ACOUSTIC NAVIGATION FOR MOBILE ROBOTS

Project Proposal

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Table of Contents

Statement/Objectives

The requirement of this project is to design an acoustic navigation module that will detect sound in a 360-degree planar environment using a microphone array that will allow a mobile robot to perform movement based on sound location. This project will involve all aspects of engineering design, from planning to implementation.

Our objectives are the following:

1. Develop a printed circuit board interface for the microphone array.
2. Design acoustic fixtures to increase directionality of microphones.
3. Develop algorithms to determine the angle and position of the transmitted sound.

Background/History

Acoustic navigation is not new; microphone arrays have been in use for over 30 years. A microphone array consists of a number of microphones distributed in some known pattern. The microphone interface converts the sound signal to electrical signals which can be processed in such a way to maximize directionality and the signal-to-noise ratio. The microphone array is becoming popular in acoustic designs due to its ability to outperform even the best directional microphones. The conversion from analog to the digital realm has introduced exciting new possibilities for acoustic navigation and microphone arrays. In the past the quantity of microphones in an array could be no greater than around 20. Modern arrays, however, can handle hundreds or even thousands of microphones. The reason for this is due to the advantages of digital over analog. First, implementation of the system with software allows ease of modification. Algorithms can easily be changed and updated without having to implement new hardware. Modeling software also allows much more complicated algorithms to be developed cheaply and quickly, rather than using analog devices. Further, the digital design of a microphone array is less susceptible to noise and interference, allowing for greater precision. Finally, the design of a digital system is becoming less costly to build.

The arrays have a number of applications including use in conference rooms/auditoriums, concert events, surveillance equipment, and robotic designs. Recently, Dr. James Flanagan of Rutgers University led a team that designed an array using 380 microphones that was implemented in an auditorium and would scan the room for sound sources. This research proved to be invaluable, as the possibility of having a low-cost and precise way to provide audio-video support to a large auditorium for conferences and meetings is of interest to many companies and organizations.

Management

Trent Foley □ Lead hardware design engineer

Josh Earley □ Development of array, acoustic directionality, board design

Chris Gonzales □ Software design, budget, procurement of parts

Thomas Garner □ Lead software engineer, board design

Schedule

M □ 6:15 p.m. (meeting with advisors)

M-W-F □ 1:30 p.m. - 4:00 p.m. (regular design meetings)

Weekends □ open (when necessary)

