Directional Hearing Aid Design

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E90 Senior Design Project
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Abstract

In this proposal we describe our intent to build a directional hearing aid under the guidance of Professor Erik Cheever and E. Carr Everbach. Currently, there is a need for reliable and effective hearing aid designs and we will attempt to implement one given our undergraduate background in engineering. This project will demonstrate our interdisciplinary education at Swarthmore College through the construction of a directional hearing aid. This senior project will serve mainly as an educational demonstration of sound engineering principles and practice.

Introduction

The unavoidable presence of hearing loss and other aural impairments presents an opportunity for engineers to conceive solutions to benefit that segment of the population. In the past, many solutions have been implemented in analog circuitry. In 2003, Emily Eddy developed such a prototype that allows the wearer to receive more audibility in the direction she faces. Our goal is to further the design in a digital landscape that has several advantages over its analog equivalent, ultimately increasing the functionality of the device.

This project is concerned with the robust design and implementation of a digital directional hearing aid. This proposal begins with a technical discussion of the underlying theory, processes, and benefits behind the device. It highlights the project progression and decisions made in the implementation. The project plan subsection further illustrates the proposed tasks in detail, and provides a timeline for anticipated completion. This is organized in the Critical Path Method (CPM) of project planning. The authors then list the relevant qualifications needed for a successful project completion. Finally a summary of required parts and their costs are provided.

Technical Discussion

Hearing loss impairment has long affected many people of all ages. Without sufficient hearing, people lose a valuable sense that can negatively affect their daily experiences. Examples include the exclusion from active family conversations, symphonic experiences, and limited awareness of surroundings. Thus the development of affordable and, more importantly, effective hearing aids is essential.

First, one must have a sophisticated understanding of hearing perception and the effects at play in hearing loss. The frequency range of normal human hearing roughly spans from 20 to 20,000 Hertz. When one suffers from hearing loss, tiny hair follicles in the ear canal wear away, and decrease the upper end before the lower end of that range. This is typical of elderly people exhibiting normal hearing loss from increasing age.

Though other forms of hearing loss are at play, one must also consider the complexities of how the brain perceives the loss and factor that into the corrective procedure. Ultimately the hearing aid's effectiveness and value come from the hearing improvement the user receives. A measure of this can be his/her speech intelligibility with and without the aid; that is, verify how much better the user understands what he/she hears. Another measure is to amplify the speech to a high enough listening level to compensate for the loss.

In our attempt of our own design, we will rely on audible amplification via a directional hearing aid. The underlying principle of such a design is to amplify desired sounds in the direction of interest (usually the one the user faces) and attenuate sources from other directions. To understand this attenuation, one needs familiarity with the operation of a directional microphone. Sound entering the microphone's front chamber creates pressure on the diaphragm, inducing it to vibrate as shown in Figure 1. The diaphragm motion generates electrical signals, which are then processed in an electrical circuit. Thus far, engineers have found difficulty extracting audible speech well from background chatter. Given the difficulty doing so in both analog and digital circuitry, noise cancellation has been regarded a viable solution instead and implemented through multiple directional microphones.

