LABESCAPE ULTRASONIC DIRECTIONAL SPEAKER

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Project Proposal for ECE 445, Senior Design, Spring 2026

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13 February 2026

Project No. 34

Contents

1. Introduction

Traditional speakers spread in all directions, regardless of where it is pointed. However, there are many applications when a directional speaker is needed. For example, a museum exhibit may need a guest to only hear information from an audio track on a painting directly in front of them. If there are audio tracks in front of every painting, there would be “audio bleed” and it would be difficult to hear. Directional speakers are important for such cultural institutions as museums. Another application of the directional speaker is in escape rooms. A participant may need to reach the point in a room where “X marks the spot”, and only there can the participant hear an audio-based clue. We are proposing an ultrasonic directional speaker for the LabEscape escape room, working on this project pitched by Professor Kwiat in contact with him.

Our proposed overall solution leverages non-linear acoustics to create a parametric acoustic array. Instead of directly transmitting audible waves, we use ultrasonic waves (roughly 40 kHz), which have very short wavelengths, which can be focused into highly directional, laser-like beams using transducer arrays. When these high-pressure ultrasonic waves travel through the air, the medium itself acts as a natural demodulator. By transmitting two slightly different ultrasonic signals, a carrier $f(t)$ and a modulated signal $f(t) + \delta f(t)$, the air molecules perform a mathematical subtraction, leaving the beat note at $\delta f(t)$, which is an audible clue for the escape room. The system is proposed to be implemented using two physically separate transmitter paths, a reference path that emits the 40 kHz carrier wave and a separate path that emits the 40 kHz carrier signal that has been amplitude-modulated (AM) with the audio signal. The paths will be emitted by respective transducer arrays, and a localized audio zone in the air at the intersection point will be created, a shown in Figure 1.

Three high-level requirements to prove the overall functionality of this project are the following. First, the audio zone created is narrow for the purpose of the escape room – the specific size of the audio zone is not determined yet and needs to be calculated. Second, the system should be able to run the entire duration of the audio clue at least three times (in case it needs to be replaced) without overheating and be able to be safely turned off. Third, the audio needs to have a high resolution at the final stage – the specific resolution has not been determined yet.

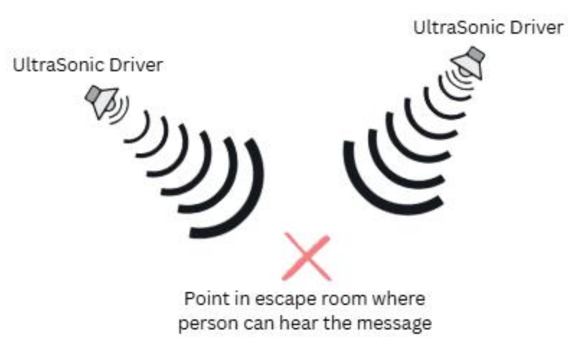

Figure 1 This image depicts our proposed solution in context. Two ultrasonic drivers meet at an intersection point in the escape room, where the audio clue message can be heard.

2. Design

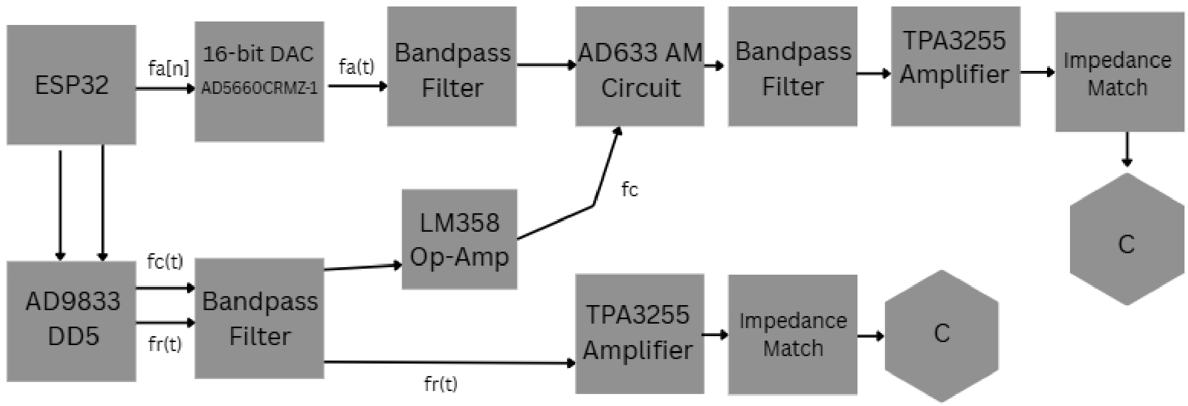

Figure 2 This image depicts our overall block diagram for our proposed solution. C denotes a transducer array.