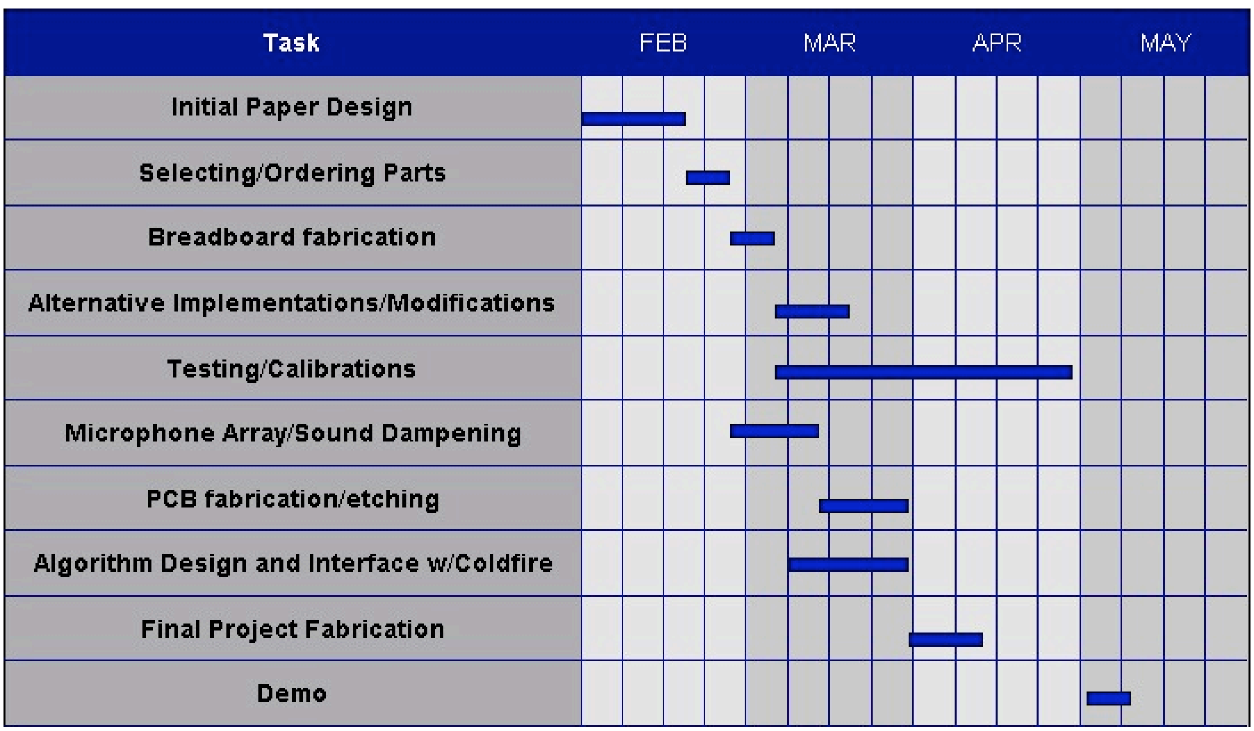

Figure 1: Gant Chart

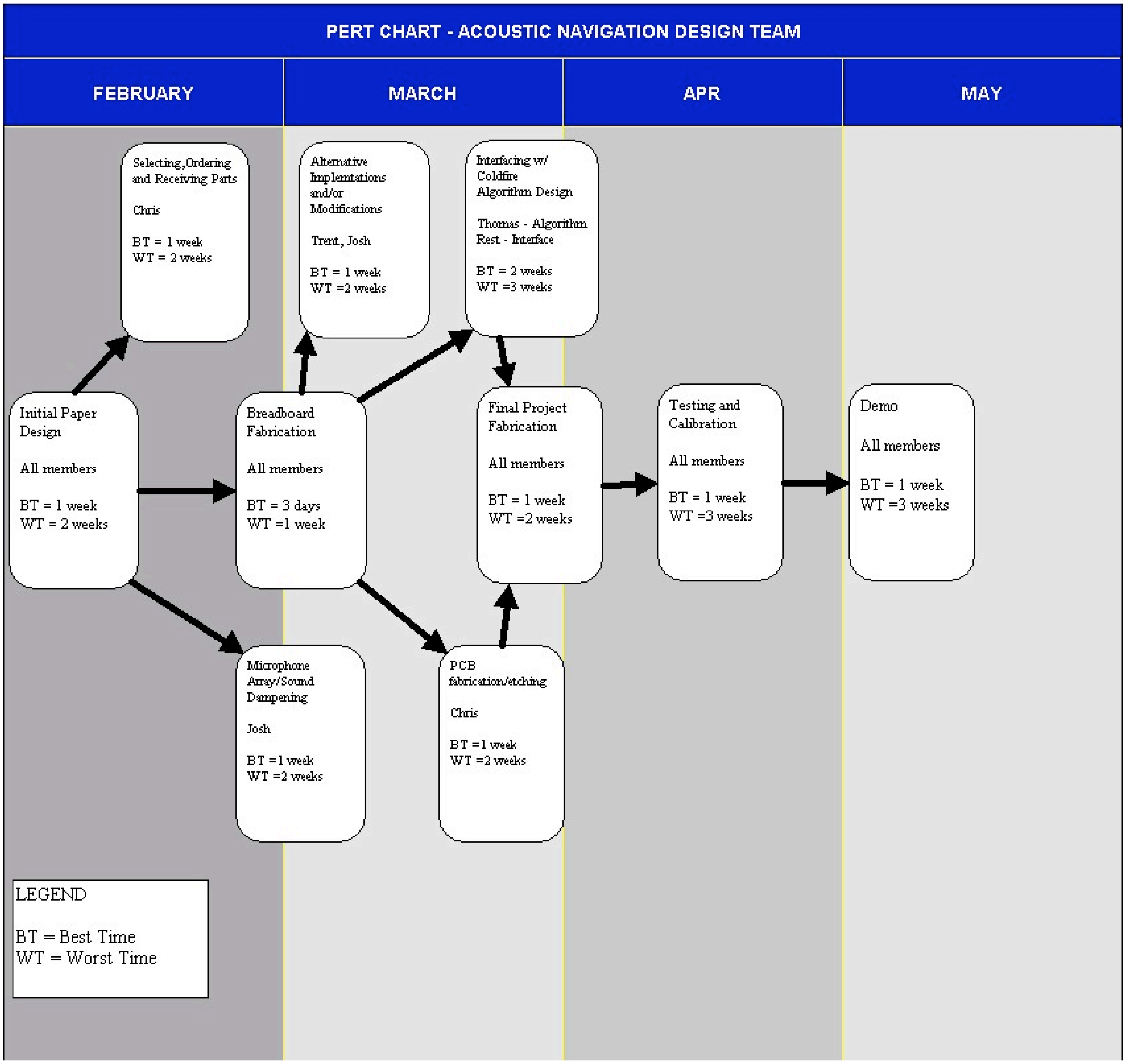

Figure 2: Pert Chart

Survey of Existing Solutions

Several implementations of acoustic navigation control for mobile robots have been carried out in recent years.

Phonotaxis

At the University of Stirling, Scotland, research into the behavior of crickets has spurred the development of a cricket robot model. The robot is to simulate the behavior of female crickets □ who locate male crickets by means of phonotaxis to the songs they produce. The sound sensor design includes an array of four microphones to model the cricket's ears. It processes the signal and outputs control commands to a six-legged robot that it is mounted on. Documentation on the project can be found at:

http://www.stir.ac.uk/Departments/HumanSciences/Psychology/Staff/bhw1/phonotaxis/

Sound Sensor

At the Convict Episcopal de Luxembourg, the Boulette I.T. group has developed a sound sensor that measures the difference in phase of two audio signals to determine the source of the sound. Their implementation consisted of two microphones and a simple hardware interface to be mounted on a robot so that it can search and advance toward a pulsing sound source. The robot then separates the signal peaks, waits to react until the signal has grown over a threshold value, determines which microphone received the signal first and measures the time difference between the triggering of both the sensors. Documentation on the project can be found at:

http://www.restena.lu/convict/Jeunes/PhaseSound.htm

Audio Frequency Sensor

Another project developed by the same group is an audio frequency sensor. Their design included a single microphone, an audio amplifier, comparators, a micro-controller with an external oscillator, an LED-net, an integrator, and a pulse-width modulator. By measuring the pulses of the audio signal, they were able to determine the frequency. This project's documentation can be found at:

http://www.restena.lu/convict/Jeunes/Spielberg/Spielberg.htm

Microphone Amplification

At the University of Florida, Steve Stancliff developed "Bob", an autonomous quadrupedal robot that implements a sound sensor to create an audio-tropic behavior resembling that of a small animal. The main issue concerning his project was dealing with distortion created by single-powered amplifiers. He was able to overcome this problem by using a dual-powered operational amplifier, but dual-powered chips are unsuitable for most mobile platforms. Documentation on the project can be found at:

http://www.mil.ufl.edu/projects/bob/bob.pdf

Signal Rectifier

Elliott Sound Products has developed an LED audio VU meter that uses simple full-wave rectifier and pre-amplifier hardware circuitry. Figure 3 shows a schematic of the preamplifier and full-wave rectifier. Documentation on the project can be found at:

