Phased Speaker Array : Phase of Our Lives

ECE 445 - Senior Design

DESIGN DOCUMENT

February 22, 2019

Dave Simley (simley2) and Rosemary Montgomery (rjmontg2)
Team 68
TA: Kyle Michal

University of Illinois at Urbana Champaign
Department of Electrical and Computer Engineering

Contents

1 Introduction

1.1 Objective

Music can really brighten the mood and complement the atmosphere. What one may not realize, however, is that interference patterns greatly affect the quality of the sound. Say someone has a stereo in their living room, but likes to listen to music while preparing food in the kitchen. Destructive phase relationships can cause the listener to hear music that's tinnier and lacking warmth.

Our solution is to design a four speaker array which can "steer" the sound towards a single direction. Our design will use one or more IR sensing cameras to detect where the listener is in the room and apply a linear time delay across the speakers to direct the sound towards the listener.

1.2 Background

A collinear array of speakers create overlapping signals with the peak intensity of sound across all frequencies extending out directly from the center of the array. By applying a linear time delay across the array, the line of peak intensity steers away from the center line at an angle. With the help of IR sensors, we can track where a person is in the room and use this information to adjust how much we need to angle our line of peak intensity so that it runs through our listener.

Our project would allow a user listening to music to walk around a room and experience little to no phase cancellation. Therefore, our project would be useful for audio enthusiasts who admire superior sound quality. While rooms can be acoustically calibrated, nothing on the market exists that actively tracks a person's location and adjusts the signal accordingly. In addition, our project holds great educational merit, as it provides a platform where one can model and physically experience the effects of wave interference.

1.3 High-Level Requirements

1. A numerical parameter we plan to use to quantify the quality of phase accordance will take the root-mean-square error (RMS error) of the time delays between signals (comparing them to an average, ideal time delay). When steering is implemented at 30° from center, phase accordance between signals will significantly improve, resulting in a lower RMS error. Our project should reduce the RMSE of the signals by 75%.

2. The volume of sound from the array at the user's location must be $\geq$6dB when compared to that of a single speaker for frequencies of 440Hz, 1kHz, and 5kHz.
3. The direction of the improved sound quality should be updated at least every two minutes so that the user can move about the room with the improved sound quality following the user.

2 Design

2.1 Block Diagram

Our design has several modules including: the signal input module, the control module, the speaker array module, the power module, and the IR cameras. The control module takes an audio signal in and delays it accordingly. These delayed signals are then sent into the speaker array and are played through four collinear speakers.

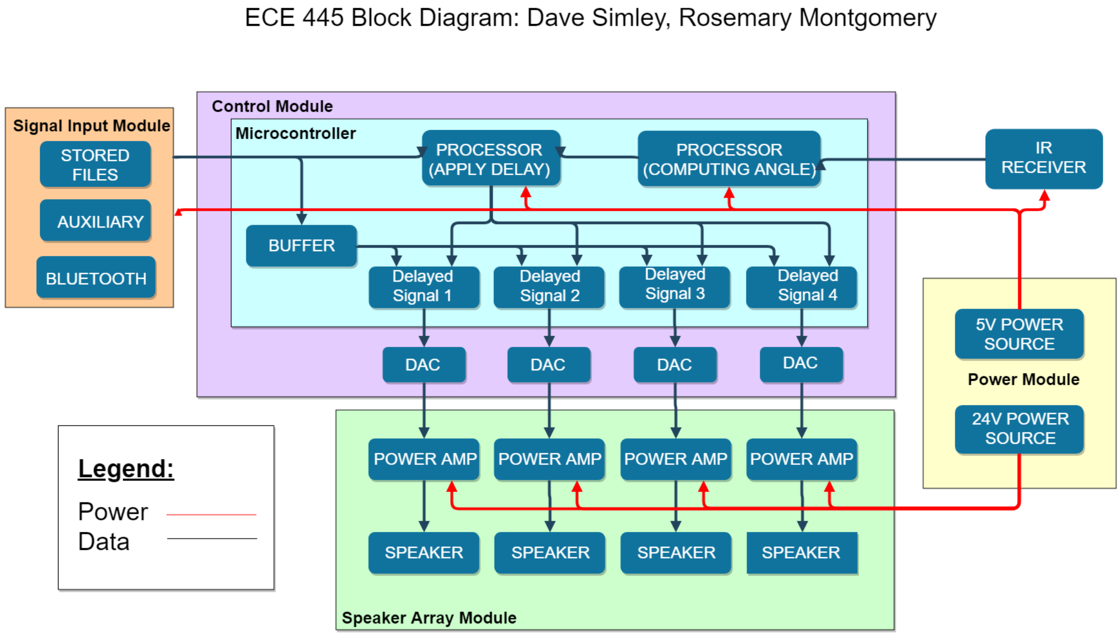

Figure 1: Block Diagram

2.2 Physical Design

The physical design will consist of an array of four speakers with two IR sensing cameras placed in the center. The speakers and cameras will be contained in an enclosure with the approximate dimensions shown in the diagram. There will also be a separate enclosure housing the control module and input module.

