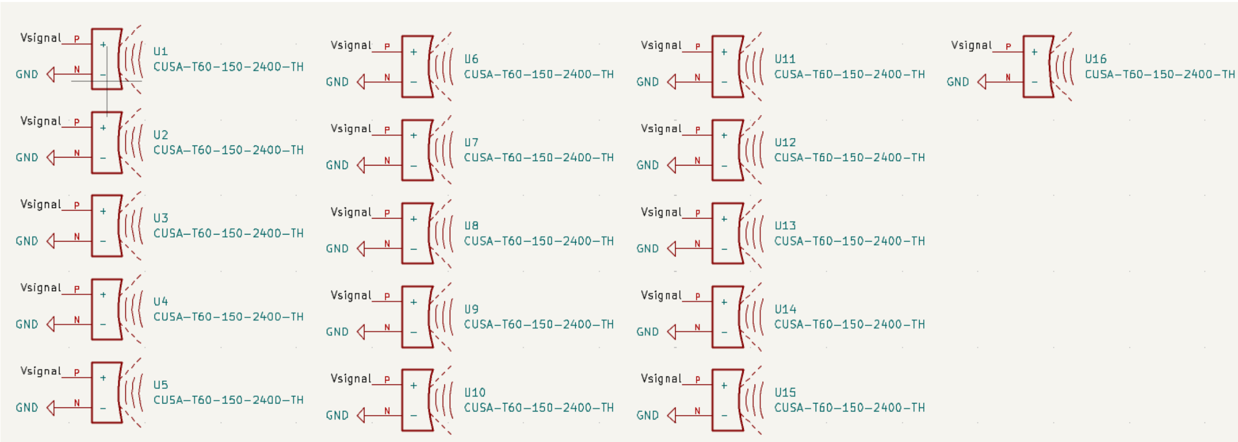

Figure 9: This image depicts the schematic for the Transducer Array.

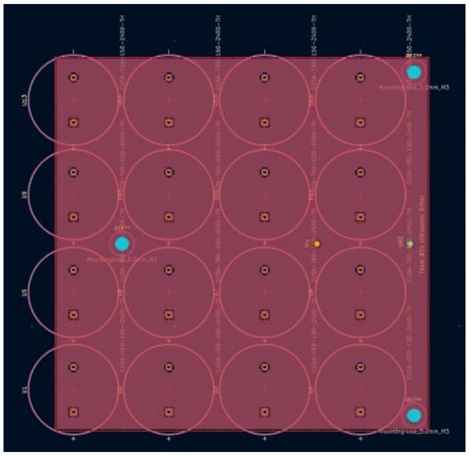

Figure 10: This image depicts the layout for the Transducer Array PCB.

2.9 Tolerance Analysis

Audio quality is important for any speaker. According to Figure 11, frequencies between roughly 38-40.5 kHz have a sound pressure level of at least 110 dB for the CUSA-T60-150-2400-TH transducers. According to Electronics Notes [10], the bandwidth in amplitude modulation is twice the highest frequency of the modulating signal. Thus, we will use a 1250 Hz frequency audio signal and higher if possible. Upon testing a low-pass filter on a voice recording in Audacity with a cutoff frequency of 1250 Hz and the highest roll-off setting of 159 dB per decade, we have determined that while the audio is of low quality, it can be understood and is useable for the purposes of an escape room. We have decided that our carrier signal will have a frequency of 39.25 kHz so that it is centered between the usable range of 38-40.5 kHz.

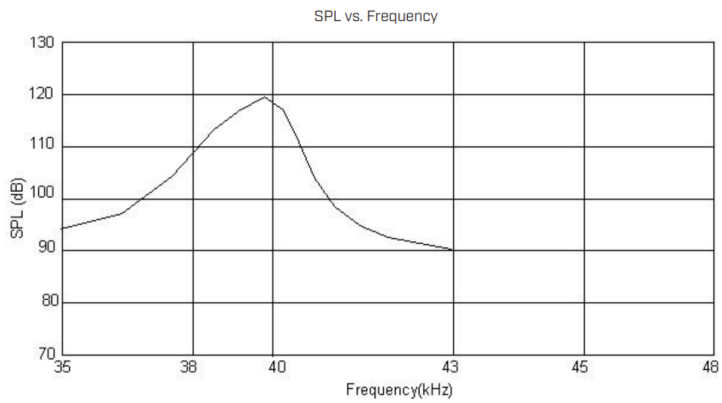

Figure 11: This image depicts the SPL vs. frequency graph for the transducers, according to the CUSA-T60-150-2400-TH datasheet [9].

3. Cost and Schedule

3.1 Cost Analysis

Below is the total cost for the project, excluding components that can be obtained through school, such as circuit components in the Supply Center or Eshop, with the box to contain the electronics being provided through the machine shop and the PCBs also being provided by the school. In Table X, the parts' costs are shown, as the team is still under the $150 budget. The components in the table are crucial for the success of the project as there were no alternatives parts in the e-shop or that could be provided by the school. It is important to note that the parts and total cost are for a singular speaker, as the double speaker method would be over the budget. Each component was cross-checked with the requirements for the project and believed to be the best possible fit for the most reasonable price. Most of the components are being bought for the retail price, as only one is needed. Buying in bulk would put the group over the budget, with many unnecessary extras. The only thing purchased in bulk were the transducers, as 20 were needed (16 for the board and 4 extras), which was a bulk order.