http://sound.westhost.com/project60.htm

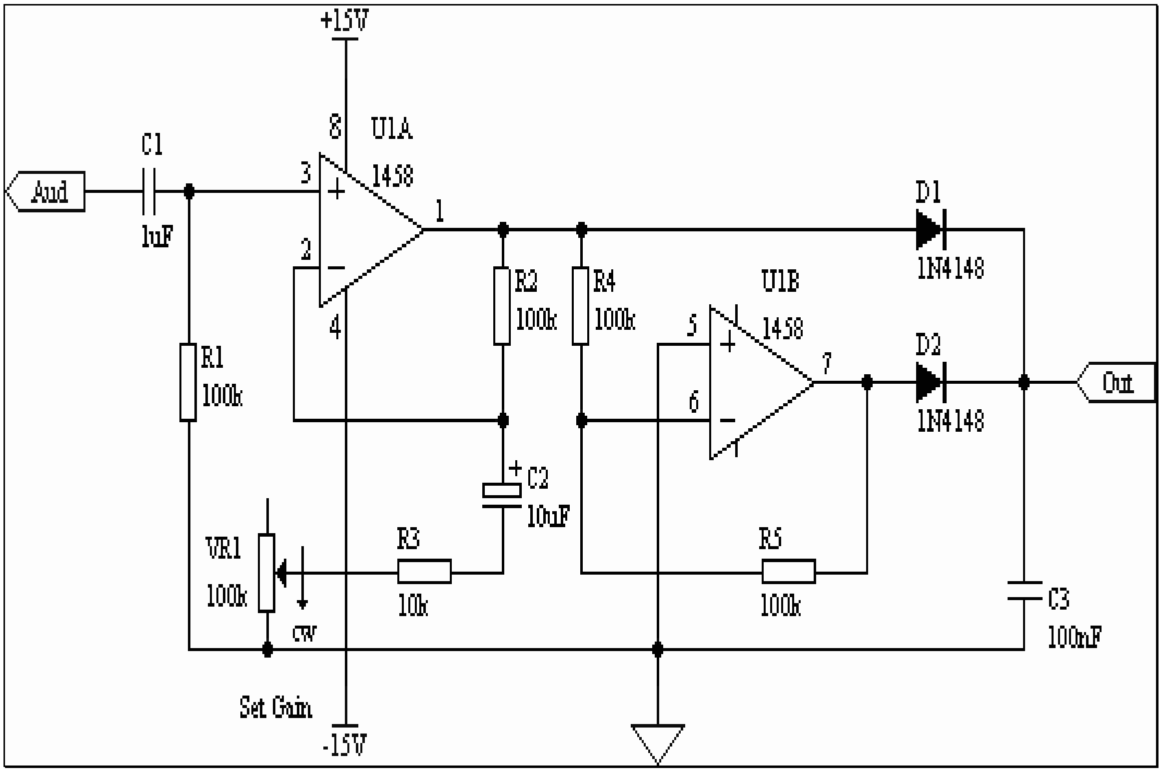

Figure 3: Elliott Sound Products Simple Full Wave Rectifier and Preamp

Design Approach

Microphone Array

Our team has decided to use 8 microphones in our array. We feel this will give us the optimal compromise between accuracy, cost, and ease of design. The microphones will be placed in a 360 degree pattern, allowing each microphone 45 degrees of coverage. Our team has decided to use electret microphones due to their properties. They are very small, allowing for multiple microphones to be placed in a small area. Further, electret microphones demonstrate a good frequency response and signal-to-noise ratio. These microphones are mainly used in audio-video applications such as television and presentations. They are widely available and inexpensive.

To introduce directionality with greater precision that the microphones can provide, we have decided to fabricate a sound dampening system to ensure that each microphone only picks up noise in its angular position in the array. To do this, we have two designs being considered. First, we were given the idea of using empty toilet paper rolls. These rolls, properly manipulated, would dampen any sound coming from unwanted angles due to the properties of cardboard and the curvature associated with the rolls. This implementation would prove to be extremely cheap and easy to design. The second idea our team is considering, will be to fabricate some fins with a sound dampening sheet over them. Dynamat, a company that makes sound dampening sheets for car audio applications, would make a great choice for the material used in our design. These sheets are inexpensive and can reduce sound from 3-6 dB, which would be more than adequate for sound localization. Due to these inexpensive applications, we will design and build both and decide which design will be best based on our testing.

Hardware Interface

Microprocessor Input

Our proposed design takes a simple approach to providing a hardware interface. Due to our familiarity with the Motorola Coldfire microprocessor and its interfaces, we designed a microphone array that interfaces with the Coldfire's 8-bit UART. We designated 4 bits to represent signal intensity, 3 bits to represent the currently polled microphone, and a single bit to determine the filter type (high-pass or low-pass). With a larger interface, we could represent the signal intensity with higher resolution and apply additional filters for determining more precise frequencies, but for the sake of simplicity and the fact that 4-bit resolution will be plenty for our purposes, we chose to keep it small.