2.1 ESP32 and DAC Subsystems

First, we focus on the function generator model from the ESP32. This system will take GPIO outputs from the ESP32 and use an onboard 12.5MHz clock signal on a AD9833 Module Board to output a 40kHz sine waveform. This output will then be amplified for testing purposes so that any artifacts of the creation of the wave such as noise can be determined, and filters can be applied as necessary. There will be two of these modules, with one being used directly as an output for one transducer array, and the other used as a carrier wave input for an amplitude modulation circuit. The requirements for this subsystem will be that through software, we are able to output a variety of frequencies of sinusoidal waves at a high enough resolution, without effects coming from harmonics which we can filter out upon looking at the wave output on the oscilloscope. The components used are the AD9833 Serial Interface Module (waveform generator), ESP32, LM358 Op Amp, and resistors (with the values to be determined to set the gain).

Next, we focus on the audio reconstruction module from the ESP32. This system will take GPIO outputs from the ESP32 along with a series output of a 16 bit encoded audio stored on the ESP32 memory to reconstruct an audio signal of myself speaking a numerical code. This will be done through the AD5660 16 bit serial load DAC to provide maximum precision in our output of the reconstructed audio signal. The requirements for this system would be that we are able to play this reconstructed and amplified signal through a speaker and hear the original audio without distortion or noise. Using an oscilloscope, we will be able to determine what frequencies we can filter out. The components used are the ESP32, the AD5660CRMZ-1 16 bit Serial Load DAC, the LM358 Op Amp, and resistors (with the values to be determined to set the gain).

