Active Noise Cancellation for a Loud Laboratory

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There are two phases to this experiment: one that is described here and one that will be working towards the active noise cancellation. In the first phase, sound waves coming from an audio source were measured to see if destructive and constructive interference were observed. The measurements were made using a Bluetooth amplifier connected to two speakers to make the sound, and using a microphone that is connected to a microcontroller and memory devices to record the sound. With this setup, it is possible to observe constructive and destructive interactions. Since it is possible to see these interactions, the experiment will move onto phase two. Phase two will consist of creating a device that is able to record the sound of a pump in a laboratory, phase shift the sound, then play out the manipulated sound where it destructively interferes with the initial sound.

I. Introduction

The objective of this project is to create a continuous noise-canceling device. The idea is that the device listens to incoming noise through a microphone and outputs a phase-shifted sound wave to cancel out the initial noise. That is our overall goal for this project, but this paper will go more in depth on the current stage of this project. Where this experiment aims to confirm that constructive and destructive interference behave as expected, laying the foundation for the next steps.

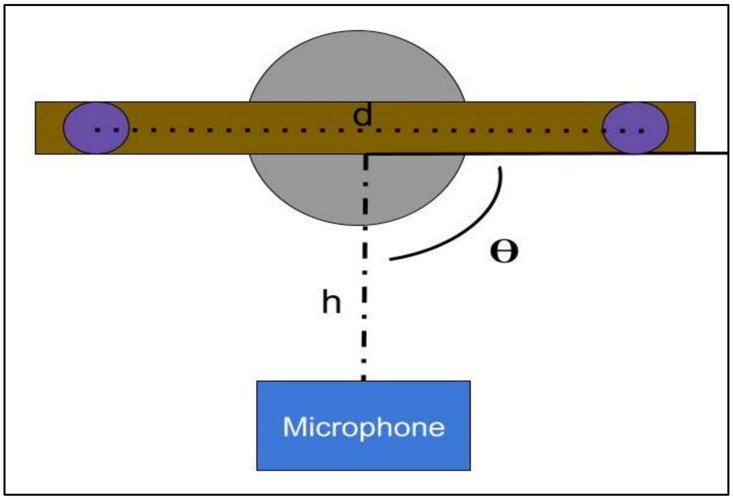

Figure 1.1 A diagram of the experimental setup where the grey circle is a table that can spin, the purple circles are the speakers, and the brown rectangle is an apparatus that holds the speakers. d and h are consistent values of one meter and forty-six centimeters, respectively.

The main concepts are constructive and destructive interference, and the relationship between the wavelength and the difference between the distance of the sound traveled from both speakers. As well as how this relationship changes when one of the speakers is phase shifted. If the distance from the center between the two speakers and the microphone are constant, then the goal is to find at what angles constructive and destructive interference occurs, as shown in Figure1.1

The hardware used for this project includes a microcontroller, microphone, amplifier, speakers, and memory. Where the microcontroller drives the system, where the microphone sends a signal to the memory. The memory stores the data and its time values in terms of microseconds. One program used is Arduino IDE which communicates with the hardware and sends instructions for it to follow, as described. Once the data are collected, it is sent to a computer where it can then be analyzed through Python. It is through Python that graphs of the amplitude and Fourier transforms are made.

The analysis examines the graphs created from the initial runs of the experiment, where a tone generator on a phone was connected to speakers through an amplifier and played at a consistent frequency. The microphone was placed at different angles relative to the center between the two speakers. The discussed graphs show the amplitude and frequency calculated from data collected at two angles of constructive interference, which display a calculated frequency lower than the one used during the experiment. Possible reasons for this discrepancy are then explored.

II. Theory

In this experiment, concepts of waves and the implications of recording versus playing them come into play. The goal is to be able to understand waves well enough to know when they cancel versus add, while also being able to know the interaction between the computer and microphone/speakers well enough to mimic a sound.

Waves have many different properties, and the most important one here is interference. For the experiment, constructive and destructive interference were focused on. Constructive interference is when two waves add to make a larger amplitude wave. This is caused when both waves are at their peak in the same place at the same time. Destructive interference is when two waves add to nothing. Instead of both waves being at the same peak value, they are at opposite values. When the waves meet at that time, they then cancel each other out.