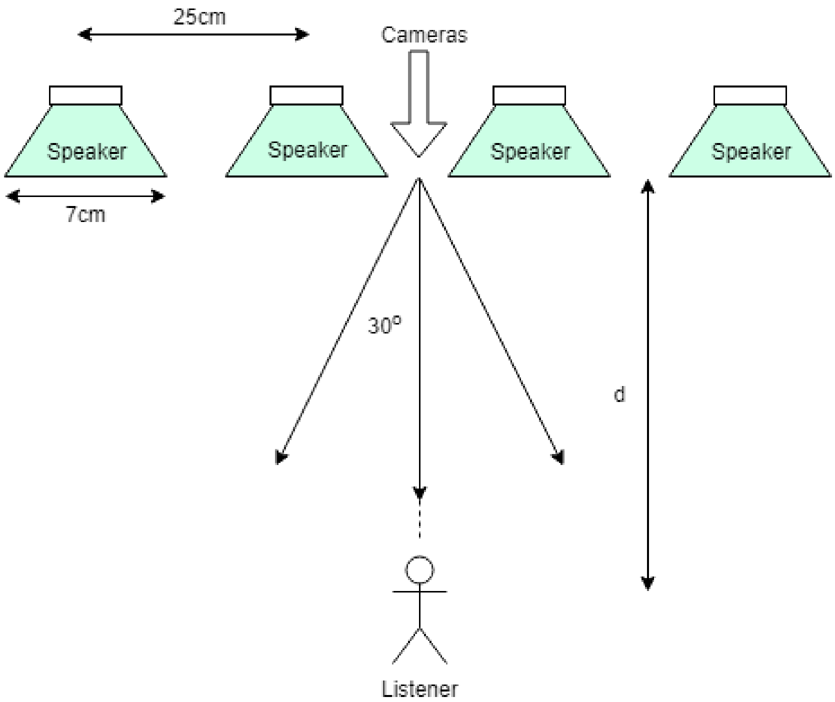

Figure 2: Physical Design

2.3 Functional Overview and Block Requirements

- **Control Module:** This module processes the location information of the listener (provided by the IR receiver), and calculates and applies the necessary delays to the four outputs (one for each speaker). This module is necessary for solving the first high level requirement: improving the phase accordance.

  - **Microcontroller:** A microcontroller will be used to process the tracking information provided by the IR sensing cameras, do the necessary calculations to determine how much to delay each speaker in order to "steer" the sound, and apply the delays. The microcontroller we will be using is the ATmega328.

Requirements	Verification
Must be able to read a digital audio signal and send its values serially to the DAC's within 11 microseconds.	a. Input a constant test voltage and connect a digital I/O pin to an oscilloscope. Program the microcontroller to serially output the test voltage every 11 microseconds. Ensure output takes less than 11 microseconds. b. Change the test signal to a signal with frequency 5.51kHz and repeat.
Must be able to store a large enough range of the audio signal to steer the sound at a 60 degree angle.	a. Create a data or loop structure with 63 fields. Load a repeating ramp signal with 32 fields into the structure, and have the microcontroller output the value across its digital output pins. Use an oscilloscope at each output pin to ensure that the value represented across the pins rises and recycles itself after 32 cycles.

– **DACs:** The DACs will be used to convert the digital signal to an analog signal that the speakers can use. They will be placed at the output of the microcontroller and input to the power amps. The DACs will have a 12-bit serial input interface.

Requirements	Verification
Must support 44.1kHz sampling rate.	a. Use the microcontroller to serially load a constant digital test signal into the DAC and connect the output to an oscilloscope. Set the strobe and load pins accordingly. Confirm that it outputs the reconstructed audio signal b. Change the test signal to a signal with frequency 5.51kHz and confirm the DAC outputs the reconstructed audio signal.

- **Signal Input Module:** This module prepares an input signal to be used in the control module. This module is important for the first two high level requirements since it provides the signal to be processed.
  - **Stored Files:** Our design gives the user multiple ways to input a sound file. The first is that there will be a .wav file already accessible to the microcontroller.
  - **Auxiliary:** Another option to input the sound file is to use an auxiliary cord to connect to a music playing device.
  - **Bluetooth:** The third option for inputting a sound file is to connect to Bluetooth, and be able to play a song remotely. This is a reach goal for the project. If we are able to include this feature, the user could also remotely adjust the volume and even be able to turn on/off the tracker.