Hardware Design

Our hardware design polls each microphone in the array sequentially, alternating the filter type on each cycle. The entire cycle then consists of 16 distinct combinations of microphone and filter that are presented at the interface sequentially and held for a designated time determined by the clock (crystal oscillator). By using a counter connected to the clock, we can attach the microphone array to an analog multiplexor controlled by the counter to poll between the microphones. The currently polled microphone signal will then be amplified and split through a low and high-pass filter (Low: freq < 800 Hz. High: freq > 800 Hz). The resulting two signals will then be polled by means of a multiplexor controlled by the higher order bit of the counter. The signal will then be rectified and passed through an analog-to-digital converter to be interfaced with the microprocessor. Figure 4 shows our preliminary block diagram of our hardware design.

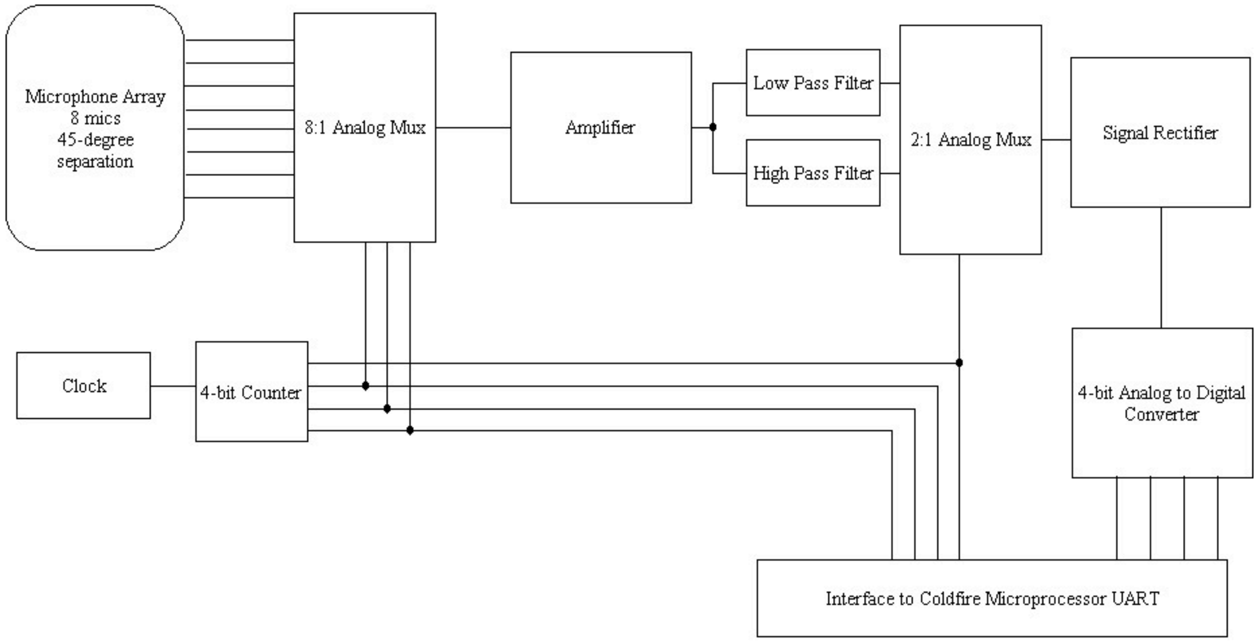

Figure 4: Preliminary Hardware Design

Societal, safety, and environmental considerations

There are relatively few societal and environmental concerns for our design project. Our project deals with electronics that are found in many places in society today. Our project also does not really create any new technology, but leverages many pieces of existing technology to create new functionality. This fact should neither surprise nor affect the public any way. The extent of the environmental concerns is also relatively minor. One of the design deliverables is a printed circuit board. This board will be manufactured off-site. The manufacturing process of this board will very likely involve hazardous chemicals. While this should be of concern to us, in that we should use an environmentally friendly manufacturer, the exact concern falls primarily to the vendor. The final design could also contain electronics, such as capacitors, resistors, etc., that, if driven beyond their operating limits, could pose an environmental threat. It is a known fact that capacitors can explode, if operated improperly. It should then concern the design team to be cautious when dealing with these electronics, so that they are operated safely. The safety concerns of our project also deal with the large amounts of electronics that will be in our project. The concerns are like any other electrical device. The device should be operated properly, and within its designed constraints. Electronics always carry an inherit risk of electrical shock. The introduction of water and moisture to or near the final product should be avoided. The final device should be properly connected to both a power source and ground source of the type prescribed. The final device should be used in the way that it is intended to be used, so that injury or accident may be avoided.