Where the waves cancel each other out is predictable using the equation below (1). d_1 and d_2 are the distances from the audio sources. λ is the wavelength of the sound wave and n is a positive integer. This equation is true for all spherical waves.

$$d_1 - d_2 = (n + \tfrac{1}{2})\lambda \qquad (1)$$

While (1) is known, it is possible to derive where the interference will occur by adding two wave functions. If two wave functions are defined, one with a phase shift (ϕ), it is possible to predict their interactions by adding them. Below wave functions $f_1\ and\ f_2$ are defined where A is the amplitude (which is the same for both waves), k is the wave number, and w is the angular frequency of the wave.

$$f_1 = Acos(kx - wt)$$
$$f_2 = Acos(kx - wt + \varphi)$$

After adding the two wave functions, the equation can be simplified using a trigonometric identity. This equation is below (2).

$$f_1\ and\ f_2 = 2Acos\left(\frac{\varphi}{2}\right)\cos\left(kx - wt + \frac{\varphi}{2}\right) \qquad (2)$$

Using this formula (2), we can see that the waves added will equal zero when the phase shift is an odd variation of pi. It is essential to know where audio signals are added and cancel due to trying to cancel the sound for parts of the room. Knowing multiple ways of calculating these areas allows the math to be double checked.

Along with knowing when the audio sources will cancel out sound, microphones will be used to record the sound of a noise. Actively recording the sound that is desired to be canceled comes with some theoretical issues. To cancel the sound, the recorded sound would have to be phase shifted 180 degrees and then played through the speaker at the same time the original sound passes by. Not only is there a timing issue, but the microphone is constantly recording, meaning it will be able to hear the speaker.

III. Hardware

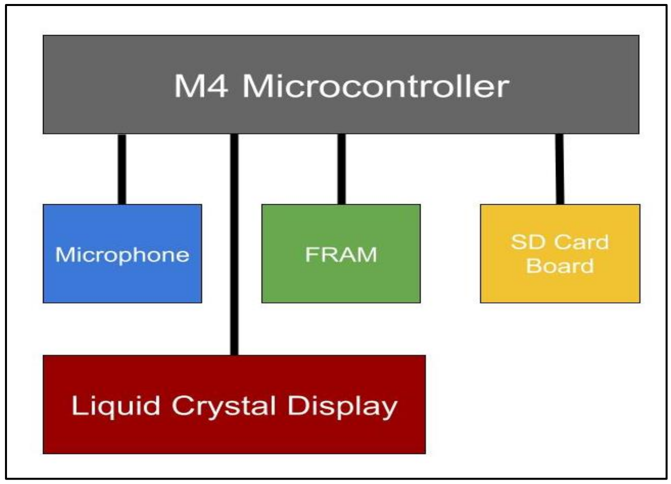

Figure 12.2: A block diagram of the hardware setup excluding the amplifier and speaker setup.

The system used in this experiment as shown in Figure 1.2 includes the Feather M4 Microcontroller, which is connected to the microphone, amplifier, and SPI FRAM. The microphone sends its signal to the FRAM, which stores the Analog-to-Digital Converter (ADC) counts, representing the electrical output. The FRAM also stores the associated time values in microseconds and then saves this data to the SD card. Throughout the process, the Liquid Crystal Display (LCD) shows the current status of the M4 microcontroller.

The M4 microcontroller serves as the central component of the system, communicating with the Arduino IDE. It is directly connected to the amplifier, microphone, SPI FRAM, and the software that runs the code.

The microphone used in this experiment is the Adafruit MAX4466 MIC AMP. It has been soldered to wires and mounted on a tripod, allowing it to be moved easily to collect data at different locations. While this method introduces uncertainties each time the microphone is moved, it provides more precision when taking measurements compared to holding the breadboard it was initially attached to. The microphone's gain, adjusted to the maximum on the back of the device, enhances sensitivity, enabling it to capture frequencies that might otherwise be missed. The gain can also be calibrated during analysis using proper scaling.

As it is currently being used, the amplifier has ports that connect it to the speakers, allowing the phase of one or both speakers to be inverted. Operating the amplifier in this way, requires it to be set to Bluetooth mode, enabling the tone generator on a phone to connect to the speakers.