Requirements	Verification
Must be able to convert an analog signal to a 12-bit digital signal, and communicate this value to the microcontroller within 25% of the sampling period	a. When given a constant DC signal, use an oscilloscope to confirm output is correct serial encoding b. When given a test signal with frequency 5.51kHz, use an oscilloscope to analyze serial output encodings, and ensure encoded values repeat the same pattern ([1, 0.7, 0, -0.7, -1, -0.7, 0, 0.7]) c. Ensure that the serial output of information takes less than 5.66 microseconds (25% of sampling period).

- **Speaker Array Module:** This module amplifies the signal and then projects the appropriate sounds for the listener to hear. This module is necessary for the second high level requirement so that the output can be heard by the listener and an increase in the decibel output can be detected.
  - **Power amplifiers:** The power amplifiers are needed to amplify the sound so that the speakers can output the audio. It powers the output of the DACs and plays the signals through the speakers.

Requirements	Verification
Must be able to output at least 1W of power to speakers when given 24V of power	Apply a load at the output of the power amplifier circuit. Set the power supply to 24V. Use a multimeter to measure the voltage and current across the load. Use P=IV to confirm output is >= 1W.
Power output must be within 20% (1dB) of other speakers.	Use NdB = 20*log(Po/Pi) where Pi is the power calculated in part 1,2 and Po is the power used by the speakers, found in the verification below. Ndb must be <= 1dB.

- **Speakers:** The speakers convert the electronic audio signal of the song being played to directional air pressure (i.e. music). It recieves the signal from the power amplifiers.

Requirements	Verification
Volume output must be within 20% (1dB) of other speakers.	Set a microphone 3m away from a speaker. Apply a constant signal to the input of the speaker. Use the microphone to record the output of the speaker. Use MATLAB to determine the volume output. Repeat for the other speakers and compare the results. Must be <= 1dB

- **IR Receiver Module:** This module receives data on the location of the listener and reports it to the control module. This module helps solve the high level requirement of updating the status of the listeners location at least every two minutes.
  - **IR sensing camera(s):** IR sensing cameras will be used to detect where the listener is in the room. This tracking information (the x-y location) will be sent to the microcontroller to be processed. We need two cameras, as each one detects a range of about 33 degrees, so two cameras would give us a field-of-view

(FOV) of approximately 60 degrees.

Requirements	Verification
Must be able to detect angle towards infrared source within 5% accuracy.	a. Set up camera so that it is horizontal to the ground and apply a 3.3V/5V voltage supply to power it. b. Measure out three positions a 0-3m radius from the camera and supply a heat source at each spot. Record the location using the IR camera at each point. Measurements must be within 5% accuracy.
Must be capable of updating the position at least every 2 minutes.	Set up the camera as described above. Record measurements every 2 minutes.

- **Power Module:** This module supplies all active components of the design with power. This includes: the microcontroller, power amps, input module, and IR cameras. This module is necessary to solve all three high level requirements since each requires a power supply.

  - **Power Supply (5V):** A 5V power supply will be used to supply power to the control and signal input module. The supply will need to be steady, well-filtered, and able to provide enough current to adequately power the digital circuitry involved in processing the audio signal.

Requirements	Verification
Must provide 5V ±5% at peak output current draw of 300mA.	Apply a load at the output of the power supply. Set the power supply to 5V. Use a multimeter to measure the voltage and current across the load. Voltage must be between 4.75V-5.25V.

  - **Power Supply (24V):** A 24V power supply is necessary to provide power to the power amplifiers. The power amplifiers require a stronger power supply because the electrical output of the amplifiers are physically moving the diaphragms of the speakers. Therefore, a higher current draw should be expected.

Requirements	**Verification**
Must supply 24V ±5% at peak output current draw of 1A.	Apply a load at the output of the power supply. Set the power supply to 24V. Use a multimeter to measure the voltage and current across the load. Voltage must be between 22.8V-25.2V.

Schematics, Calculations, and Supporting Material

Physical Spacing of the Speakers

Given that we have four collinear speakers, how do we determine "a", or the distance between the speakers? Consider the diagram in figure 3.