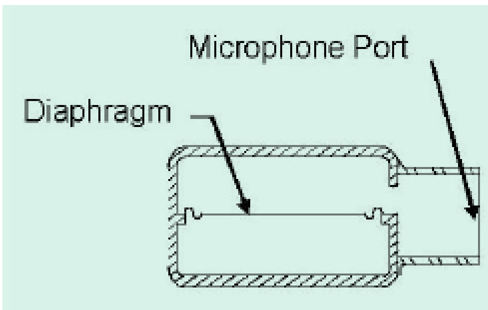

Fig. 1. *Omnidirectional mic function. (Figs. 1 & 2 adapted from Thompson.[1])*

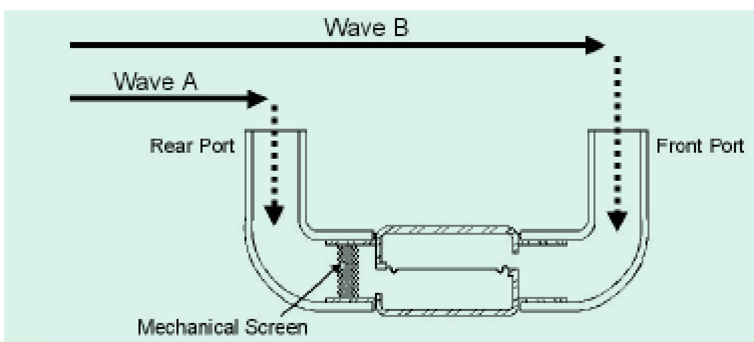

Fig. 2. *A traveling wave and its interaction with a directional microphone.*

Consider the simple case of a two microphone system with a sound source from a rear direction in Figure 2. The rear port will encounter a partial wave (A) first, and find a delay element in the mechanical screen. Wave B will continue its propagation until seeing the front port. With an appropriately set *internal* delay to the time elapsed from wave A and B entering the ports (called the *external delay time*), the two waves will reach the diaphragm together. Because the waves induce the same pressure changes on the diaphragm on opposite ends, the net effect is no vibration from the cancellation. Thus a sound source from the rear direction is attenuated. This feature of non-single directional microphones is summarized in Figure 3.

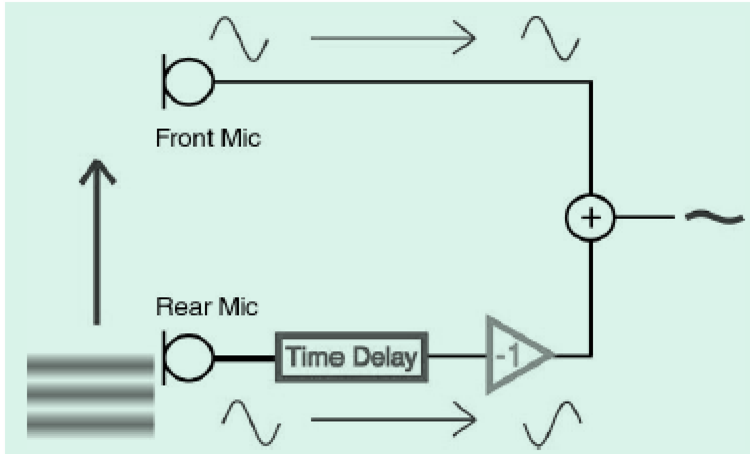

Fig. 3. *Interaction of two omnidirectional mics with a sound wave.*

To attenuate sources in other directions, the process involves setting the delay times appropriately. In the simple example presented earlier, the internal time delay created by the mechanical screen in analog and electronic filters in digital implementations was set to the external delay time. The ratio of the internal to external delay (referred to as β) is set to one and gives the cardioid response shown in Figure 4. Note that the rear direction is completely attenuated and to a lesser degree progressing to the front angle. Changing either delay time in the ratio produces a different profile as seen in Figure 5. Here, the microphone system's β is set to .58 and allows for sound sources from the 125 and 235 degree directions to have the nulls—the point of greatest sound attenuation. For the intuition, the result can be derived from a trigonometric view in Figure 6.