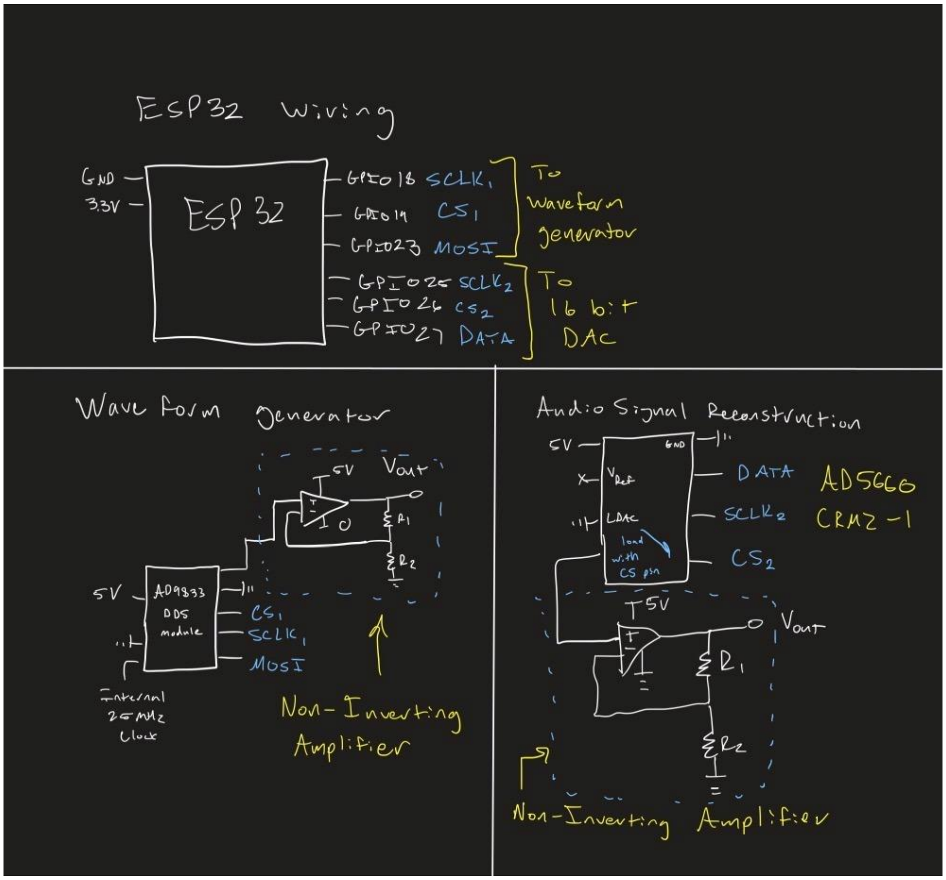

Figure 3. Block Diagrams of the ESP and its Audio Reconstruction and Function Generator Modules

2.2 Filters Subsystem

For the filter sections, we plan to use a bandpass filter to filter out any unnecessary noise, harmonics, and anything else that can possibly affect the signal. In the system there are 3 filter subsystems: the first taking in the output of the DAC and then acting as the input of the AM Circuit and the next having the AM circuit as the input and then outputting the signal into the amplifying chip, TPA3255. Finally, the third filter would take the waveform generator and then output the filtered signal into an op-amp which leads to AM Circuit and outputs the same filtered signal to another TPA3255. These connections can be shown in the overall layout of the system for more clarity. We chose to model the bandpass filter in LTspice as shown in Figure 4.

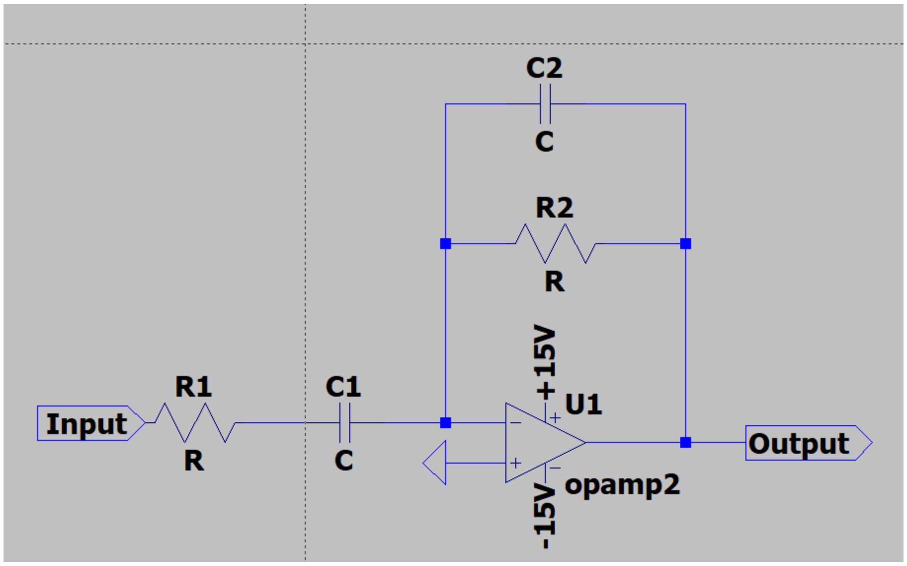

Figure 4. Bandpass Filter Circuit

As shown in Figure 4, the filter consists of 2 resistors, 2 capacitors, and an op-amp. For the values of the resistors and capacitors, we will need to do testing to see what range for the bandpass will allow for the best filtering out of noise and harmonics while keeping the necessary signal in the best possible condition. For the amplifier, we are currently deciding between the LM4562 as it is a good audio filter, and the OPA1656 as it is very good with low noise and low distortion. Both will be used to test and see what fits better with the system. An indicator of success for this subsystem would be successfully filtering out noise and harmonic distortion from its inputs, sending a filter input into the following subsystem. One potential risk of the filter is making sure enough voltage is provided into the system for the op-amp to function properly.