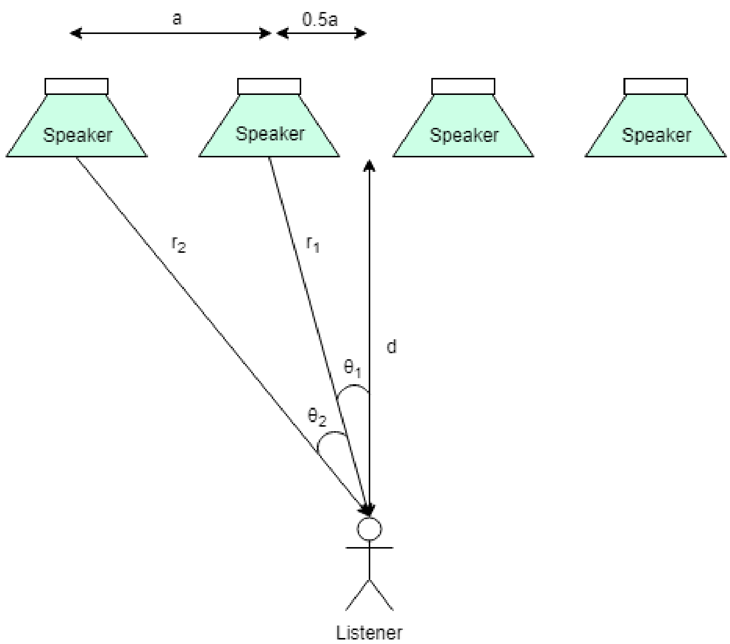

Figure 3: Our Physical Design

Here, we define the distance the user from the speaker as d, the distance between the speakers as a, and the distance from the listener to the far left and center left speaker as r_2 and r_1, respectively. Our design relies on the assumption that we can assume a "far-field" approximation; that is, we can assume the listener is far enough away from the speaker array that the different distances the user is between the speakers is negligible. If we define a, we can determine the difference between the r_1 and r_2 as a function of d.

Let's define a as equal to two-thirds of a meter. If we vary d from 0 to 10 meters, the difference between the distances from the speaker $(r_2 - r_1)$ is plotted in figure 4.

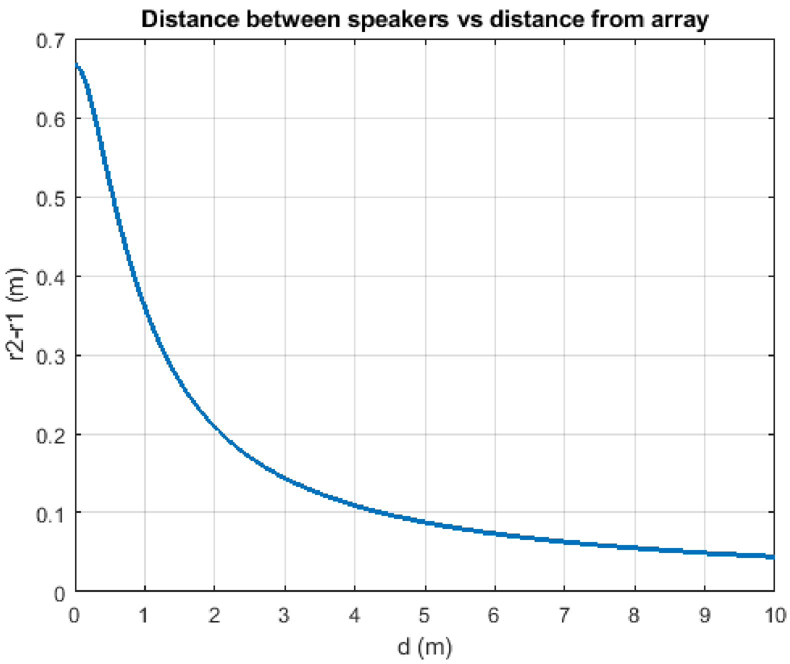

Figure 4: $r_2 - r_1$ vs. d

At $d \approx 4m$, we see that $r_2 - r_1 = 0.1$m. This will be an issue, as signals with a frequency as low as 1.7kHz ($\lambda = 0.2$m) will completely cancel out due to destructive phase interference. Considering human hearing is centered around 2k-5kHz [1], we should design the speaker array so that no destructive interference occurs with frequencies below 5kHz at $\sim$3.5m from the array.

Theoretically speaking, we could just shrink a to as small as the speaker diameter, but in reality, we have another constraint. Let's say we try to steer the sound at a 90° angle. To do this, we have to have the sound from the far right speaker reach the far left speaker as the far left speaker plays the original sound. Essentially, we need to delay the sound from the far left speaker by the time it takes for the sound from the far right speaker to reach the far left speaker.

This is an easy calculation, as the speed of sound is 343m/s, and the distance from the far right speaker to the far left is $3a$. Therefore, the time it takes for the sound to travel across the speaker array is $t_{\max} = \frac{3a}{c}$. For $a = \frac{2}{3}$m, we have find $t_{\max} = 5.83$ms. Therefore, the microcontroller must be able to store up to 5.83ms of sound. At a sampling frequency of 44.1kHz, we find that the total number of samples we need to store is equal to $f_s t_{\max} = \frac{3 f_s a}{c}$. For $a = \frac{2}{3}$, this we find $T_{max} = 257$ samples.

Considering we are applying a linear delay across the speakers (i.e. speaker 1 is delayed by 0x[1ms], speaker 2 is delayed by 1x[1ms], speaker 3 is delayed by 2x[1ms], and speaker 4 is delayed by 3x[1ms]), having 257 stored samples means we have $\frac{257}{3} = 85$ increments between 0° and 90° to which we can steer the signal, meaning our project can steer sound in 1.2° increments.