Alternate Solutions

The design team arrived at several other alternate designs, during the discussion of the final design. The team's first design found the microphone with the largest amplitude. This was accomplished by feeding each of the microphones into an amplifier and then into a rectifier. This signal would then be fed into a 2-way analog comparator. Assuming the use of eight microphones, four comparators would then have the four largest signals. These signals would also be compared in pairs, resulting in two signals, which would also be compared. This final output would then be that of the loudest signal of the loudest microphone. This design would give accuracy of 45 degrees. This design can be further improved upon by adding multiple band pass filters. These filters would be placed between the microphone and the amplifier. An analog multiplexer would be used to choose between the several filtered signals. The filtered signal would be fed into the amplifier and rectifier. Another design variation could be introduced at this point in the circuitry. Either the signal could then be passed to the arrangement of analog comparators for comparison, or it could be directed to a digital path. The signal at this point could then be fed though a sample and hold mechanism and an analog to digital converter. This resulting signal could then be compared digitally, by digital comparators in hardware, or fed to a larger processor to be processed off of our device in other hardware or software.

Another possible design is based upon the way that humans detect the direction of sound. By utilizing some form of digital signal processing, the time difference between microphones that occurs due to the non-infinite speed of sound, and the level difference between microphones that will occur due to the varying distances between the source and each of the microphones, the exact direction of the sound source should be able to be located much more accurately.

Budget Analysis

This is a proposed budget of our design. As we start to implement our design, we may need to include more items. However, the following list consist of the main parts that are needed. Any additional items should not significantly raise our budget.

Part:	**Qty.**	**Estimated Price:**
Microphones	8	$13.00–$20.00
IC's/Capacitors/Resistors	5-10	$1.00–$2.00
Mobile Robot	1	$200.00+/-
A/D Convertor	1	$2.95–$5.39
Filter	2	$1.00
Comparators/ Mux	2	$2.65–$5.39
Construction of PCB	1	$33.00++
Amplifiers	1	$2.95–$19.95
Tools, Wires, other parts	–	$30.00
		Overall Max Total: $481.12

Design Constraints

The following is a list of design constraints that will either limit are capabilities of design or give us guidelines for testing. These constraints will be taken into consideration when implementing and testing our design.

Cost

For this design project, we have been given a budget to purchase the necessary equipment needed for implementation. We have not been given a definite budget limit but it must be reasonable. Therefore, for example, we should not necessarily purchase the most expensive microphones or mobile robot. On the other hand, we should not buy items that are too cheap or not reliable. The selection of parts is very important for our project and price must be considered for everything.

Size / Weight

We have not specified dimensions of our design but it will be of reasonable size. It can not be too small or too big. The size of the project will be based on the size of our microphone array and on the mobile robot we choose to work with. The robot must be of a size to be able to move a reasonable distance forward and backwards in a 360° environment. The weight of the PCB and attached components (e.g. microphone array) can not weigh more than what are mobile robot can carry. If we chose a lightweight robot that can not carry a significant amount of weight than the robot will not function properly.

Compatibility

Our design of our PCB must be compatible with other parts of our design in order for it to function properly. It must be able to connect with the microphone array we make and with the mobile robot we choose. The PCB and robot must also be compatible with the software we use to control the direction of the robot. This is a very crucial constraint that can be important if not taken into consideration.

Power

Our design must consist of a reasonable power supply. Obviously, it must be portable for the mobile robot and can not be of massive size or weight. In our design, we must design our PCB to take a power supply that is affordable and does not require its own generator, in an extreme case. The robot's power source will be assumed to be in the range of 0–10 Volts. This also then constrains the types of chips that are available. To minimize circuitry and unnecessary voltage conversion, the final design will contain chips whose operating voltages (V_{DD}, V_{CC}, etc.) are the same. This exact voltage will depend on the exact robot or robots upon which our design will be tested. Again, this will decrease the complexity of our design, by not requiring any more power source conversion than is absolutely necessary.

