# Engineering 72A Final Project: Binaural Locator

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# Abstract

A binaural locator was constructed to measure the lateral location of a 1 kHz sound source by measuring the phase difference in the signal at two microphones. The signals were amplified and filtered to obtain a DC signal representing the phase difference between them, and they were analyzed with a flip-flop to determine the signal to which the sound source was closer. Our final design behaved as expected, although the angular sensitivity was poor.


# Introduction

Figure (1) shows the setup for our project: two microphones are located a distance $L \approx 15$ cm apart, and a sound source outputing a 1 kHz sine wave is held a distance $r \approx 90$ cm away from the midpoint of the microphones. The angle $\theta$ that the source is located from center can be determined based on the phase difference between the signals at the microphones because the signals must travel different differences, $d_1$ and $d_2$. If $\phi$ is the phase difference between the two signals and $\lambda = v_s/f \approx 0.34$ m is the wavelength of our sound waves ($v_s \approx 340$ m/s in air), then

$$d_2 - d_1 = \frac{\phi}{2\pi}\lambda.$$

We also know geometrically, by the law of cosines, that

$$d_2^2 = r^2 + \frac{L}{4} - rL\cos(\frac{\pi}{2} + \theta), \tag{1}$$

$$d_1^2 = r^2 + \frac{L}{4} - rL\cos(\frac{\pi}{2} - \theta). \tag{2}$$

Since we have three equations and three unknowns ($d_1$, $d_2$, and $\theta$), it is therefore possible to determine $\theta$. Note, however, that this model can only tell us the magnitude of $\theta$; to determine its sign, the inputs at the microphones must be examined to see which signal is leading the other.

For humans, the smallest noticeable change in $\theta$, known as the "localization blur," has been experimentally determined to be between $1^o$ and $4^o$, and it is smallest directly in front of the subject and largest at right angles to the subject. Spatial hearing expert Jens Blauert reports that most authors in this field agree that "interaural time differences," or arrival time differences between the two ear input signals, are the most important factor in laterally locating sound events. They also believe that the ambiguity about the sign of $\theta$ is resolved by examining the amplitude of the signal at each ear, an effect that is too small to be noticeable in our design. Our circuit is unable to differentiate between vertical changes in the location of the sound source, which, interestingly, is also something that human subjects have difficulty with. Since the input to each ear is approximately the same whether the sound source is in front, behind, or directly overhead, humans often guess wrong when asked to locate these sources.[1]

# Theory of Final Design

Our final circuit design is seen in Figure (2). The microphones used have a built-in amplification, resulting in a measurable voltage at their outputs. Since a large capacitor behaves as a short circuit to AC signals while preventing a DC signal from passing through, placing a $300\mu F$ capacitor at the outputs of the microphones allowed us to amplify only the alternating part of these outputs. This amplification was accomplished with a simple inverting amplifier with a gain of -300. Since our circuit had to be powered from a single 5 V source, we used the TLV2774, which contains four op-amps which can output near the voltage rails.

The signals were each sent through a second-order band-pass filter similar to the one constructed in Lab 8, seen in Figure (3). In our design, $R_2$ and the positive input to the op-amp are connected to virtual ground

[1] Jens Blauert, *Spatial Hearing: The Psychophysics of Human Sound Localization* (Cambridge: MIT Press, 1997).

instead of real ground, but this does not change the transfer function that we derived:

$$H(s) = \frac{V_o}{V_i} = \frac{H_0 \beta s}{s^2 + \beta s + \omega_0^2}$$

where the maximum amplitude is $H_0 = -\frac{R_3}{2R_1}$, the center frequency in radians is $\omega_0 = \frac{1}{C\sqrt{(R_1 \| R_2) R_3}}$, and the bandwidth in radians is $\beta = \frac{2}{CR_3}$. We chose a large amplification of $H_0 \approx 300$, a center frequency at the frequency of our sound source, $f_0 = \frac{\omega_0}{2\pi} \approx 1000 Hz$, and a narrow bandwidth $B = \frac{\beta}{2\pi} \approx 50 Hz$. The amplifiers in the band-pass filters were the other two amplifiers on the TLV2774.

The outputs of the band-pass filters were two amplified sine waves, $A$ and $B$, centered at virtual ground. By comparing these signals to virtual ground with a LM311, and connecting the 1k pull-up resistor to 5 V, we obtained the digital square wave outputs $A'$ and $B'$, which were offset based on the angle $\theta$ of the sound source, as seen in Figure (4).

These digital outputs were sent to an XOR gate. The output of the XOR, $C'$ in Figure (4), is always zero when the signals are competely in phase, always high when the signals are completely out of phase (which never occurs for our design), and high half the time when the signals are half in phase. By sending this output through a low-pass filter with $\omega_0 \approx 100$ to attenuate the AC part of the signal while amplifying the DC part, we obtain a DC signal that tells us the magnitude of the phase difference between the two signals, and thus the magnitude of the angle $\theta$. Since we had six LEDs, we could have three ranges for the magnitude of $\theta$, so we sent our analog DC signal to two comparators.

The outputs $A'$ and $B'$ were also sent to a positive-edge-triggered D flip-flop, which was used to determine the sign of the angle. As seen in Figure (4), when $A$ is leading $B$ ($\theta > 0$), $B'$ is high when $A'$ is rising, and when $A$ is lagging $B$ ($\theta < 0$), $B'$ is low when $A'$ is rising. Thus, the output of the flip-flop was always high when $\theta > 0$ and always low when $\theta < 0$.