Now, if we want to decrease a, so as to improve our phase response, we will also decrease the accuracy of the array's ability to steer the sound. Given we want ±5% accuracy over a FOV of 60°, we want to be able to steer the sound in 6° increments. Using the same logic, but in reverse, we determine that the minimum size for a in which we still have control over our accuracy is 23.3cm, which we can round up to 25cm.

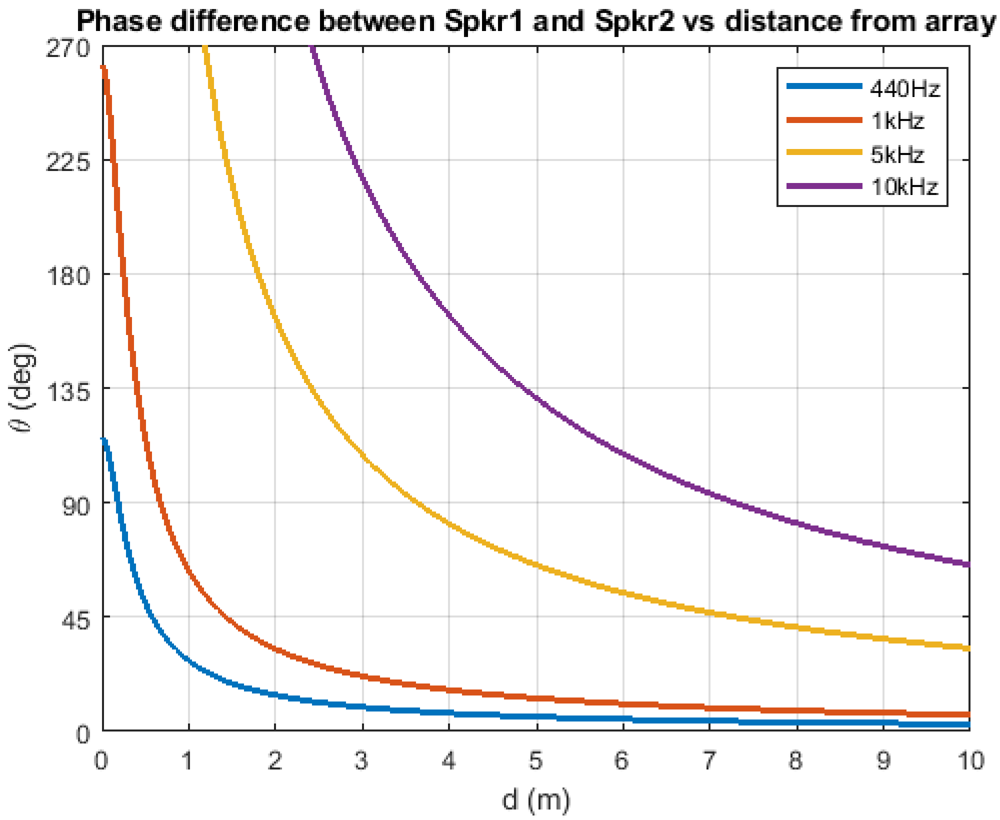

Figure 5: Phase shift vs. d for $a = 25$cm

In figure 5, we see that for distances greater than ∼3.5m away from the array, all frequencies under 5kHz experience no destructive interference. Therefore, our design will use a value of $a = 25$cm.

Power Amplifier Schematic

In figure 6, we can see the schematic design of the power amp, along with it's frequency response. Given that we want to deliver 1W of power to the speaker, we find that to deliver 1W of power, we need to apply a voltage of approximately 3V and deliver approximately 0.5 Amps. Therefore, we need to use a Darlington pair of BJTs to deliver the current, and the preamp has a low gain.