Sound Frequency Range

The acceptable sound frequency used for testing will be based on the microphones we use for the array. We will want the microphones to be able to recognize sounds in the spectrum that a human can hear. They must be able to respond to tones that are both high and low.

Outside Interference/Environment

The robot will be tested in a very quite environment so that it will only pick up those sounds that we produce. Any outside sounds can effect the testing of the robot. Obviously, we will not be yelling during our testing to see if the robot responds to the tones. This will also be based on the selection of microphones we choose.

Robot Mobility

The robot we choose must be able to move in a 360° environment. It must be able to move at angles, not just up, down, left and right. Our placement of microphones and the number of them we use will contribute to its movement.

Single Tone Testing

In testing our robot, it will respond to a single tone. We are not designing the robot to respond to multiple tones that are going off simultaneously from different angles. Doing so will confuse our robot and it will not be able to behave accordingly to the tones.

Time

For this senior design project, we have the remainder of the semester to build and test our design. The final presentation and demonstration is due May 5th or May 7th. Our progress will be monitored on a weekly basis and we will submit bi-weekly reports of our status.

Design Validation

Once our design of our project is approved and we have a completed construction of the model, we will test it. In order to validate that the model we created actually meets our design criteria, we have included a checklist in order to test it. This list may be altered if other requirements or limitations must be met while implementing our design.

Design Validation Checklist
In 360° environment, robot responds to a single generated tone.
Robot moves directly toward or away from source of sound.
Robot responds to both high and low frequency tones.
Robot stops when sound source is inactive or turned off.
Robot moves to sound source at 0°/360° (directly above it).
Robot moves to sound source at 45° (north-east of it).
Robot moves to sound source at 90° (directly to the right of it).
Robot moves to sound source at 135° (south-east of it).
Robot moves to sound source at 180° (directly behind it).
Robot moves to sound source at 225°/ -135° (south-west of it).
Robot moves to sound source at 275°/ -90° (directly to the left of it).
Robot moves to sound source at 320°/ -45° (north-east of it).

The list checks to see if the robot responds to sound in every direction in a 360° environment and does what it is supposed to do. An example of the 360° environment is show in Figure 5. If the robot successfully completes the entire task list then our design is correct and our project is complete.

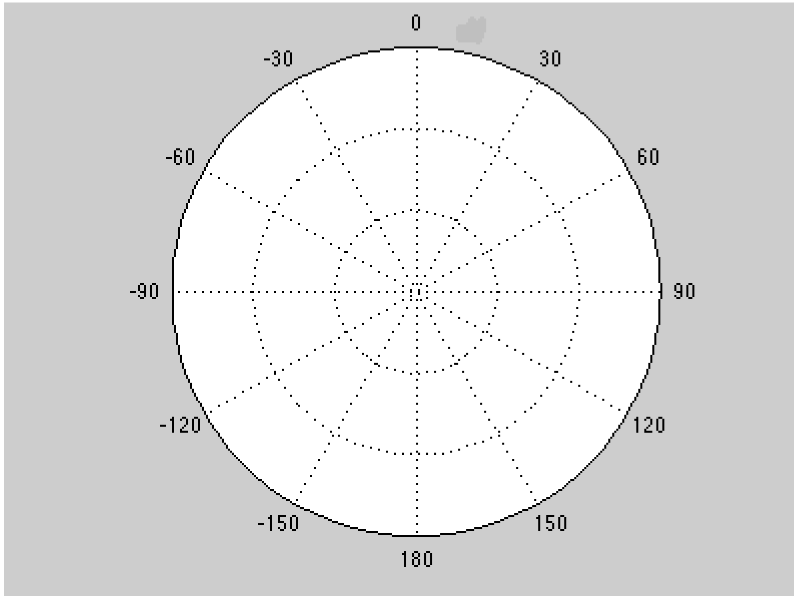

Figure 5: 360° Testing Environment

Feasibility/Conclusion

After reviewing our design constraints and design approach, we feel that are proposed design is feasible given the amount of time we have. The parts and tools necessary for completion can be accessed without much trouble and are within a reasonable budget range. We will be able to complete the project with enough time for testing and to make whatever changes that may occur.