For future use of amplifier as it relates to the main goal of this project, has auxiliary ports that connect to the M4, enabling the transmission of signals from the microphone. To operate the amplifier in this manner, it must be set to Aux mode, enabling the microphone signal from the M4 to be sent to the speakers.

The current setup for the speakers is mounted on an apparatus that securely holds them in place. This setup ensures better control of their positioning and the distance between them, contributing to consistency and repeatability when collecting data. Currently, the speakers are configured to play frequencies for the microphone to detect, using the amplifier's Bluetooth functionality.

The SD breakout module is used to save collected data. It connects the micro-SD card to the system, as the M4 does not have a built-in SD card reader. Data sets are sent to the SD breakout from the SPI FRAM.

The SPI FRAM provides memory to store microphone data for up to eight seconds. Once the storage capacity is reached, the FRAM transfers the data to the micro-SD card in the SD breakout. The M4 drives the FRAM's operations.

When the speakers and amplifier are set up separately from the rest of the system and are playing a frequency, the Arduino code is uploaded to the M4. The system begins recording ADC counts and corresponding time values using the FRAM, then saves the data to the SD breakout. The data on the SD card is later transferred to a computer, where Python code is used to generate amplitude and frequency graphs.

IV. Analysis

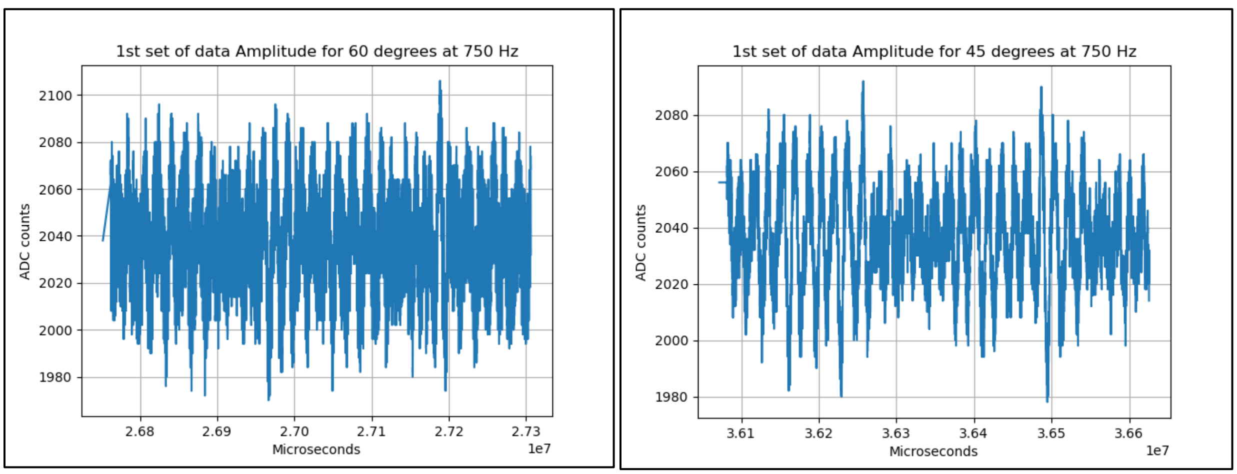

Figure 1.3: Two signal graphs where theta is 60 degrees on the right and 45 degrees on the right.

The graphs in Figure 1.3 are ADC count graphs versus the time in microseconds. This allows a visualization of what the microphone records. These graphs are beneficial for many reasons. One reason is they can be analyzed to determine if there was some type of interference present. There should be a smaller amplitude on the graphs where there is destructive interference and a larger amplitude where there is constructive interference. Another reason these graphs are beneficial is the possibility of seeing how much 60 Hertz is present. 60 Hertz is the frequency for power in the United States, so if the measurements are made close to a power source there is a possibility that it becomes present in the data. If there is a lot of this frequency present it is possible to reduce that signal by altering the setup. Eliminating this signal would make the data more accurate. The data used to make these graphs can also be put through a fast Fourier transform.