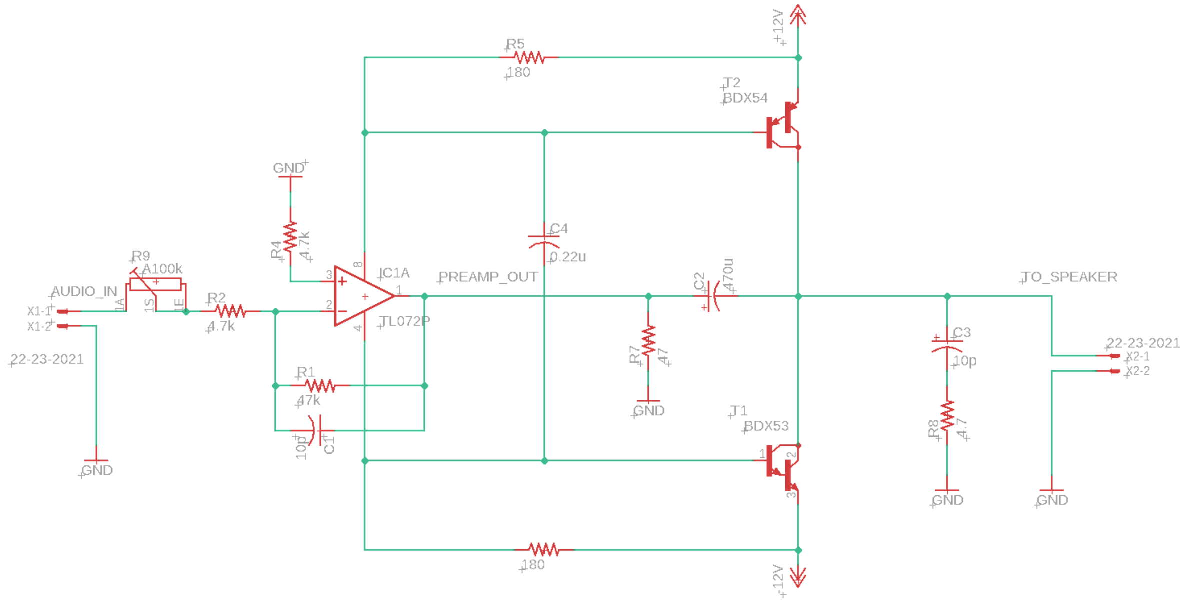

Figure 6: Power Amplifier Schematic [2]

In figure 7, we can see that the signal reaching the speaker has gone through a high pass circuit. Due to our speakers'smaller diameter, the viable frequency range of our speaker lies between 250-20kHz. Given that most fundamental frequencies in music exist between 250-2kHz, a high pass filter before the speaker would allow the fundamental frequencies to punch through, and provide a clearer image of peak intensity.

$$P_{tot} = V_{tot} I_{tot} = \frac{V_{tot}^2}{R} = \frac{V_{tot}^2}{8\Omega} = 1W$$

$$V_{tot} = 2.82V$$

$$I_{tot} = 353mA$$

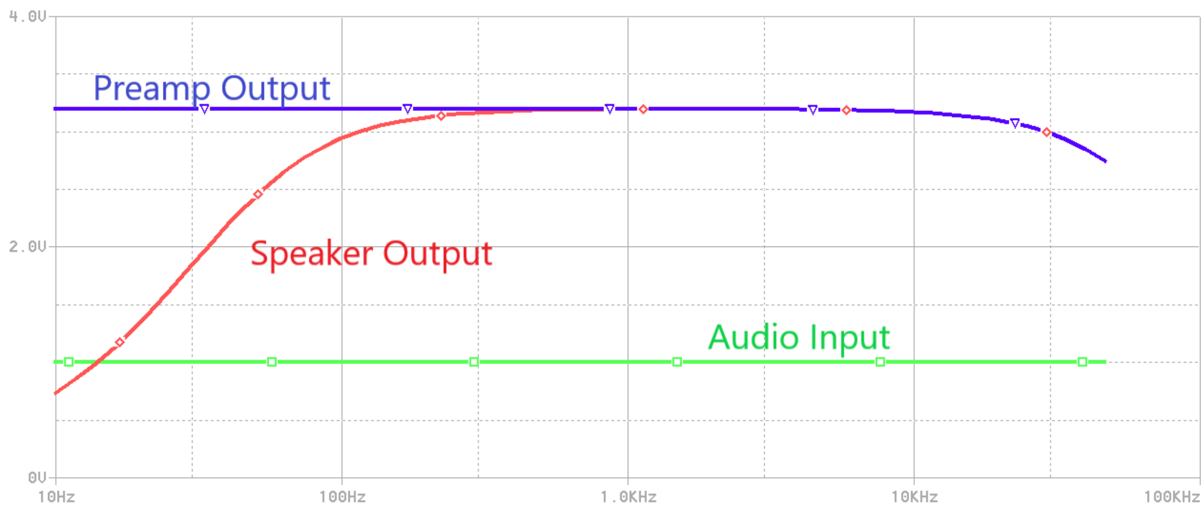

Figure 7: Frequency Response of Power Amp

When simulated, we find that we are delivering approximately 1.2W of power to the speaker when our potentiometer is set to 10kΩ. As the gain of the circuit can be adjusted, we find that at maximum gain, we are delivering 12W of power to the speaker, and at minumum gain, we are delivering 0.025mW of power to the speaker. Depending on the rating of our speaker, we can easily limit the circuit from overpowering the speaker at the output by replacing the first two resistors seen at the input with larger values.