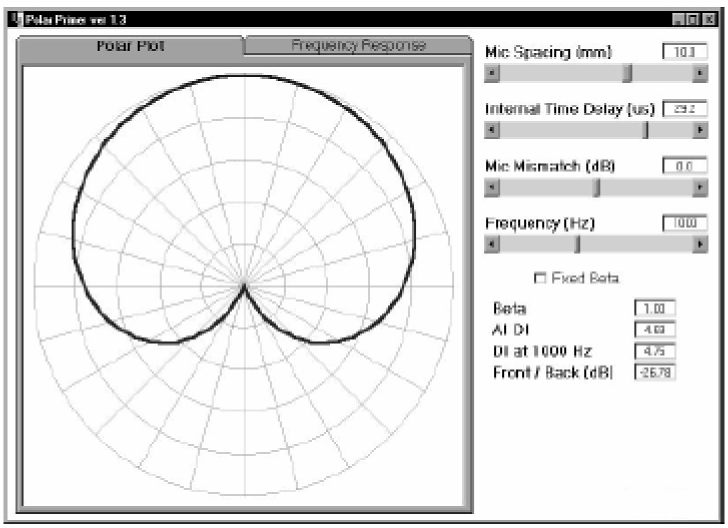

Fig. 4. *A cardioid response derived when beta is equal to unity.*

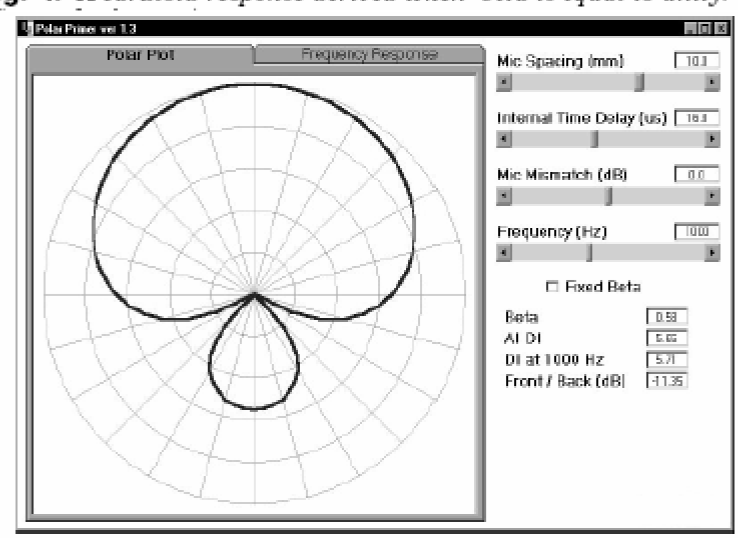

Fig. 5. *Supercardioid pattern, reducing sound from the directions of 125° and 235°, derived from a beta equal to 0.58.*

Call the external time delay X. If the angle of the source comes from an angle of 125 degrees, the time it takes for the wave to travel from the rear to the front port is cos(125). Setting the internal time delay to that or .58, complete sound attenuation is achieved for this angle. Hence, a β set to .58 does indeed produce nulls at this angle and its complement. Through other delay settings, other nulls are accommodated

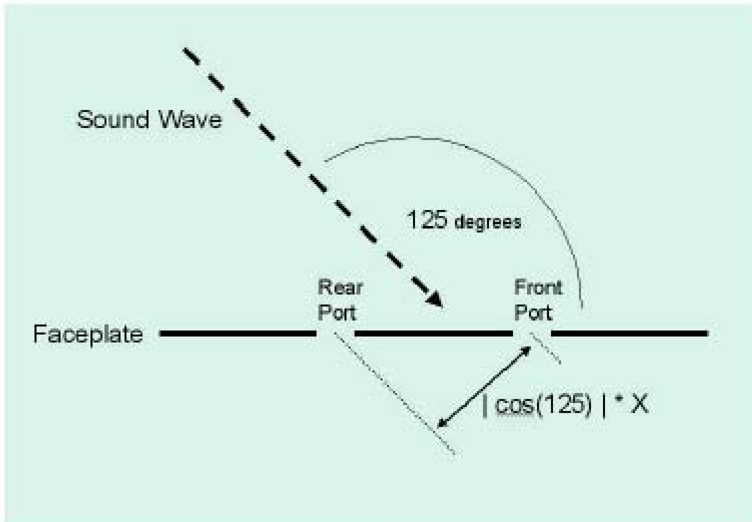

Fig. 6. *A dual microphone faceplate and a sound wave being received from a direction of 125°.*

In our project, we intend to construct a four directional microphone array fixed on a garment (e.g. belt, eyeglass frame, etc.), to realize the benefits of the digital implementation and directionality. The advantage over analog design is the flexibility in digital where software instead of fixed hardware governs the device's operation. In many directional hearing aids, the value of beta is fixed by the manufacturer, allowing only for a single hearing mode. We plan to add flexibility to the user to change it to a mode best suited for particular environments. We also plan to include an omnidirectional mode where normal hearing is best; an example would be in busy intersections where hearing all sounds are important for safety and awareness.

From an engineering design perspective, we expect to work extensively with hardware and software, focusing on testing for robustness and the user interface. We also plan to investigate a number of sound cancellation schemes in code to find the optimal mode(s). Once satisfied, the digital processing would take place in a lightweight box the user would carry in his/her pocket and transmit/receive accordingly with the microphones and loudspeakers. Wires would be needed to connect the microphones and the DSP circuitry as well as the headphones to play the processed sound.

The overall purpose of the project is mainly an educational demonstration of proper engineering practice and design. It would ideally draw from several aspects of our engineering background and allow us to apply them. Depending on the progress made, we leave open the possibility of human testing for commercial distribution.