The two comparator outputs that specified the magnitude of $\theta$ and the flip-flop output that specified the sign were meant to be used as inputs to the PIC microcontroller, which would then light the appropriate LED, locating the sound source within a 30 degree span.

# Design Process

Our design is based primarily on Erik Cheever's project description.[2] After coming up with a preliminary design, which is seen in Figure (5), we tested the built-in amplification of our microphones. Using an oscilloscope, we found that their maximum outputs were about 0.5 V peak to peak. We then built a circuit based on our design.

After placing a large capacitor at the outputs of the microphones, we passed the signals through basic inverting amplifiers with a gain of -91, then through comparators. These digital signals, $A'$ and $B'$, were sent through an XOR gate to determine the magnitude of the angle $\theta$. Based on some fallacious logic, we sent $A'$ and the output of the XOR, $C'$, through an AND gate to determine the sign of $\theta$. $C'$ was sent through a low-pass filter with $f_0 = 100$ Hz.

In testing our original design, we found that the amplification was not sufficient, that the output signals were noisy, and that the sign-checker did not function as expected. We thus increased the gain of the inverting amplifier, added a narrow band-pass filter, and replaced the AND gate with a positive-edge-triggered flip-flop. With these changes, the outputs behaved as expected, as measured by an oscilloscope.

Once we had a working prototype, we were able to design our PC board, the layout of which is seen in Figure (6). Unfortunately, we forgot that we had to convert our analog signal which specified the magnitide of $\theta$ to digital signals that the PIC could read, and so we did not add our final two comparators to the PCB design. After our board was manufactured, we soldered our (incomplete) circuit to the board. It did not, however, behave as expected, even though we tested each component before soldering it. Since we would be

[2] http://www.swarthmore.edu/NatSci/echeeve1/Class/e72/E72Binaural/E72Binaural.html

unable to put our entire design on the board anyway (because of the extra two comparators) we decided to abandon the PCB.

We wrote a program for the PIC microcontroller to take the outputs of the final two comparators and flip-flop and light one of six LEDs. The PIC, however, did not function as expected.

## Results

Figure (7) shows the output of the microphones for $\theta \approx 0^o$; the maximum value is about 0.5 V peak-to-peak. Figure (8) shows the output of the band-pass amplifiers; the signal is now filtered and has an increased amplitude of about 5 V. Figure (9) shows the digital output of the comparators, and the output of the XOR for these signals is seen in Figure (10). After the XOR output was sent through a band-pass filter, the signal was a DC voltage, also shown in Figure (10). The output of the flip-flop was high for positive $\theta$ and low for negative $\theta$, as expected.

Table (1) contains measurements of the output of this low-pass filter at various $\theta$. We took measurements from $\theta = -90^o$ to $\theta = +90^o$ in steps of $5^o$. We measured $\theta$ using a protractor and kept the distance of the sound source from the midpoint of the microphones constant by attaching the speaker to a wire. $\theta$ is accurate to $\pm 3^o$, and the output of the low-pass filter is accurate to $\pm 0.1$ V. As Table (1) demonstrates, the output signal generally increases at $\theta$ increases, but it does not increase consistently, and it increaes faster for negative $\theta$. The upper limit on both our magnitude measurements and our flip-flop output was $\theta \approx 75^o$.

Table 1. Output of Low-Pass Filter Measured for Various $\theta$

| $\theta$ | Output | $\theta$ | Output |
|---|---|---|---|
| $0^o$ | 0.4 V | | |
| $5^o$ | 0.5 V | $-5^o$ | 0.3 V |
| $10^o$ | 0.4 V | $-10^o$ | 0.3 V |
| $15^o$ | 0.5 V | $-15^o$ | 1.0 V |
| $20^o$ | 0.7 V | $-20^o$ | 0.9 V |
| $25^o$ | 0.8 V | $-25^o$ | 1.3 V |
| $30^o$ | 0.9 V | $-30^o$ | 1.5 V |
| $35^o$ | 0.7 V | $-35^o$ | 1.8 V |
| $40^o$ | 0.6 V | $-40^o$ | 1.8 V |
| $45^o$ | 0.4 V | $-45^o$ | 1.1 V |
| $50^o$ | 1.3 V | $-50^o$ | 1.5 V |
| $55^o$ | 1.7 V | $-55^o$ | 1.5 V |
| $60^o$ | 1.9 V | $-60^o$ | 2.0 V |
| $65^o$ | 1.5 V | $-65^o$ | 2.0 V |
| $70^o$ | 1.5 V | $-70^o$ | 2.0 V |
| $75^o$ | 1.2 V | $-75^o$ | 1.5 V |

## Conclusion

The binaural locator we designed was moderately successful at measuring the magnitude of the angle at which a 1 kHz sound source was located, and very successful at differentiating between positive and negative angles. The variations in the output of the band-pass filter, which measures the magnitude of $\theta$, were most likely caused by non-idealities in the microphones, such as not being perfectly omnidirectional.

## Attached Figures

1. Setup of Microphones
2. Final Circuit Design
3. Band-Pass Filter
4. Ciruit Output Signals
5. Preliminary Circuit Design
6. PCB Design
7. Output of Microphones
8. Output of Band-Pass Amplifiers
9. Output of Comparators
10. Output of XOR and Low-Pass Filter