2.4 Tolerance Analysis

The part of our project that poses the greatest risk of failure is the control module. It demonstrates the most complex interfacing between modules and the success of the project relies nearly entirely on the ability of the microcontroller to delay an analog signal to four different degrees and send them to four different speakers. This process flows from point A, to point B, to point C, and so on. If the process that takes us from point B to point C doesn't perform as intended, our project will not operate properly. Every process needs to be tested and verified independently, then integrated seamlessly. Furthermore, the final device calibration cannot be done by ear or human inspection; a professional-grade digital audio workspace, microphones with good audio clarity, and a reliable computer are needed to make sure the device is operating properly.

The interfacing required by the microcontroller poses many challenges. The first being that the microcontroller must be able to take in digital samples at a frequency of 44.1kHz (sampling period of 22.7us). Given that our microcontroller has a clock frequency of 16MHz (62.5ns/inst), we predict that it will take approximately 24 instructions to read a

12-bit signal, and up to 8 instructions to establish the connection and prompt the data transfer. This process should take less than 10% of the sampling period (by the end of which we must process a new sample), but we will allocate up to 25% of the sampling period to perform this process.

The second challenge is storing the new sample, and then accessing multiple samples simultaneously, and sending these values serially to the 4 DAC's. We predict that it will take up to 6 instructions to send one bit of information to the four DAC's: 4 instructions to set the digital output pins, and 2 instructions to strobe the STB pins of the DAC's. Once 12 bits of information have been transferred, we must strobe the LOAD pin of the DAC's, so that our samples are represented at the analog end of the DAC's. In total, this will take 74 instructions, or 21% of the sampling period, but we will allocate up to 50% of the sampling period to perform this operation.

The third challenge is interfacing the microcontroller with the IR camera. Given that the IR camera can communicate with a microcontroller with an I^2C connection, we predict this to be an efficient process, and will allocate up to 25% of the sampling period for the microcontroller to update the tracking information. In addition, the tracking information only needs to be updated once every 10,000 samples (once per quarter second) to follow typical human movement in real time, and once every 5 millions samples (once per 2 minutes) to satisfy our lower bound case of updating the tracking information every 2 minutes. In conclusion, we've allocated approximately 2.5x more time needed for each process to account for possible tolerances in each process.

In addition, our power amp poses some risks. Power amplifiers typically require large coupling capacitors. These capacitors are electrolytic, large, and have very high tolerances, usually ±20%. Because the coupling capacitor creates a low pass filter with the speaker, the value of the coupling capacitor has a great effect on the frequency response of the system.

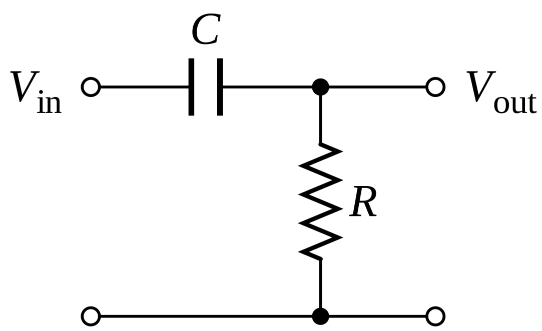

Figure 8: Low-Pass Filter Circuit

Figure 8 shows how our coupling capacitor plays into our circuit. The resistor represents our load (the speaker), and the capacitor represents our coupling capacitor. Our capacitor has a value of 470uF±20%, our speaker has an impedance of 8ohms±15%, and our circuit contains another resistor in parallel with the speaker that has a value of 4.7ohms±5%. Using the following equation to describe the relationship between the values of an RC circuit and it's -6dB cutoff frequency, we find that the -6dB frequency of the circuit varies from 88Hz to 157Hz (with the actual value centered around 114Hz).

$$f_{-6\mathrm{dB}} = 2\pi RC$$

Given that the speaker's frequency range is 250-20k Hz, the effect of the varying tolerance of the components won't have a great effect on the overall operation of the speaker array.

The greater risk the power amp poses is its feasibility. The circuit operates properly when simulated, and shows promising results, but the power required by the circuit to operate casts some doubt on its feasibility. The Darlington pair of transistors exist in the circuit to provide the massive current draw needed to power the speaker, but how it interacts with the preamp section of the circuit remains a mystery. While this circuit is technically proven to work, other designs for the power amp are still being considered.

In order to interface with an auxiliary cord to a music playing device, the analog signal of the song needs to be converted to a digital signal so that the microcontroller can digitally process it. There is an ADC built into the ATmega328P microprocessor but we are concerned about it's robustness. As a result we have identified alternative options including a separate ADC chip to handle the conversion. Another option is to take an input from a memory device such as an SD card, in which the data would be ready to be processed.

The other modules in our design pose a much smaller risk. The DAC's, speakers, and IR cameras are robust devices that have plenty of documentation to allow for seamless integration.