Project Plan

A. Acoustic Theory

Meet with Professor Carr Everbach to discuss directionality algorithms for end-fire and broad-side microphone arrays. We hope to derive greater understanding of the system for the purpose of implementing such algorithms. This will likely need follow-up individual research to explore possibilities. Sufficiency is achieved once we are fairly confident in our ability to begin development of an algorithm.

B. Set up microphone array with DAQ system

Test a single microphone to ensure desired functionality and suitability. We will first build a circuit with appropriate microphone gains. Once we can reproduce the sound given to the microphone for the frequency range of human hearing, we will position all microphones in desired array configuration. Amplification will likely be necessary, so simple operational amplifiers gain circuits will be designed and implemented. Initial tests will be performed on a breadboard at first, but ultimately implemented on a printed circuit board laid out in Multisim and Ultiboard. After Professor Everbach tests the DAQ board, we will need to connect it to a PC in preparation for further testing. When we are finished we will have fully characterized the array of 4 microphones sampled at various frequency ranges from 20 to 15,000 Hertz.

C. Gather sound samples from anechoic chamber

First, we need to acquaint ourselves with the Sound Booth Lab. With the DAQ system and microphone configuration in place, we plan to make several test sets by recording tonal sound sources. Fixing a sound source at a particular distance—one approximate to human distance in conversations—away, we will rotate the microphone array on a rotating stand. Through this we will have measured sound samples from several directions. We desire a series of test cases where the sound samples have single and multiple frequency content. As a challenge case, we would like one trial to be a human voice. Prerecording our tests allows for greatly increase convenience and consistency.

D. Algorithm development in MATLAB

Here we are to focus on the software implementation to achieve directionality in our output signal. Desired functionality includes the option to switch directionality angle on command by either by relying on a control loop or user definition, as well as the ability to switch back to an omni-directional mode which may be preferable in particular environments.

E. Algorithm testing and comparison

Apply algorithms to the gathered sound samples and plot the associated directionality response. Ideally, we desire an algorithm where maximum directionality is obtained in the direction of interest while all other directions are attenuated. This will require comprehensive testing and analysis of the responses of each algorithm to the test cases acquired from the Sound Booth Lab. Such testing would first include PC analysis of sound profiles from each algorithm, comparing the level and type of attenuation to those expected. We would then attempt a few real-time samples using live sound sources seen from different directions, comparing sound profiles and surveying hearing improvement from human subjects.

F. Design A/D Hardware

Here we design and construct the printed circuit board assembly to which the microphones connect. This PCB contains our A/D converter that provides sufficient sampling rate and channel capacity. Also required is a space reserved for a DSP chip providing sufficient memory, processing speed, and channel capacity for the A/D output. We anticipate four channels for each of the microphones in the array, and a sampling rate 30 kHz to meet the Nyquist Criterion; we are designing for a maximum hearing frequency of 15 kHz based on the target user.

G. Convert MATLAB algorithm to compatible DSP Chip

Here we acquaint ourselves with our DSP chip (specified in the Project Cost section) enough to translate our chosen algorithm into a form that the DSP recognizes. Extensive time will be devoted to learn the chip's compiler such that we may implement the desired functionality of our algorithm.

H. Construct D/A Hardware

We will connect a chosen D/A chip to a method of output. Initially, we will work with standard headphones as a way to transmit the processed signals into audio. However, we will seek to find a more cosmetic implementation such as tiny loudspeaker as part of a moldable ear mount or one that fits inside the ear canal.

I. Combine Hardware

Connect the A/D receiver portion of the system to the DSP chip for processing. Then, connect the processed output of the DSP chip to our D/A chip for purposes of transmission. All of this installation will be done on a PCB.

J. Real-time Test Functionality and Debug in Anechoic Chamber

This step provides realistic testing of hardware implementation in a real-time setting to ensure that the device works as expected. We will also perform any necessary final debugging and modification. This requires a large effort to comprehensively test functionality.

K. Write Report

Compile all previous data, analysis, and documentation in a coherent and cogent report.

L. Present

Prepare for a well-thought out and comprehensive presentation. Run through practice runs at advisor's discretion.