2.3 Amplifier Subsystem

The audio amplifier we chose to use is the TPA3255, as it was the best audio amplifier that could be found. In the system, it is used twice for two different parts, both times taking inputs from filters and then outputting into the audio transducers. Its purpose is to amplify the signal before it gets put into the transducer, so the audio is loud enough to hear and there is a proper message being outputted. An indicator of success for the amplifier is for the signal to be amplified to a set amplitude and kept around 40kHz before it is put into the impedance match and then into the transducer array. A potential risk for the amplifier is making sure it can function properly for the set amount of time needed and enough voltage is supplied to provide the proper output.

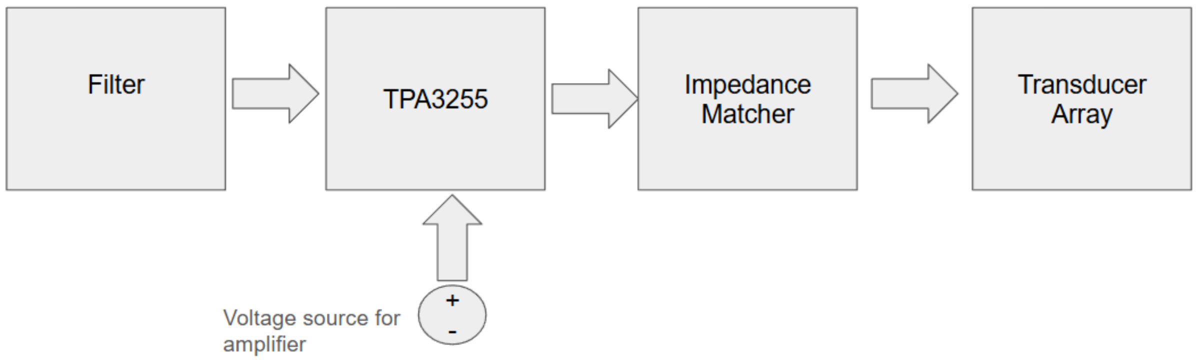

Figure 5. Subsystem Diagram of TPA3255

As shown in Figure 5, it is a more zoomed in block diagram on the TPA3255 chip, as it is a singular chip and we only have to worry about supplying the proper voltage amount for it to function and hooking up the correct input and output pins.

2.4 Modulation Subsystem

We plan to use the AD633, a four-quadrant analog multiplier, for the modulation subsystem, which is of central use: it will be used as a linear amplitude modulator. The AD633's transfer function is defined by the equation W = [(X_1 – X_2)*(Y1 – Y2)/ 10 V] + Z. The "Y1" input receives the 40 kHz carrier signal, while the "X1" input receives the filtered, centered, audio signal via the non-inverting amplifier output. The "Y_2" and "X_2" inputs will be grounded. The "Z" input receives the same 40 kHz carrier signal as "Y1". The output is "W", which is defined as [1 + (Audio / 10 V)]*Carrier*sin(ω*t). This output's high harmonics will be filtered out in the subsequent filtering subsystem. There are several design considerations for this subsystem and potential issues. Below we describe some potential implementations; however, this will be refined and potentially revised in the future more comprehensive design document.

The AD633 needs dual supply power inputs "V_s+" and "V_s-," so we will need to supply DC power in that way, at a minimum of +8 V DC and -8V DC, respectively, and possibly as high as +15V and -15V, depending on design considerations and exact calculations. A potential issue for the AD633 is high-frequency noise and lower-frequency fluctuations. One possible implementation is to connect a ceramic capacitor (possibly around 0.1 µF) between "V_s+" and ground, and an electrolytic capacitor (possibly around 10 µF) in parallel with it; the same type of capacitors would be connected between "V_s-" and ground in the same way. The ceramic capacitors are intended to provide a low-impedance path to ground for high-frequency noise, and the electrolytic capacitors are intended to handle temporary current demands during audio peaks.

The AD633 "X_1" filtered audio (modulation) signal input should be centered around 0 V for symmetry and proper modulation; however, the DAC may not necessarily produce such a signal. One possible solution is the following: the input audio signal is fed through a film capacitor (possibly around 1 µF), which connects to an op-amp; a resistor (possibly around 10 kΩ) is connected between “X_1” and ground. Another potential issue is that this signal may not be scaled properly in proportion to the carrier wave. A possible solution is this op-amp, which is set up as a non-inverting amplifier, with a potentiometer (possibly around 10 kΩ) as the feedback resistor, and the modulation index is tuned using the input potentiometer. Another issue is that since a 40 kHz carrier signal needs to be split into inputs of the AD633 “Y_1” and “Z,” there may not automatically be stable amplitude for both pins. One possible implementation is that the signal will first enter a voltage follower using an op-amp. The voltage follower’s output is simultaneously connected to “Y_1” and “Z.”