While calibration may be tedious, the end result will be impressive. In addition, our team has access to not only every item needed to calibrate the system, but also access to a wealth of university resources and insightful advice from our professors and TA's. In conclusion, our project will be challenging, but our determination and the supportive course staff will keep us focused and assure our project's success.

3 Cost & Schedule

3.1 Cost Analysis

To calculate our labor costs, we are assuming a pay of $40.00 an hour, for two people working 10hrs/week for 14 weeks, multiplied by 2.5x. We calculate our costs due to labor to be $28,000.

$$\text{Total Costs} = (2\text{ people})\frac{\$40.00}{\text{hr}}\frac{10\text{hrs}}{wk}(14\text{ weeks}) = \$28,000$$

Part:	Cost (prototype):
IR cameras	$46.00
DACs (serial input) (x4)	$42.92
Speakers (x4)	$16.56
Microcontroller (ATmega328P)	$2.14
Power Supply	$12.00
Arduino Uno	$22.00
ADC	$7.00
Heat Source	$0.00 (in lab)
PARTS SUBTOTAL:	$148.62
LABOR SUBTOTAL:	$28,000
TOTAL:	$28,148.62

Table 1: Cost Table

3.2 Schedule

Date	Dave	Rosemary	Notes
2/4/19	Research for proposal	Research for proposal	Proposal Due
2/11/19	Research components, designs, feasibility	Research components, designs, feasibility	
2/18/19	Research power amps, power supplies, and acoustical considerations	Research DACs, microcontrollers, and interfacing	Design Document Due
2/25/19	Finalize schematics and prepare circuit board designs	Prepare first round order list and order parts	Design Review
3/4/19	Finalize signal input module design, order necessary components	Design PCB for control module, ready to order	
3/11/19	Prototype and revise, schematics, design preliminary PCBs and place order	Interface DACs with microcontroller, test and verify proper communication	1st Round PCB Orders
3/18/19	Construct Enclosure	Interface microcontroller with signal input module	Spring Break
3/25/19	Populate and test PCB's, ensure proper operation, make necessary adjustments and order new PCB's if needed	Interface control module with IR camera, finish interfacing microcontroller with signal input module	Final Round PCB Orders
4/1/19	Interface speaker array with control module, perform preliminary tests to ensure proper operation	Perform tests to ensure proper interaction of control module with IR cameras and with signal input module	
4/8/19	Integrate modules into single enclosure, test operation	Integrate modules into single enclosure, test operation	
4/15/19	Refine project, begin final report	Refine project, begin final report	
4/22/19	Prepare for demonstration	Prepare for demonstration	Demo
4/29/19	Prepare for presentation	Prepare for presentation	Final Presentation

4 Ethics and Safety

On the surface, our project poses no immediate ethical or safety concern. Our project cannot endanger others, won't have a minuscule effect on global health, nor does it deal with any personal information. However, our project does pose a few health risks to those who operate it and to those who are in the direct path of sound.

The greatest risk our project poses to public health is that exposure to loud sounds for long periods of time can lead to hearing loss. To address this, we put proper constraints in place at multiple levels of the project to assure auditory safety. Our power amps will deliver at most 1W of power through each speaker, which will provide approximately 80dB of sound to a listener standing one meter away from the speaker array. We predict that we will likely restrict our speakers to only put out a quarter watt of power, but may require extra headroom in case outside noises become a concern while calibrating our system.

In addition, our project involves designing and constructing power amplifiers for the speaker array. The power amp will run off of a 24V power supply, so we must be conscientious of those who construct and operate this device. Our power amps will be enclosed and wires coming to and from the amplifier module will be fastened securely. In addition, multiple circuit breakers will exist at every level of power consumption to assure that any shorts won't harm the circuitry or, more importantly, the user. We will adhere to #1 in the IEEE Code of Ethics and make design decisions "consistent with the safety, health, and welfare of the public" [3].

Finally, we will act in accordance of #5, #6, and #7 of the IEEE Code of Ethics [3] in order to contribute to the development of audio technology. We will do this by seeking out criticism of professors and teaching assistants that have expertise on the details of our design, and properly credit them for their involvement.

References

[1] S. A. Gelfand, *Essentials of audiology*, 3rd ed. Thieme Medical Publishers, 2009.

[2] Electronic Circuits and Diagrams-Electronic Projects and Design. (2019). *Electronic Circuits and Diagrams-Electronic Projects and Design - An Authentic Resource on Electronics.* [online] Available at: http://www.circuitstoday.com/ [Accessed 19 Feb. 2019].

[3] Ewh.ieee.org. (2019). [online] Available at: http://ewh.ieee.org/cmte/substations/posted_documents/ieee_codeofethics.pdf [Accessed 19 Feb. 2019].

[4] Kinsler, L. (2000). *Fundamentals of acoustics.* 4th ed. New York: John Wiley & Sons.