Part	Manufacturer	Retail Cost ($)	Bulk Purchase Cost ($)	Actual Cost ($)
AD9833 Dev Board	Amazon: NOYITO	10.99	N/A	10.99
AD5660 16 bit DAC	Digikey: Analog Devices	12.36	9.71 (when buying 10)	12.36
AD633ARZ-R7 (for PCB)	Digikey: Analog Devices	15.97	12.65 (when buying 10)	15.97
AD633ANZ (for breadboard)	Digikey: Analog Devices	20.85	16.66 (when buying 10)	20.85
CUSA-T60-150-2400-TH (transducer)	Digikey: Same Sky	4.69	3.18 (when buying 20)	80.6
Total				140.77

3.2 Schedule

The plan going forward is as follows:

- Week 3/2: The team will meet for design review. At the review, we will discuss any necessary changes and important deadlines moving forward. We communicate with the TA regarding ordering components and picking up any parts that are already in the building (preferably at the beginning of week). If subsystems have all necessary components, we begin building and testing circuits. It is important to note any results and discuss with the team steps moving forward or any roadblocks. We begin the PCB design for the third round order.
- Week 3/9: We have a completed PCB design in time for the third round order, provided that subsystems are working. The team meets for the breadboard demo and addresses any necessary fixes. We give final dimensions to the machine shop. Every component is working together in the design, and the design works, and we debug if necessary. We perform the team Evaluation.
- Week 3/23: We meet to make sure every subsystem is working together properly (ideally the system should be working by the week of 3/9). We perform last minute fixes if needed. If any fixes are needed for the design, we address and solve them at the beginning of the week, and we have updated the PCB ready for the Fourth Round order.
- Week 3/30: Individual progress reports are due, and we work on putting the PCBs and full system together.
- Week 4/6: The progress demo is done, which is ideally a finished product at this point. The team contract assessment is due.
- Week 4/13: We perform final touch ups and the mock demo.
- Week 4/27: We perform the final demo and give the presentation.
- Week 5/4: Final papers and the Lab Notebook are submitted.

4. Discussion of Societal Impact, Engineering Standards, Ethics, and Safety Considerations

Our ultrasonic directional speaker project makes a positive contribution to public welfare by reducing noise pollution in cultural centers and public spaces. The ability to hear audio only in a narrow range in front of a speaker is the key benefit. In spaces such as museums, crosswalks, or hospitals, directional speakers can deliver targeted information without contributing to ambient noise. Another example of an application of this technology is for escape rooms, as we are developing the project for the purpose of the LabEscape escape room, as it was pitched by Professor Kwiat. By being able to hide a clue in an escape room, a key feature for this cultural and educational site is enabled. Another potential positive contribution to society is the ability to demonstrate the physics of demodulation to students. Demonstrations of physics tend to engage students and encourage learning.

Our project is committed to the IEEE Code of Ethics and the ACM Code of Ethics, which require engineers to prioritize public safety and avoid causing harm [11, 12]. We address these ethics by limiting the device's output power to prevent hearing damage and establishing strict rules for use, such as maintaining a minimum safe distance from the speaker. Following ACM Section 1.3 [11], we are committed to being transparent about potential risks, such as high-intensity ultrasound, by disclosing all findings during testing. Furthermore, our team adheres to ACM 1.5 [11] and IEEE Section II [12] by fostering a professional environment based on respect and non-discrimination. To prevent misuse of the technology, we will ensure supervised operation and take responsibility for the device's impact on society.

The engineering standards we will be applying are IEEE standards. The standards that will apply for this project are as follows:

IEEE 269 [18] – Deals with measurement of audio, both digital and analog, which we will need to measure for our dB level output.

IEEE 830 [19] – Deals with software requirements for structuring and terminology.

IEEE 1100 [20] – Deals with powering and grounding practices for electronics.

The primary electrical concern for our directional speaker is thermal management. The 16-transducer array draws significant current, which can cause the driver or power stage to overheat if not properly cooled. Mechanically, we must address the sharp edges of the PCB and ensure the array is mounted securely. Another safety concern is the high sound pressure level (SPL) of 110 dB required for the air to demodulate the sound. While this is necessary for the technology to work, it presents a risk of ear fatigue or discomfort if a user is positioned too close to the emitter face for an extended period.

To mitigate these safety risks, we will implement a button-triggered operation that enables the speaker for a short period of time, preventing accidental overexposure or long-term overheating. Our Lab Safety Protocol includes: 1) Acoustic Distancing: Requiring all participants to remain at least 1 meter away during operation; 2) Thermal Monitoring: performing periodic temperature checks on the driver chips; and 3) Documentation: Creating a detailed hazard log to remain transparent about any risks discovered during the demo. These design decisions, combined with supervised testing, ensure that both the developers and the public are protected from unsafe conditions while enjoying the educational benefits of the project.

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Appendix A: Requirement and Verification Table

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.	
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.	
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	

	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. Check if the output signal 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) and there is not an excess amount of noise.	
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 AD9833 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 theAD9833 with an oscilloscope	
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	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)	

(±5%).		before soldering to ensure it/they match the calculated resonance requirement.	