There are several testable requirements to determine the functionality of this subsystem. The modulation index “m” is defined as the ratio of the peak of the audio signal applied to “X_1” and the internal 10 V scale factor. Index “m” should be fairly close to 1.0 without exceeding it, between 0.5 and 1. The carrier signal should successfully and stably split to the “Y_1” and “Z” pins. The output should be a modulated signal defined as [1 + (Audio / 10 V)]*Carrier*sin(ω*t).

2.5 Transducer Arrays and Impedance Matching Subsystem

We plan to use 25+ identical piezoelectric transducers for the transducer array and impedance matching subsystem. They will be on a custom PCB in a hexagonal shape, tightly packed next to each other in a regular, aligned manner, so that they are in phase and most efficiently shaped. The transducers will be wired in parallel with each other, with each positive terminal connected to the input and each negative terminal connected to ground. A matching inductor will “impedance-match” the transducers, creating a series resonant circuit, depending on their overall equivalent capacitance (individual capacitance times the number of capacitors). This inductor will be in series with the entire grid of transducers. The inductance value is determined by the resonance formula, f_r = 1/[2*pi*sqr(L*C)]. Solving for L, we obtain L = 1/[C*(2*pi*f)^2]. The frequency “f” is the carrier frequency value; capacitance is the overall capacitance.

There are two physical implementations of this entire subsystem, with one having only the carrier signal as an input and the other having the modulated signal as the input. The demodulation will occur in the air with the physical intersection of the two arrays. To align the arrays so their beams intersect, one will be mounted steady. One possibility is for the other to be controlled by a stepper motor and drive. There are several subsystem requirements that can be tested. The signal after the inductor should be a clean carrier wave at 40 kHz for one array and a modulated wave for the other array. The intersection should be able to be precisely controlled by a user. In the intersection, the audio should be audible; this can be tested as long as the input waves are usable.

3. Ethics, Safety and Societal impact

We now look at the components such as ethics, safety and societal impact and the IEEE Code of Ethics and ACM Code of Ethics. There are multiple components that can be addressed within our project. First, we address IEEE Code of Ethics I.1 and section 1.2 in the ACM code of ethics; this is for our project and we as engineers need to always be aware of the public's safety and try to keep whoever uses our device away from any dangerous or harmful situations. This can be addressed in our design by making sure the output power is not too much, so that the driver does not damage the person listening or if someone is operating it that they can get harmed. Also, we plan on establishing rules for when the device is active, such as a minimum distance away from the speaker, so no participant's hearing is harmed, or the device causes them any sort of discomfort. Another key action that should be taken is to be transparent about any factors that can cause harm to the public; this goes along with Section 1.3 of the ACM Code of Ethics. We should be open to the public and people we work with on any potential dangers and harms that the device holds. This can be done through noting everything during testing and disclosing any potential risks we may see. When working on the project together and with others it is important that our group members follow Section 1.5 of ACM and II of IEEE. These sections note the importance of respecting one another and being open-minded towards new ideas. In our case, we should follow this through by not discriminating against one another or anyone involved in the process of creating the device. When looking at possible societal impacts and any misuse of the product we must take responsibility and possibly prepare precautions when possible. These possible misuses are someone using the device to harm someone else or the device being left by itself and being a possible safety hazard. In our case this can be done through having supervised operation of the device. Also, we plan on implementing safety features where the device is triggered by a button and only on for about 10 seconds before the button needs to be pressed again to enable the device. Overall, the team hopes to create a device that is safe, able to solve the issue, and enjoyable for the public, while allowing the public to be knowledgeable about its incredible features.

References

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[3] IEEE, “IEEE Code of Ethics | IEEE,” Ieee.org, 2020. https://www.ieee.org/about/corporate/governance/p7-8