Activity	Needs	Feeds	Duration	Effort	Action
A	-	D	1w	12h	Acoustic theory
B	-	C	4d	8h	Set up microphone array with DAQ system and characterize
C	B	E	6h	6h	Gather sound samples from anechoic chamber
D	A	E	4w	3d	Algorithm development in MATLAB
E	C,D	G	2w	10h	Algorithm testing and comparison
F	-	I	2w	1d	Design + Construct A/D hardware
G	E	I	6w	3d	Make MATLAB algorithm compatible with DSP chip
H	-	I	3w	10h	Design + Construct D/A hardware
I	F,G,H	J	10d	12h	Combine hardware
J	I	K	2w	1d	Test real-time functionality in anechoic chamber
K	J,L	L	2w	1.5d	Write report
L	K	K	4d	12h	Prepare presentation
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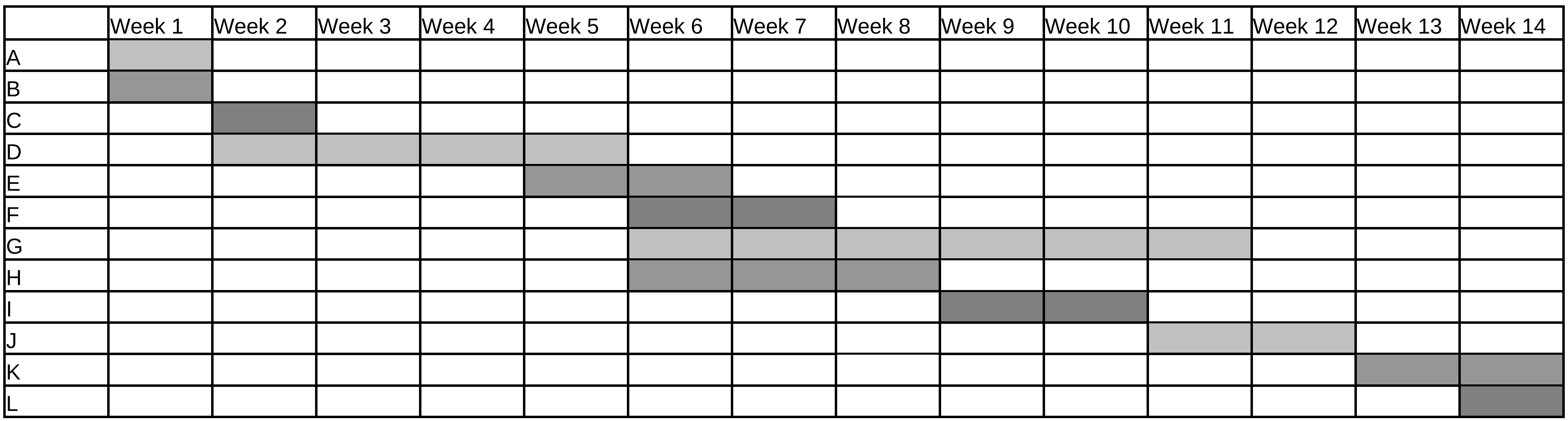
Critical Path Method



Gantt Chart

Project Qualification

As senior engineering students at Swarthmore College, we feel confident of our technical backgrounds and knowledge to complete this project and accomplish our goals. We both have experience working with electronic circuit design and digital logic in our coursework as well as a strong understanding of engineering methodologies and design constraints. We have cultivated a positive working history in past coursework and projects that should enable us to collaborate efficiently and smoothly. We have chosen our faculty advisors well—ones who are experienced engineers in acoustics and electronics— to propel us toward the project's completion.

In terms of technical expertise, both of us are adept at using MatLab as a computing tool. From David's experience working at Bose, he is well versed in performing measurements and running analysis in acoustic applications. Mark's minor concentration and interest in computer science will prove valuable in the algorithmic development of the device.

The nearby Sound Booth Lab in Papazian Hall will function well as an appropriate project environment. It includes an anechoic chamber ideal for sound equipment testing as well as data acquisition systems already in place. Furthermore, it is likely we will have the facilities to ourselves during the course of the project and removes any concerns of its availability.

Project Cost

Microphones	4 x \$16 = \$64
Texas Instruments DAQ Board	(already ordered)
A/D Chip: TI ADS8364	\$28
DSP Chip: TI TMS320C6203BGNY173	\$95
D/A Chip: TI DAC7641	\$11
PCB's	~\$60 from expresspcb.com
Operational Amplifiers, Resistors, Wiring	Available in Hicks
	Total Cost: ~\$258

References

1. Thompson S: Dual Microphones or directional-plusomni: Which is best? In S Kochkin
2. Csermak, Brian: A Primer on a Dual Microphone Directional System http://www.gennum.com/hip/pdffiles/DualMic.pdf