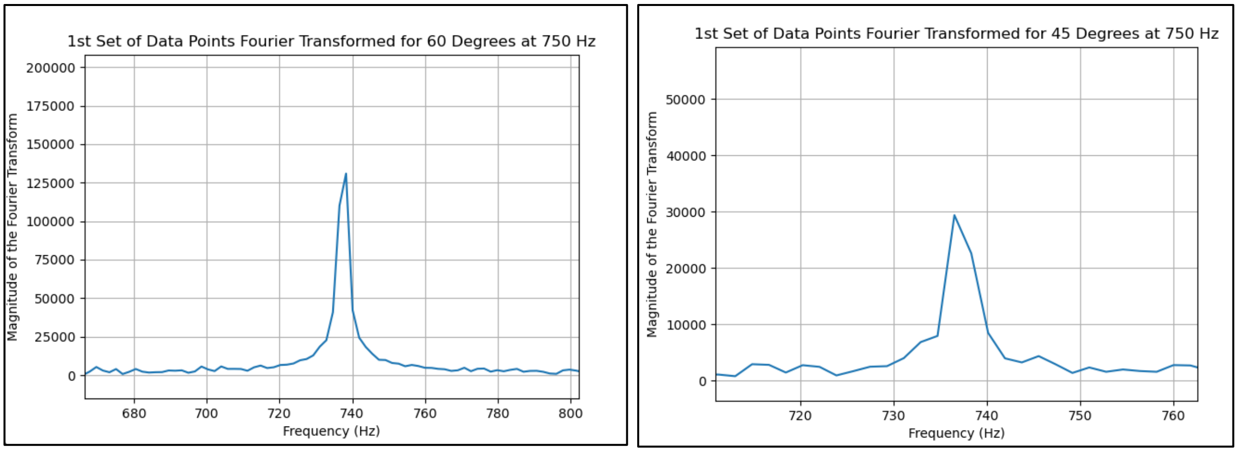

Figure 1.4: Two frequency versus magnitude graphs depicting the fast Fourier Transform of the ADC values from the microphone.

To obtain the results in Figure 1.4, the equipment was set up as shown in Figure 1.1. The speakers were in phase with each other both playing 750 Hertz. For the graph on the left, theta was 60 degrees. Similarly, for the graph on the right theta was 45 degrees.

After using equations (1) and (2), it was determined that at 60 degrees the sound waves should have constructive interference while they should have destructive interference at 45 degrees. On the y-axis it has the magnitude of measurements at a given frequency. This magnitude is expected to be larger where there is constructive interference and small where there is destructive interference. In Figure 1.4, the left graph shows a magnitude of over 125,000 while the right graph shows a magnitude of close to 30,000 this is a magnitude of around four times lower than the left graph. This shows constructive and destructive interference.

Looking at the graph and the information given there appears to be a discrepancy. The graph says the frequency is under 740 Hertz, yet it was said the speakers were given 750 Hertz. This issue has been observed with many graphs and it is likely an issue with the calculation of the frequency. Another discrepancy that was discovered is that the graph that was claimed to be the destructive interference graph still has a sizable amplitude. This could be due to the objects in the room bouncing the sound off it, incorrect placement of the microphone, or a miscalculation. Most likely it is due to incorrect placement of the microphone. The distance from the speakers got measured with a tape measure. It is likely the exact spot of deconstructive interference was not found.

Though there is an error with the calculation of frequency, the graphs still show both types of interference where they were calculated. Since the graphs show both types of interference, it has been determined that the experiment will move on to phase two.

V. Conclusion

The initial goal of this experiment is to verify that the data shows the type of interference that is expected at different locations. Currently the system is a smartphone tone generator app to play a frequency connected by Bluetooth to an amplifier so the microphone can record and save it. With the current setup the microphone can pick up where the waves are constructively interfering versus destructively interfering. There are some issues that must be addressed concerning the measurements and analysis. Though there are some issues, the data shows that constructive and destructive interference is recorded. In this experiment as a whole, the goal is to create an active noise canceling device that records audio and plays it back 180 degrees phase shifted. The goal is for the system to become a microphone connected to an M4 that takes in the audio, phase shifts it, plays out the altered audio through two speakers, and then saves the audio for later processing. The next steps will be to implement this system in a room with a loud pump and make the necessary adjustments to the setup.

References

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